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A Routing Delay Reduction Mechanism Based on Dynamic Switching through Congestion Prediction in Wireless Mobile Network

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Abstract*: The advancement in wireless mobile networks (WMNs) and their technologies are becoming the most popular technologies today and are widely used in various communication services. Devices for communication are equipped with wireless network interfaces through which they can efficiently stream data to their network-wide devices. However, due to the extreme use of different services, and TCP incapacity to control congestion, it causes a high overload in wireless communication, which makes the network extremely unstable and congested. Conventional routing methods with congestion control mechanisms for multimedia streams are not reliable for transport layer protocols. In such a case, it makes wireless networks very complex and challenging to perform smooth routing due to its dynamic characteristics and resource constraints. It can be overcome through scheduling flow control in multiple network interfaces for smooth packets streaming data, and even it can play an important role in congestion control by overwhelming the buffers of intermediate nodes and reduction in packet loss. In this paper, we propose a Routing Delay Reduction Approach (RDRA) based on dynamic switching by estimating the transmission rate and delay probability through congestion prediction. The goal of RDRA is to reduce routing latency and packet loss and improve throughput at different traffic rates. Experimental evaluation measures at different traffic rate variation it shows an improved performance in terms of throughput and low latency compared to the related protocols.

***Index Terms*: WMN, Routing, Delay Reduction, Dynamic Switching, Congestion Prediction, MANET.**

I. INTRODUCTION

Self-configuration and dynamic routing make MANET the future of communications. Due to the increasing demand for communications in various fields from text data to multimedia data at the same rate, the transmission rate also increased. Increasing the data rate causes congestion in the network and at the same time reducing the rate leads to reduced throughput and quality of service, which is a research challenge in the wireless network [1], [2], [3]. They are more complex and challenging in MANET due to dynamic nature, resource constraints and dependency on intermediate nodes. Congestion is often caused by excessive input of the traffic rate into limited network bandwidth and congestion control methods limit this traffic rate to prevent congestion [2], [5]. The harmful characteristic of MANET makes it difficult to provide long network stability with low congestion and overhead [4], [6]. Current TCP-based congestion protocols avoid congestion by minimizing traffic or routing through a different route. This may be efficient in a one-to-one node connection, but in the case of a one-to-one node, it is completely ineffective.

The growing use of the wireless network challenges the essential elements of wireless design to support the current high-speed service and future applications. Transmission Control Protocol (TCP) is the trusted primary transport protocol on the WMN network [4]. TCP provides excellent performance while remaining fair to other connections. TCP uses a congestion control aircraft based on the windows method. Despite these advantages, TCP has many limitations and is not suitable for all application models [5]. With these cutting-edge technologies and cost-saving trends, devices can have different network interfaces such as "Ethernet" or "Wi-Fi" to access the different application services network, most of which are known as multi-home. The term homing is frequently described as a transmission among the transmitter and the receiver with numerous links at the equivalent time. It can enhance the number of different network standards in terms of reliability, productivity, and transmission by designing appropriate route scheduling mechanisms.

Congestion control techniques are widely used to reduce congestion over a wired or wireless network [6], [7], [8]. However, these technologies are unable to cope with the increased use of streaming multimedia applications, resulting in increased congestion, packet loss over the network and inappropriate broadcast service. Mostly in a wireless network, it has a negative impact on performance due to the loss of high number packets. The challenge in the MANET environment is very high because of the unreliable networking situation and the heavy reliance on intermediate node collaboration. It can be handled efficiently by ensuring that packet loss is minimized by routing traffic to congestion to a minimum. In addition, traffic routing is defined as a distributed and scalable routing algorithm and simply requires "local information" from the local node and its immediate neighbors, as well as a "routing congestion status" message that is relatively easy to obtain.

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In [9] suggests that QoS aware routing protocol (CADM) is the best effort. The protocol is an adaptive and reliable congestion control derived from multi-rate adaptation.

It utilizes wireless resources efficiently and provides improved QoS assist for delay-sensitive communications in multi-tariff MANETs. In CADM, the source wants to find multiple node paths that meet QoS requirements for latency and selects high data rate links for high speeds when sending packets. However, since congestion is caused by insufficient control of the buffer pool, the performance of TCP is very low, so an algorithm can be designed to overcome this problem. It has been observed that most existing recommendations tend to use the shortest data transmission path, but at the same time, there is a high degree of congestion. Therefore, it is important to ensure that delays and packet loss are handled efficiently before routing data in the route.

In this paper, we propose a Routing Delay Reduction (RDRA) approach based on dynamic switching by congestion prediction by estimating the transmission rate and probability of delay. This system will adapt a dynamic route to switch to the optimal route instead of following the previously detected path or shorter path unlike the traditional routing or current routing method. To perform the switching dynamically, RDRA implements a Route Control (RC) device that provides the best node congestion on the road which will achieve the goal of the proposal to achieve minimum routing delay and improve throughput.

