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Research Article

Congestion-Adaptive and Delay-Sensitive Multirate Routing Protocol in MANETs: A Cross-Layer Approach

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With a growing demand of multimedia communication over MANETs, to support quality of service (QoS), the MAC standards such as 802.11a/b/g operate with multiple data rates to efficiently utilize the limited resources. Since the wireless channel is shared among the neighbors in MANETs, determining delay-sensitive and congestion-aware routes using the IEEE 802.11 MAC is still a challenging problem. This paper proposes a novel cross-layer approach called congestion-adaptive and delay-sensitive multirate (CADM) routing protocol in MANETs. The CADM protocol exploits the cross-layer interaction between the network layer, MAC, and physical layer. The CADM accesses the correlation between data rate, congestion metric, and MAC delay in delay-sensitive applications to provide enhanced network efficiency in MANETs. The protocol discovers multiple node-disjoint routes and facilitates optimal data rates between the links based on the estimated delay to admit a flow with the certain delay requirement in multirate MANETs. The proposed CADM protocol discovers the route through less congested nodes and also actively handles the congestion if it occurs. The performance of the CADM protocol is comprehensively assessed through the simulation, which highlights the advantages of our cross-layer mechanism.

1. Introduction

In recent times, there has been an immense growth in demand for support of multimedia applications over MANETs. Most of the real-time multimedia traffic tends to be in bursts and bandwidth demanding and is responsible for the congestion. The congestion factor results in packet losses, retransmissions, and bandwidth degradation and also incurs additional time and energy on to recover from congestion [1]. For real-time traffic, the data rate and delay are the crucial QoS factors. Therefore, to satisfy these QoS requirements, each route in the network should provide a correct estimate of the available data rate and delay.

The main design goal of the network design is to allocate network resources effectively and fairly among the nodes in the network. The most commonly shared resources are bandwidth and the node queues, which are limited in capacity. While the node is contending for the channel access, packets get buffered in these nodes queues. In a high traffic flow network, the multiple neighbor nodes contend for the channel. This leads to collision and drop in data packets. When the packet drop ratio crosses the threshold limit, the network is identified to be congested [2]. In traditional ad hoc routing protocols, packet losses are assumed as a consequence of congestion. On the contrary, in MANETs, the routing protocols assume the packet losses are due to link failure and initiate the reroute discovery process to find the alternate route to the destination [3]. But in MANETs, the packets loses could be either due to link failure or congestion [4]. In congested networks, performing route discovery may not only be unnecessary, but it may also further increase congestion. Due to limited resources of energy and bandwidth available in MANETs, it is necessary to initiate the congestion control mechanism to improve the network performance.

Wireless standard IEEE 802.11a/b/g supports rate adaptation to accommodate time-varying channels [5]. More often, the IEEE 802.11 DCF uses low data-rate next hop node for transmission even under heavy traffic conditions and thereby decreases the throughput. In multirate networks, if lower data-rate link is followed by the higher data-rate link, packets will build up at the node heading the lower data-rate link, leading to long queuing delays. This further leads to packet drop and causes congestion due to packet retransmission. This is more evident in large traffic of intensive data such as multimedia and has a negative influence on the QoS. Unlike the well-established networks such as Internet, it is very expensive in terms of time and energy to overcome in MANETs [6].

The past researchers have done the research survey that, due to the rapid changes in the channel conditions for MANETs, substantial time and energy is consumed on retransmissions only [7]. Thus, the existing rate adaptation algorithms must go through further development to reduce packet losses and retransmissions and optimally utilize network resources.

Thus, the probable solution in MANETs could be to design a routing protocol, which supports data transmission at a higher data rate to improve the overall network throughput. Also, to provide an acceptable level of QoS to delay-sensitive application, the traffic is routed through a less congested network. The solution should provide a mechanism to overcome the congestion effectively and efficiently with minimum overhead.

Different applications have different QoS requirements. To guarantee these QoS requirements, optimization of cross-layer functionality, where higher layer functioning is improved based on the information available at the lower layer, is necessary. In this paper, a distributed congestion-adaptive and delay-sensitive multirate (CADM) routing protocol in MANETs is proposed, which consists of rate adaptation and congestion-aware optimization to improve the overall performance in terms of throughput, packet delivery, and latency for the MANETs. The CADM discovers a less congested, high throughput route based on the QoS metrics data rate, packet-forwarding delay, and buffer queuing delay. In this approach, each node takes advantage of sharing the interlayer information, such as packet-forwarding delay, from the MAC layer, and the queue length from the network layer as the congestion metric.

2. Literature Survey

This subsection presents the detailed analysis of the existing congestion-aware routing mechanisms and discusses their limitations when applied in MANETs.

2.1. Congestion-Aware Schemes. Chen et al. proposed congestion-aware routing protocol for MANETs (CARM) [8], which is a modification of DSR protocol. It collects the congestion information from the MAC layer to discover congestion-free routes. The protocol uses a weighted channel delay (WCD) metric to assign the cost to each of the link in route based on MAC overhead and packet queuing delay. The WCD metric measures the congestion level and adopts effective link data-rate category (ELDC) to avoid the mismatch data-rate route (MDRR) problem. However, the CARM does not reactively deal with congestion during data transmission.

Tripathi et al. proposed a congestion-aware distance vector (CADV) [9] protocol based on proactive protocol DSDV, which maintains the route information of all the nodes in the network. The routing decision is made based on the hop count to the destination node and the estimated delay which is a measure of congestion at the next hop. The CADV gives higher priority to the route with low estimated delay. However, estimated delay is a weaker metric to measure the congestion condition. Network scalability is limited due to excessive control overhead. Also, CADV is not feasible in MANETs where frequent topological changes are inevitable and incurs high overhead for maintaining the routing table updates.

Song et al. proposed delay-based load aware on-demand routing (D-LAOR) [10] protocol, which discovers route based on estimated route delay and hop count. It is an extension of the AODV protocol. In the route discovery process, RREQ packets are dropped at the congested node, preventing the congested node being the part of a route. D-LAOR protocols conclude the node to be congested by comparing the estimated total node delay and the number of packets that are being queued in the interface of a node in an RREQ packet-forwarding path. In D-LAOR, the data rate for packet transmission is assumed to be static and, moreover, does not provide a solution to overcome the congestion.

Tran et al. [11] proposed a congestion-adaptive routing protocol (CRP) that prevents the congestion at the first place and also reactively deals with congestion during data transmission. The novelty in the CRP design is the bypass concept. In the CRP protocol, a bypass route is constructed by connecting a previous node and the next noncongested node. Every node that forms the integral part of the route cautions its previous node whenever the congestion is likely to occur and thus minimizes the traffic coming to the potentially congested node. Since CRP adapts to the congestion, the queuing delay is minimized. The main drawback of this mechanism is packets are transmitted at a static data rate. The protocol uses a single congestion metric, which is the ratio between the number of packets currently buffered to the buffer size which is not sufficient to predict the congestion.

The above protocols are congestion-aware-based protocols which discover the routes that are less congested. These protocols do not suggest a mechanism to adapt to the congestion if takes place. In the above schemes, the congestion is taken into consideration only when establishing a new route and remains the same until node mobility or link failure results in route disconnection. Moreover, these protocols use the static base data rate for data communication.

2.2. Rate Adaptation Schemes. IEEE 802.11a/b/g supports multirate capability which has been studied at length by several researchers to exploit the possibility of improving the network performance. Many researchers used the rate adaptation technique which uses the variable data rate, to cope up with the deteriorating channel condition [12, 13].

The automatic rate fallback (ARF) [14] is a simplest and widely used sender-based rate adaptation scheme. In ARF, the node lowers the data rate if two consecutive ACKs are not correctly received by the sender node and starts a timer. If a source node receives 10 consecutive successful ACKs, then the next data transmission takes place at the next higher rate, and the timer is set to zero. The main drawback of ARF is that the throughput decreases because source node decreases its rate even though transmission failures are caused by collisions. Moreover, the ARF is not designed to get adapted to MANETs characteristics.