The following paper is organized into four sections. The section-2 discuss the related works, the section-3 described the proposed routing delay reduction approach, this section-4 present the experiment evaluation, and the final section-5 summarized the conclusion of the paper.

II. RELATED WORKS

In today's requirements, wireless networking plays an important function in the next generation of Internet connections. Increasingly new services are increasingly available in business, entertainment and social networking applications in all wireless networks because of the advantage of seamless mobility and avoiding congestion caused by the widespread use of the Internet [10], [11]. However, these current technologies are unable to cope with the increasing use of streaming multimedia applications that raise high congestion and packet loss on the network and cause inappropriate broadcast service [12], [13]. For instance, the data produced in a crisis is the most important, and the loss of such data may damage the goal of deploying a wireless network. At the same time, congestion control in the wireless network must be based not only on the efficiency of the network but additionally must have basic reliability of applications.

Congestion tends to focus on preventing congestion, as it results in a significant loss of network in conditions of quality of service and resource utilization. Congestion prevention mechanisms provide a solution based mainly on methods of "network packet scheduling", "congestion forecasting", "congestion awareness", and "congestion mitigation".

- *Network Packet Scheduling*: It is a fraction of a node that is used to examine the reception and scheduling of packets [12]. It is also fully responsible for active queue management, such as dynamically maintaining the priority of packets as needed [25], *i.e.*, packet sequencing, packet dropping mechanism, priority change, etc.
- *Congestion Prediction*: It is an extremely significant way to control congestion [2]. If it is predictable early, his control will be more effective. There are several congestion prediction mechanisms such as "queue-based congestion detection", "throughput-based congestion detection", and "round-trip time-based congestion detection".
- *Acknowledgment of Congestion*: It is the subsequent stage of the mechanism for detecting and avoiding congestion [6]. After congestion is detected, the most important way is how it informs the congestion node. Both implicit and explicit measurement techniques are known for reporting congestion. Implicit techniques reduce data transfer, but sometimes explicit mechanisms work well when compared to implicit mechanisms.
- *Congestion Alleviating methods*: After identifying congestion at several nodes, a congestion prevention algorithm is executed at the source node [7]. It generally aimed at the speed alteration of the flow to the provisional clarification of the transport layer. However, for a while the routing algorithm is modified to maintain the same flow rate that is maintained so that the source can switch packets in the other direction whenever possible. Repeated congestion has a significant impact on network throughput, so it cannot achieve better results.

The most widely used transport protocol, TCP (Transport Control Protocol) [14], does not apply to wireless applications, such as streaming over mobile wireless networks. This is due to the fact that TCP interprets the lost packets as evidence of network congestion, which cannot always be applied to networks of ad hoc mobile network types. The packet loss able to due to the exclusive attributes of MANET, for example, "node navigation", "channel bit error", "intermediate competition", and "path failure." Due to this special nature, the packet loss rate of the wireless link is much higher than its wired connection. TCP responds to these wireless losses in the same way as packet loss due to congestion. This is because it aims to deal with losses in this path. The congestion control is a major problem in wireless networks, especially in MANET.

In [15], link-level protection and acknowledging mechanisms are liked to advance TCP performance on ad hoc wireless networks. In [14], the TCP problem in dynamic multi-hop wireless networks is identified, and other systems for accessing media and routing layers are suggested to develop TCP performance. The "Explicit Link Failure Notification (ELFN)" procedure was considered in [16], [17], which is relied on clearly notifying TCP sources of link failure to advance TCP performance.

More recently, the WMN "Adaptive Response Transmission Protocol (AR-TP)" has been suggested in [18] to equitably distribute network resources between multiple streams whereas reducing performance load. AR-TP comprises competent 1-hop modulation and dependability methods to accomplished reliable high-performance data transfer in WMN. In compared to the end-to-end tariff control system, AR-TP's rapid mobility adaptation strategy allows each router to monitor dynamic wireless routing requirements.

W. Song et al. [11] proposed a method of preventing congestion on the basis of two factors of perception of congestion. These two features are due to the congestion complications recognized in the routing protocol, namely "buffer occupancy rate (BOR) " and “successful frame transmission rate” (SFSR). The "BOR" operator identifies the resource that the node uses and is available on the node, and calculates the ability to forward data based on the resource. The "SFSR" factor calculates the cost of sending data packets in the zone within the zone. However, this Recommendation shows low productivity performance in different mobility scenarios because the entire path depends on the path chosen, and the failure of the correlation failure due to mobility cannot maintain the desired product performance.