Kim et al. proposed the collision-aware rate adaptation (CARA) algorithm [15]. Unlike ARF, CARA discriminates between frame collisions and transmission failures caused by the channel error. It uses the RTS/CTS mechanism to estimate the quality of the channel. The sender node decrements its data rate based on the cause of the transmission failure, i.e., only if there are consecutive channel errors, but not in case of collisions. When the number of transmission failures reaches a certain threshold value, the RTS/CTS mechanism is enabled. The CARAs performance degrades due to RTS fluctuation when a hidden terminal exists around. The CARA performs better in terms of throughput than ARF. However, the CARA scheme is well suitable for WLANs yet not suitable for dynamics in MANETs and does not support system fairness.

Xi et al. proposed the adaptive multirate auto rate fallback (AMARF) scheme for IEEE 802.11 WLANs [16]. In AMARF, each data rate is assigned a unique success threshold, which is a criterion to switch from one data rate to the next higher data rate. The success threshold is dynamically changed in an adaptive manner according to the current network conditions, such as packet length and channel parameters. The AMARF protocol is implemented without any modification to the existing IEEE 802.11 standards. Although AMARF outperforms the ARF scheme, it does not take into account the competing nodes in the MANETs context.

However, so far not much effort has been made to utilize this multirate enhancement in mobile ad hoc networks. Many of these existing proposed rate adaptation protocols focus on the current channel state, overlooking the congestion impact on overall network performance. It is vital to have detailed, real, and precise metrics made available for understanding of the traffic and channel models.

2.3. Multipath Routing Schemes. Some on-demand routing protocols based on multipath path selection are discussed below. They are categorized as a variant of AODV and DSR.

Marina et al. proposed AOMDV [17] as an extended version to the AODV protocol for computing multiple loop-free and link-disjoint routes. The loop-free route is assured by using a mechanism of "advertised hop count." The link disjointness of multiple routes is attained by using a flooding mechanism. AOMDV improves fault tolerance by selecting disjoint paths. AODVM [18] is a modification of the AODV protocol to find multiple node-disjoint routes. AODVM do not consider any measures to control the routing overhead due to packet flooding.

Reddeppa Reddy et al. proposed a variant of the AODV protocol called the scalable multipath on-demand routing protocol (SMORT) [19]. The main objective of the SMORT protocol is to minimize the routing overhead. It implements the fail-safe multiple path mechanism instead of the node-disjoint path. In SMORT, the source initiates the route discovery by flooding the RREQ packets in the network. The intermediate node replies to RREQ, if it has a path entry in the route table to reach the destination; otherwise, it forwards the RREQ packets to its neighbors.

Split multipath routing (SMR) [20] is a multipath source routing protocol which discovers multiple routes using the request/reply cycle. SMR finds an alternative route that is maximally disjoint from the source to the destination. It uses the packet flooding mechanism. SMR is a source initiated routing protocol wherein the RREQ packet includes the information about the intermediate nodes which forms the route. SMR splits the traffic load in multiple routes available to reduce a load on a single path.

Multipath source routing (MSR) [21] is a modification of on-demand DSR protocol which uses weighted round robin packet distribution to improve the packet delay and network throughput. In MSR, the route discovery phase discovers multiple paths, unlike the DSR protocol. MSR protocol distributes the traffic load among the multiple available routes. MSR does not provide any QoS support for the traffic in the network and does not have any measures to control the routing overhead.

Although all of these above discussed protocols find the multiple routes between source and the destination, the broadcast storming of routing packets is overlooked in the process of route discovery and maintenance. Most of the above protocols do not discard the duplicate RREQ packets at the intermediate nodes. Above protocols does not provide any solution to control the routing overhead. Moreover, these protocols do not provide any support for QoS provisioning. The major challenge that has not been addressed in the above protocols is utilizing minimum bandwidth and network overhead to avoid congestion in the network and enhance the network performance.