Lochin et al. [19] present a congestion prevention mechanism on the basis of "TCP-Friendly Rate Control (TFRC) " and "Selective Acknowledgment (SACK)" systems based on entirely on the data transmission rate. The proposed design mechanism is a traffic management factor that adjusts the transmission rate of data transmission to the receiver to avoid data loss due to congestion or delay. This mechanism effectively reduces packet loss through its flow management factors and maintains a smooth flow of data, but does not keep the required packets delivered in time, resulting in reduced packet delivery with high latency.

Lee et al. [20] have been proposed to "Split multipath routing (SMR)" mechanism. It is a multi-routing protocol that detects multiple paths through a request/response sequence. SMR is looking for greater alternative routes to origin and destination. It exploits package flood system. SMR is a source routing protocol that contains information about the intermediate nodes that comprise the RREQ packet. SMR divides traffic loads into multiple routes offered to reduce loads on a particular route.

A "Multi-path source routing (MSR)" is proposed by L. Wang et al. [21]. It is an on-demand DSR modulation that utilizes influenced circular packet allocation of packets to develop packet transmission time and network throughput. In MSR, unlike the DSR protocol, multiple paths are detected during the path detection phase. The MSR protocol allocates the traffic load between multiple routes existing. MSR does not provide any QoS support for network traffic and there is no action to control routing overhead.

Tran et al. [22] proposed a "Congestion-adaptive routing protocol (CRP)", which primarily avoids congestion, as well as reactive congestion during data transmission. The CRP protocol builds a bypass path connecting the earlier node and the subsequent non-congested node. Each node that outlines an essential fraction of the route warns of its earlier node when congestion is expected to happen and thereby diminishes congestion. As CRP adapts to overcrowding, queue delays are reduced. The major disadvantage of this method is that packets are transmitted at a constant data rate. The protocol utilizes one congestion metric, which is currently the many packets in the buffer size, which is not enough to forecast congestion.

M. A. Gawas [9] has proposed a routing protocol based on "congestion-adaptive and delay-sensitive multi-rate (CADM) " for MANETs, consisting of exchange rate adaptation and conscious optimization of congestion to improve overall performance and delivery performance. CADM detects a fewer congested, high-output route utilizing the "QoS metrics", "packet delay", and "buffer queue delay". In this approach, every node uses interrelated information such as packet transfer delay, MAC layer, and network layer queue length as a congestion meter. But in order to guarantee these QoS requirements, it is necessary to optimize the intersection functionality, where based on the information in the lower layer, it is essential to improve the performance of the higher layer.

All of the above mechanisms have shown that the technology of the responders begins only after congestion has occurred. The proposed RDRA approaches offer a mechanism based on explicit routing delay (RD) notification instead of the traditional explicit loss notification to dynamically switch routes to minimize route delays. However, due to the interference, the packet loss is not in the scope of this paper.

III. PROPOSED ROUTING DELAY REDUCTION APPROACH

The proposed "Routing Delay Reduction (RDRA)" method applies two components: "Dynamic Switching Controller (DSC)" and "Path Congestion Prediction (PCP)" to reduce routing delay. DSC manages multipath routing activity, where the PCP calculates each congestion in the path to relieve congestion correctly. Typically, depending on the state of incoming traffic, the network path consists of different data packets. This affects transmission delay, loss rate, and productivity during the transmission phase. Inadequate receiver caching can eliminate some packets that have already been received and reduce the size of the transmission window. This illustrates the RDRP flow method is shown in Fig.1.

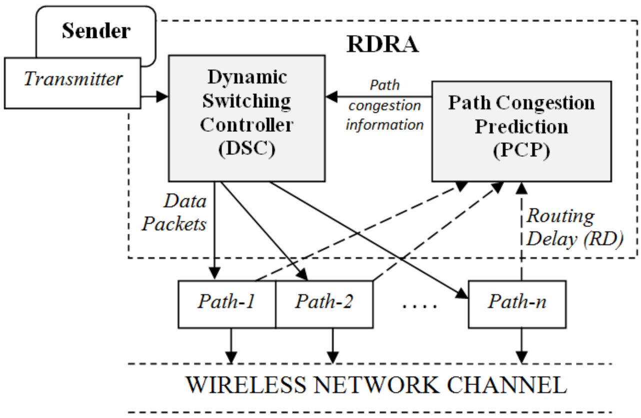

Fig.1: RDRA System Architecture

A. Dynamic Switching Controller (DSC)

The DSC function depends on the information provided by the PCP. The PCP provides three parameters for the DSC to decide the most appropriate path switch. The first parameter provides a quantitative estimate of the packet retention period before transmission as IN-OUT Delay as "IO_{Delay}", and the second parameter calculates the transmission delay as "TD_{Delay}", which measures the time interval for the packet to arrive at the queue to reach the queue header for transfer. The third parameter is the probability of delaying congestion as "CON_{Delay}". The DSC using this information parameter for each individual path calculates the probability of delaying switching the path as "PS_{Delay}" before setting the route for routing.