3. Proposed Cross-Layer Architecture Design

Different applications have different QoS requirements. To guarantee these QoS requirements, optimization of cross-layer functionalities where higher layer functioning is improved based on the information available at the lower layer is necessary. In this section, congestion-adaptive and delay-sensitive multirate (CADM) routing protocol for MANETs is proposed. CADM consists of rate adaptation and congestion-aware optimization to improve the overall performance in terms of throughput, packet delivery, and latency for the MANETs presented. The CADM discovers a less congested, high throughput route based on the QoS metrics data rate, packet-forwarding delay, and buffer queuing delay. In this approach, each node takes advantage of sharing the interlayer information, such as packet-forwarding delay from the MAC layer and the queue length from the network layer as the congestion metric. Figure 1 demonstrates the proposed cross-layer architecture design in CADM.

In general, the existing protocols do not consider the delay incurred due to packet collision while calculating packet-forwarding delay which is a crucial factor in determining the backoff factor and becomes more significant when the network is large. Hence, the proposed protocol focuses on packet-forwarding delay which includes MAC delay, queuing delay, transmission delay, and delay due to packet collision while discovering the congestion-free route for delay-sensitive application.

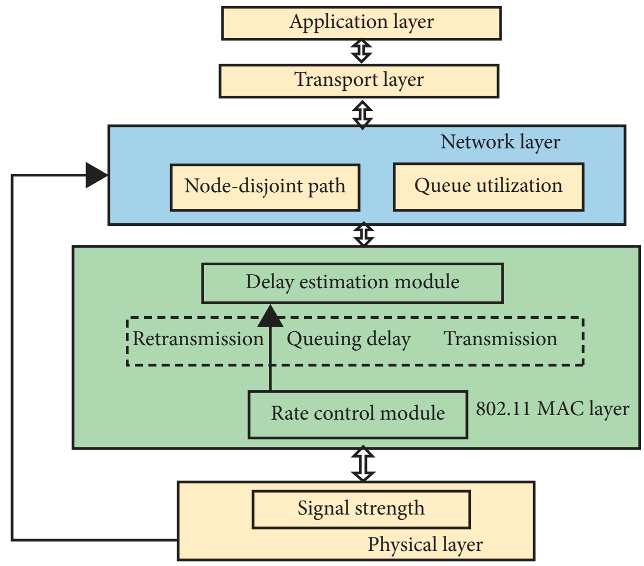

Figure 1: Cross-layer architecture design proposed in CADM.

3.1. Issue with Mismatched Link Data-Rate Route. The underlying 802.11 physical layer in a network has a multi-data-rate transmission capability. Since there are no specific rules mentioned to select the data rate, based on the underlying channel condition, links choose different data rates for the given path. Consider an example network shown in Figure 2 with path S-A-B-D where *S* and *D* are the source and destination node, respectively. The links between *S* to *D* have adapted to different data rates based on underlying channel conditions. For example, link S-A has chosen a rate of 11 Mbps followed by a link A-B with a rate of 2 Mbps. This may result in packet accumulation at a node *B* buffer since the data inflow is at a higher rate compared to data outflow, creating possibilities of a bottleneck. Therefore, if lower data-rate links is followed by the higher data-rate links, this may lead to long queuing delays at nodes queue and subsequently may cause in packet loss. The increase in packet losses may lead to more congestion because of packet retransmissions, which may further degrade the network performance with extended end-to-end delay.

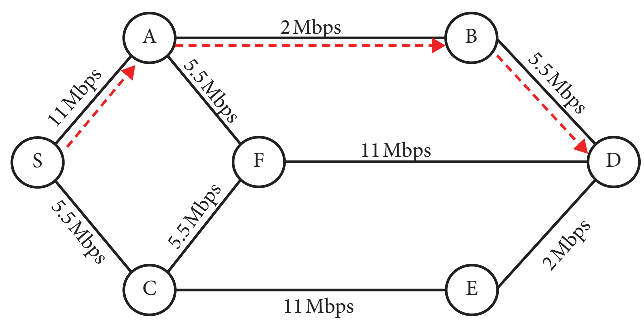

Figure 2: Mismatch link data rate.