Initially, "IO_{Delay}" will be calculated for the path. Suppose that C is the maximum length of the data buffer, d is the number of data packets already in the buffer queue waiting to be transmitted, that is, $(d \le C)$, the packet rate is transmitted at T_{Rate}, and the packet rate arrived as A_{Rate}. These parameters are collected over a period of time by the PCP and submitted to the DSC to calculate the IO_{Delay} path as described in Eq. (1).

$$IO_{Delay} = \frac{(d-1)}{C} \times \left(\frac{A_{Rate}}{T_{Rate}} \right) \quad (1)$$

Second, the transmission delay, TD_{Delay} is calculated for a path that uses the transmission interval for a data packet. To calculate this, we periodically buy information about how many packets of data have been accessed and how many packets are transmitted over each specified time period tp, here tp =1 second.

According to the method specified in [23] for the standard transfer rate δ for one second, $\delta = \left(C \times \left(1/A_{Rate} \right) \right)$.

Therefore, based on this derivation, we can calculate TD_{Delay} as given in Eq. (2), the probability of delay switching path, PS_{Delay} is calculated using Eq. (3).

$$TD_{Delay} = \frac{1}{d} \times \left(\frac{T_{Rate}}{\delta} \right) \quad (2)$$

$$PS_{Delay} = \left(IO_{Delay} + TD_{Delay} \right) \quad (3)$$

According to the derived equation, the DSC can predict each individual path delay before switching to the appropriate transmission path. The lower the PS_{Delay} value, the higher the transmission efficiency. Let's consider that the buffer capacity, $C = 10$ and the standard transmission, $\delta = 10$ for the path. In this case, if A_{Rate}.= 1 and T_{Rate},=1 for $d = 1$, its value of $PS_{Delay} = 0.1$ as the lowest and the highest will be 1 based on Eq. (3). Therefore, based on the set of PS_{Delay} values, DSC switches the path required for better efficiency accordingly.

B. Path Congestion Prediction (PCP)

The effective scheduling support assigns packets that initially arrive at a valid path, but the packet data transfer rate must be controlled based on two factors, such as "packet delivery rate, PD_{Rate}" and "Queue availability, Q_{Avail}". This transmission control is able to avoid congestion and reduces packet loss and latency.

In order to utilize PCP, the proposed mechanism monitors these two factors and predicts the probability of congestion in the possible paths and controls the transmission rate. The sender receives a Route Delay Acknowledgement (RD_{ACK}) from the PCP to control the packet transmission rate to three different transmission modes, namely as "Minimum Transmission (MIN-TR)", "Maximum Transmission (MAX-TR)" and " Average transmission (AVG-TR). The Q_{Avail} calculation is the ratio of the maximum size of the data saves buffer to the number of packets waiting to be sent in the buffer queue.

$$Q_{Avail} = \frac{\text{Maximum buffer length}}{\text{No.of data packets in the buffer Queue}} \quad (4)$$

Initially, the transmission starts at *MIN_TR* and then proceeds to *MAX-TR*. If the threshold notification about queue availability Q_{Avail} is reduced to < 60 % and > 50 %, the transfer rate will be reduced to *AVG-TR*, if it still drops below 50%, and then to *MIN-TR*. If B_{Avail} > 60%, the transfer rate is restored to *MAX-TR*, as shown in Table-1.

Table-1: Rate of Transmission flow Vs Q_{Avail}

Q_{Avail}	Type of Rate of Transmission
>50%	MIN-TR
>=50% - <60%	AVG-TR
>60%	MAX-TR

In addition to Q_{Avail}, the PCP calculates the packet transmission rate PD_{Rate} derived from the ratio of the number of packets transmitted during a one-time interval to the number of transmitted packets. Based on the change of the PD_{Rate} threshold, the PCP proposes a change in the transmission rate T_{Rate} according to the "Additive increase and Multiplicative Decrease (AIMD)" [24] method, which can be expressed as:

$$T_{Rate} = \begin{cases} TRate + \alpha, & \text{if congestion} < \text{threshold} \\ TRate \times \beta, & \text{if congestion} > \text{threshold} \end{cases}$$

where, α is an additive increase parameter, and β is a multiplicative decrease parameter. If the PD_{Rate} threshold is <0.5, the transmission rate will be increased by an additional alpha value, and if the PD_{Rate} threshold is > 0.5, the product of the reduced beta value is reduced to regulate congestion when the RD_{ACK} acknowledgment message is received.

Based on the PS_{Delay} value calculated above, the proposed RDRA will effectively switch incoming new packet arrivals and control congestion and reduce routing delays by controlling the transmission rate based on Q_{Avail} and T_{Rate}.

IV. EXPERIMENT EVALUATION

A. Simulation Setup

To perform the evaluation, it configured the simulation parameters in a network simulator called "GloMoSim". It supports the creation of a must-have wireless situation with the "MAC 802.11" standard.

We configured the essential parameters required to simulate the proposed RDRA. To measure performance, it assumed two situations, depending on "Mobility" and "Traffic intervals". The configured parameters for both cases are listed in Table 3.

Table-2: Simulation Parameter

Configuration	Parameter Values
Simulation Area	1000m X 1000m
No. of Nodes	50
Packet Size	512 bytes
Mobility	RWP
Pause Time (sec)	5
No. of Senders	20
No. of Receivers	20
Case-1:	*(Varying Mobility Speed)*
Mobility (m/s)	10, 20, 30, 40, 50
CBR Rates	4 pkts/sec
Case-2:	*(Varying Traffic Interval)*
Traffic Interval (milliseconds)	1000, 800, 600, 400, 200
Mobility (m/s)	30

In the case of Case 1, it changed the movement speed from "0 to 50 m/s" and CBR rate is 4pkt/s, in Case 2, the network traffic interval was changed from 1000 to 200 milliseconds to increase packet delivery to improve the traffic rate with a moving speed of 30 m / s. Each simulation was run for 600 seconds in a "Random Waypoint Model (RWP)" with zero pause time.

The mobility of a node is an activity determined in the MANET situation, which can seriously affect node-link be unsuccessful, and increased traffic can cause congestion in any routing protocol. We evaluated the variability of the proposal to analyze the stability of the network and its throughput performance and to change the traffic rate to analyze the processing power and meet the throughput requirements.

The performance of the comparison between RDRA and CADM [9] is derived from four popular metrics: "Packet Transfer Rate (PDR)", "Delay", "Packet Loss", and "Control Overhead". The results acquired are estimated in the subsequent sections.

B. Result Evaluation

- ***Packet Delivery Ratio***

The calculation of the packet transfer ratio (PDR) depends on the ratio of the "total number of data packets originated" to the "total number of data packets transmitted". Figure 2 shows a comparison of PDR between RDRA and CADM. The impact of the change in mobility on both is similar, that is, due to its continuous monitoring and dynamic routing, both show PDR decline in the average gap difference, but RDRA is excellent at a constant bit rate of 4pkts / sec. For CADM it shows an average PDR of 10% higher than a CADM with a change in mobility.

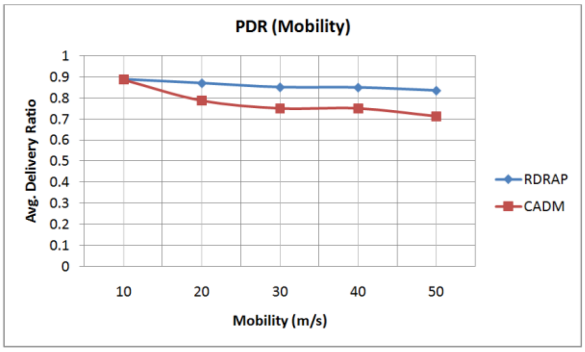

Fig. 2: PDR Result at different Mobility

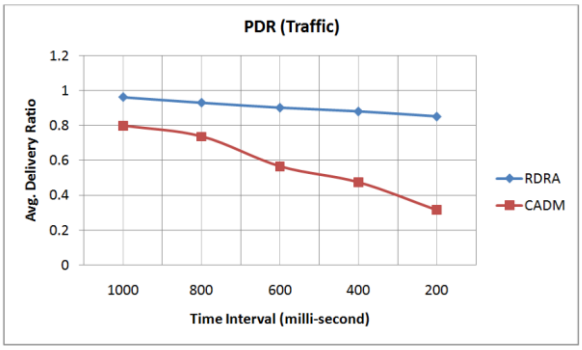

Fig. 3: PDR Result at different Traffic Interval

Fig.3 shows a PDR comparison between RDRA and CADM at different flow intervals. The proposed RDRA showed superior performance over CADM over extended time intervals. Although CADM has a congestion control mechanism, the configuration of multi-stream transmission by nodes reasons high congestion of intermediate nodes to cause a large loss of data packets, while RDRA maintains good throughput due to parallel flow and dynamic routing with congestion prediction. The average change. In low traffic, PDR is 20% higher, and in high traffic intervals, PDR is 70% higher.