3.2. Estimating Queue Utilization. The queue load of a node indicates the number of packets in the queue at any given time *t*. At the sender side, when congestion occurs, the queue load increases, and when it crosses the threshold, sender should quickly send out the packets by increasing the data rate. But at the receiver side, data rate cannot be increased to support the data rate increase at the sender if the queue load of the receiver is already at the threshold level. Therefore, to offer a congestion control, considering queue load only at the sender is not enough. We must balance the data rate between the sender and receiver by knowing the queue loads at both sender and receiver to avoid high level congestion. Hence, the CADM approach uses the adaptive feedback mechanism that helps to select data rate based on queue loads of both sender and receiver.

The metric, average queue load indicates the prolonged congestion in the network due to traffic variation. The average estimated queue load of the node over time interval Δt is computed according to the following formula:

$$Q_{\text{avg load}}(t) = \delta \times Q_{\text{current load}}(t) + (1-\delta) \times Q_{\text{avg load}}(t-1), \quad (1)$$

where $Q_{\text{avg load}}(t)$ denotes the estimated average queue load at time t, $Q_{\text{current load}}(t)$ denotes the current queue load, and δ can be any number selected from the range [0, 1].

δ is considered as a weighting parameter to regulate network congestion. If δ is chosen to be too small, then it would not imitate the prolonged network congestion. If δ chosen is too large, the average queue length follows the current queue length, which also worsens the congestion estimation technique. This may result in metric chosen to be less effective. In the proposed CADM protocol, the congestion status of a node is determined by the queue utilization metric. Let Q_{size} be the size of the buffer in a node. The average queue utilization Q_{util} for node N_i is calculated as follows:

$$Q_{\text{util}} = \frac{Q_{\text{avg load}}(t)}{Q_{\text{size}}}. \quad (2)$$

3.3. Congestion Evaluation. In the proposed CADM protocol, the congestion status is indicated by three levels, i.e., forward level, attentive level, and drop level. The threshold value of queue utilization of a node, i.e., Q_{Thresh}, is defined in the range of 80–85% of Q_{size}. When the average queue utilization status is less than the threshold value, i.e., $Q_{\text{util}} < Q_{\text{Thresh}}$, the congestion status of a node is indicated as the forward level, and the packets are forwarded to the next node. When $Q_{\text{util}} = Q_{\text{Thresh}}$, the congestion status of a node is indicated as the alert level, and the queue load balancing procedure is invoked. When $Q_{\text{util}} > Q_{\text{Thresh}}$, the congestion status of a node is indicated as the drop level, and the packets are dropped as an indication of the high congestion level.

3.4. Packet-Forwarding Delay Estimation. The IEEE 802.11 protocol uses (CSMA/CA) the MAC protocol. It uses distributed coordination function (DCF) as the fundamental mechanism to access the channel. DCF uses the RTS/CTS scheme to eliminate collision and resolves hidden stations problem, as shown in Figure 3. To support asynchronous data communication, the DCF uses two kinds of frame spaces: DCF interframe spaces (DIFS) and short interframe spaces (SIFS). Each node wanting to transmit the data packets senses the channel to be idle for at least DIFS time interval, while SIFS is used to guarantee the higher priority for control packets ahead of data packets. Hence, the SIFS interval is smaller than the DIFS interval.