- ***Delay Comparison***

The Delay is an important metric in multi-streaming that depends on the latency in the packet flow between source and destination. Fig. 4 shows the delay comparison between RDRA and CADM for different mobility. In both protocols, latency increases with mobility, but CADM can greatly affect individual routing dependencies and link failures that occur frequently as mobility increases. Even with RDRA, the delay from the low mobility to the maximum delay is from 0.016 to 0.029 seconds with a 14% change, while CADM shows a delay of 0.042 to 0.073 sec with a 31% change. Overall, RDRA showed an average 25% lower latency than different mobility comparisons.

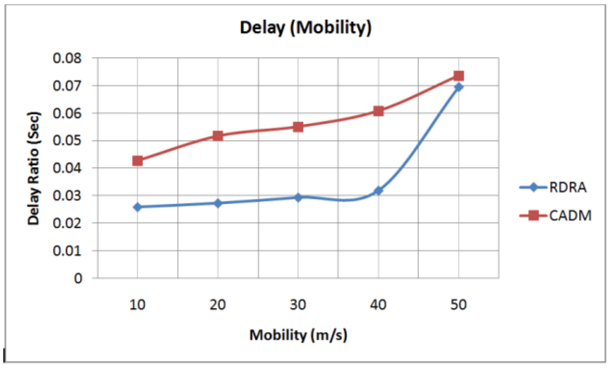

Fig. 4: Delay Comparison Result at different Mobility

Fig.5: Delay Comparison Result at different Traffic Interval

Fig. 5 also shows the comparison of delays between RDRA and CADM at different time intervals. The results show that as the communication interval changes, the delay increases. Since both have congestion control mechanisms to manage congestion, the average difference can be maintained. However, RDRA shows lower latency due to low-crowded node selection at runtime, but in some cases, it may get longer routes and observe more delays. Even RDRA has an average latency of 15% lower than CADM.

- ***Packet Loss***

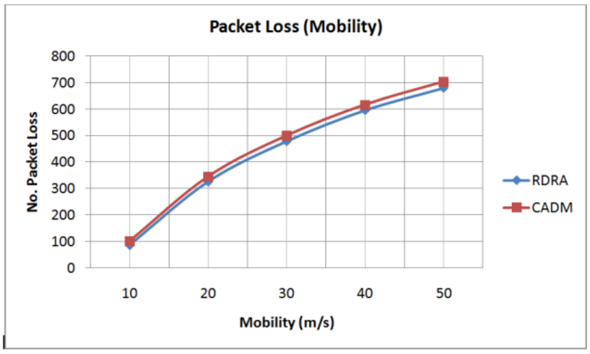

Fig. 6: Packet Loss at different Mobility

Packet loss in the network can affect throughput results, which can be the result of "denial of service", "link failure" or "high congestion." It depends on the sum of the "number of dropped packets". Figure 6 shows the packet loss comparison for different mobility. Compared with RDRA, the loss of CADM increases exponentially. Increased mobility leads to higher link failures and more packet loss. Multiple attempts by CADM in similar routes can result in a large amount of packet loss. In the case of RDRA, dynamic selection of nodes helps. Minimize this loss and help increase efficiency throughput.

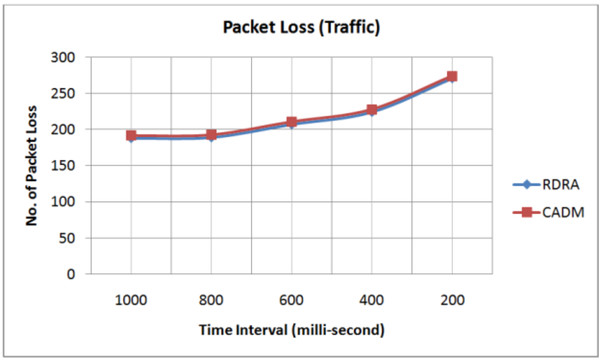

Fig. 7: Packet Loss at different Traffic Interval

Fig. 7 shows the loss comparison of RDRA and CADM at different flow intervals. In this case, RDRA shows that the packet loss rate increases as the flow interval increases, but it shows a lower level than CADM. The increase in loss is due to the fact that the node buffer limits the maintenance of queued packets. CADM keeps packet loss by reducing the transmission rate, which causes a linear increase in loss, but RDRA instead of reducing it continues to send in the alternate route, which leads to higher losses. However, the observed losses can be considered affordable because doing so can better achieve PDR and lower latency.

- ***Control Overhead***

The control overhead is computed utilizing the summation of the number of control packets used during the communication cycle. High overhead makes the network unstable. Fig. 8 shows a comparison of the control overhead of RDRA and CADM under different mobility. Network overhead increases due to high packet loss and latency. As mentioned above, we explain that CADM has higher latency and packet loss than RDRA, which results in higher overhead than RDRA.

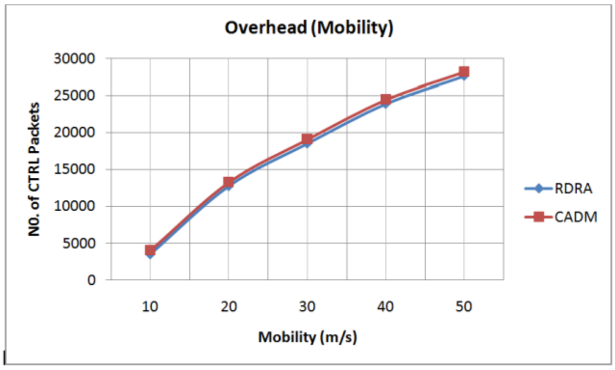

Fig. 8: Control overhead comparison at different Mobility

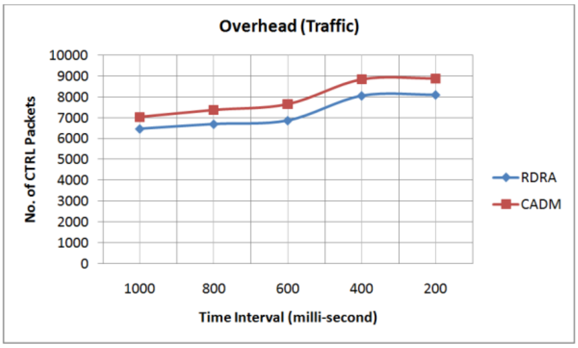

Fig.9: Control overhead comparison different Traffic Interval

Fig.9 shows the comparison of control overhead for RDRA and CADM at different traffic intervals. In Fig. 7 and 9 above, CADM has greater latency and packet loss than RDRA. However, RDRA shows the raise in packet loss over period in Fig. 7. In the case of RDRA, this results in additional overhead and increases the rate of traffic intervals, but it shows the least overhead compared to CADM. All the above comparison results show that RDRA is superior to CADM in all aspects, but the packet loss rate is negligible in the case of high traffic intervals. This demonstrates the stability and better throughput for a multi-stream TCP-based congestion protocol based on MANET.

V. CONCLUSION

In wireless mobile networks, packets are mainly lost due to congestion caused by buffer overflow and high transmission delay between connection paths. Routing protocols flood the network to discover routes that congest the network, regardless of the state of traffic on the network. Additional traffic can make the network more congested and increase communication latency and high packet loss. In this paper, we propose a Routing Delay Reduction Approach (RDRA) to reduce routing delay and improve PDR through the dynamic exchange of congestion prediction. It proposes a dynamic switching controller for guiding the selection of an appropriate path based on the routing delay acknowledgment provided by the path congestion prediction controller.

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Extensive simulations at different mobility and traffic interval rates to measure packet transmission ratios, delays, packet loss, and control overhead, which show impromptu performance of PDR while reducing routing latency, control overhead, and Low packet loss.

REFERENCES

1. L. Zhang, C. Wan, "Dynamic Path Planning Design for Mobile Sink with Burst Traffic in a Region of WSN", *Wireless Communications and Mobile Computing(Hindawi)*, pp. 8, 2019.
2. P. Aimtongkham, T. G. Nguyen, C. So-In, "Congestion Control and Prediction Schemes Using Fuzzy Logic System with Adaptive Membership Function in Wireless Sensor Networks", *Wireless Communications and Mobile Computing(Hindawi)*, pp. 19, 2018.
3. R. Melki, M. M. Mansour, A. Chehab, "A fairness-based congestion control algorithm for multipath TCP", *IEEE Wireless Communications and Networking Conference (WCNC)*, pp. 1 - 6, 2018.
4. S. C. Nguyen, X. Zhang, T.M.T. Nguyen, G. Pujolle, "Evaluation of Throughput Optimization and Load Sharing of Multipath TCP in Heterogeneous Networks", *In Proc. Eighth Int. Conf. Wireless and Optical Communications Networks*, pp. 1-5, 2011.
5. J. Wang, M.Song, "Rate-Based Active Queue Management for TCP Flows over Wired and Wireless Networks", *EURASIP Journal on Wireless Communications and Networking*, Vol. 2007, pp. 8, 2007.
6. M. Ikeda, E. Kulla, M. Hiyama, M. Takizawa, "Congestion Control for Multi-flow Traffic in Wireless Mobile Ad-hoc Networks", *IEEE Sixth International Conference on Complex, Intelligent, and Software Intensive Systems*, pp. 290 - 297, 2012.
7. F. Ren, T. He, S. K. Das, and Chuang Lin, "Traffic-Aware Dynamic Routing to Alleviate Congestion in Wireless Sensor Networks", *IEEE Transactions On Parallel And Distributed Systems*, Vol. 22, No. 9, September 2011.
8. A. K. Dwivedi and G. Prajapati, "Study of Congestion Control in Multi-Flow in MANET", *International Journal of Computer Applications (0975 - 8887)*, Vol. 122 - No.14, July 2015.
9. M. A. Gawas, L. J. Gudino, K. R Anupama, "Congestion-Adaptive and Delay-Sensitive Multirate Routing Protocol in MANETs: A Cross-Layer Approach", *Journal of Computer Networks and Communications(Hindawi)*, pp. 13, 2019.
10. M. Shafiee, J.Ghaderi, "A Simple Congestion-Aware Algorithm for Load Balancing in Datacenter Networks", *IEEE/ACM Transactions on Networking*, Vol. 25(6), pp. 3670 - 3682, 2017.
11. W. Song, S. Jiang, and G.Wei, "A Congestion-Aware Multipath Routing with Cross-Layer Design for Wireless Mesh Networks", *Proceedings of the 15th Asia- Pacific Conference on Communications*,157, 2009.
12. M. F. Tsai, N. Chilamkurti, J. H. Park, C. K. Shieh, "Multi-path transmission control scheme combining bandwidth aggregation and packet scheduling for real-time streaming in the multi-path environment", IET Communications, vol. 4, no. 8, pp. 937-945, 2010.
13. K. C. Rahman, Syed Faisal Hasan, "Explicit Rate-based Congestion Control for Multimedia Streaming over Mobile Ad hoc Networks", *International Journal of Electrical & Computer Sciences IJECS-IJENS*, Vol. 10, No. 04, 2010.
14. M. Ikeda, E. Kulla, M. Hiyama, L. Barolli, M. Younas, and M. Takizawa, "TCP Congestion Control in MANETs for Multiple Traffic Considering Proactive and Reactive Routing Protocols", *15th International Conference on Network-Based Information Systems*, 2012.
15. C. -Ya Huang, P. Ramanathan, "Network Layer Support for Gigabit TCP Flows in Wireless Mesh Networks", *IEEE Trans. on Mobile Computing*, Vol. 14(10), pp. 2073 - 2085, 2015.
16. A. Darehshoorzadeh, R. E. De Grande, A. Boukerche, "Toward a Comprehensive Model for Performance Analysis of Opportunistic Routing in Wireless Mesh Networks", *IEEE Trans. on Vehicular Technology*, Vol. 65(7), Pp. 5424 - 5438, 2016.
17. H. Shen, L. Cai, and X. Shen, "Performance analysis of TFRC over a wireless link with truncated link-level ARQ", *IEEE Transactions on Wireless Communications*, vol. 5, no. 6, June 2006.
18. A. A. El-Sherif, A. Mohamed, "Joint Routing and Resource Allocation for Delay Minimization in Cognitive Radio Based Mesh Networks", *IEEE Trans. on Wireless Comm.*, Vol. 13(1), Pgs. 186 - 197, 2014.
19. E. Lochin, G. Jourjon, S. Ardon P. Senac, "Promoting the Use of Reliable Rate Based Transport Protocols: The Chameleon Protocol", *International Journal of Internet Protocol Technology*, Vol 5, Issue 4, pp 175-189, March 2010.
20. S. J. Lee and M. Gerla, "Split multipath routing with maximally disjoint paths in ad hoc networks", *IEEE International Conference on Communications*, Vol. 10, pp. 3201– 3205, 2001.
21. L. Wang, Y. Shu, M. Dong, L. Zhang, and O. W. W. Yang, "Adaptive multipath source routing in ad hoc networks", *In IEEE International Conference on Communications*, vol. 3, pp. 867–871, 2001.
22. D. A. Tran and H. Raghavendra, "Congestion adaptive routing in mobile ad hoc networks", *IEEE Transactions on Parallel and Distributed Systems*, vol. 17, no. 11, pp. 1294–1305, 2006.
23. J. D. C. Little, S.C. Graves, "Little's Law", *In book: Building Intuition*, DOI: 10.1007/ 978-0-387-73699-0, pp.81-100, 2008.
24. C. M. Lagoa, H. Che, B.A. Movsichoff, "Adaptive control algorithms for decentralized optimal traffic engineering on the internet", *IEEE/ACM Trans. Networking*, Vol. 12 (3), pp. 415-428, 2004.
25. C. Wan, J. Chang, and L. Zhang, "Dynamic path planning design of mobile sink for single region bursty traffic", *Computer Science*, Vol. 44(5), pp. 61–65, 2017.

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