Whenever an i^{th} mobile node N_i wants to send packets either generated by itself or received from neighbor nodes to node N_{i+1}, it senses channel for the DIFS period. If the channel remains idle for a DIFS period, then node N_i delays the transmission for the time duration during which random backoff time counter reaches zero to avoid any attempt by other nodes. The random backoff time counter is decreased each slot time if the medium remains idle. Slot time is the time unit in the backoff process. If during the backoff process, the medium becomes busy, the backoff counter is paused and resumed only when the medium becomes idle. Let $P_{\text{idle}}(t)$ be the probability that no other nodes are transmitting data, and the channel is sensed idle. The probability that channel is busy and node N_i enters the backoff state is given as $1 - P_{\text{idle}}(\text{DIFS})$ and consumes a delay of $\text{DIFS} + \text{Bf}$, where the Bf will incur additional delay of $\text{RTS} + \text{SIFS} + \text{CTS} + \text{SIFS} + \text{DATA} + \text{SIFS} + \text{ACK}$ before its next attempt.

When backoff counter reaches zero, the node N_i attempts to transmit the control packet RTS to the next node N_{i+1} and waits for SIFS time to receive the CTS packet from the next node. The probability that node N_i senses the channel idle is given as $P_{\text{idle}}(\text{DIFS})$. The node N_i enters again into the backoff state, if it fails to receive CTS during time slot. The probability that the node N_i receives CTS is given as $P_{\text{idle}}(\text{slot})$ and incurs delay of $\text{RTS} + \text{SIFS} + \text{CTS} + \text{SIFS}$ time period to make sure the channel reservation is success. The probability of failing to receive CTS is given as $1 - P_{\text{idle}}(\text{slot})$ and incurs delay of $\text{RTS} + 2 \times \text{SIFS}$. The packet-forwarding delay at node N_i, which comprises MAC contention and transmission delay, is calculated using the following equation [22]:

$$D^i_{\text{delay}} = P^i_{\text{idle}}(\text{DIFS}) * \left(\text{DIFS} + \text{avg}_{\text{bt}} + \text{DA}(i)\right. \\ \left. + \left(1 - P^i_{\text{idle}}(\text{DIFS})\right)\right) * \left(\text{SIFS} + \text{DB}(i) + \left(\frac{L}{R}\right)\right), \tag{3}$$

where $P^i_{\text{idle}}(t)$ is the probability that a given node N_i will not encounter any other neighbor node transmitting data during time interval t, avg_{bt} is a mean random backoff time interval before transmission, $\text{DA}(i)$ is the predictable delay by node incurred during the packet-forwarding state, $\text{DB}(i)$ is the predictable delay incurred during the backoff mechanism, and L and R are packet length and data rate, respectively.

The probability $P^i_{\text{idle}}(t)$ is given as follows:

$$P^i_{\text{idle}}(t) = e^{\lambda t}, \tag{4}$$

where λ is the cumulative packet arrival rate (including neighbor nodes) at node N_i.

The predictable delay $\text{DA}(i)$ is computed as follows:

$$\text{DA}(i) = P^i_{\text{idle}}(\text{slot}) * (\text{RTS} + 2 * \text{SIFS} + \text{CTS}) \\ + \left(1 - P^i_{\text{idle}}(\text{slot})\right) * (\text{RTS} + 2 * \text{SIFS} + \text{DB}(i)). \tag{5}$$

The predictable delay $\text{DB}(i)$ is given by the following equation:

$$\text{DB}(i) = \left[\frac{1}{\{P^i_{\text{idle}}(\text{DIFS}) * P^i_{\text{idle}}(\text{slot})\}}\right] \\ * \left[P^i_{\text{idle}}(\text{DIFS}) * \left(\text{DIFS} + \text{avg}_{\text{bt}} + \text{RTS} + 2 * \text{SIFS}\right.\right. \\ \left.\left. + P^i_{\text{idle}}(\text{slot} * \text{CTS})\right)\right] + \left[\left(1 - P^i_{\text{idle}}(\text{DIFS})\right) * X\right], \tag{6}$$

where $X = \text{RTS} + 3 * \text{SIFS} + \text{CTS} + L + \text{ACK}$ and ACK is the length of the acknowledgment packet.

The proposed protocol assumes contention window as $\text{CW}_{\text{min}} = 32$ and $\text{CW}_{\text{max}} = 1024$. According to the binary exponential backoff algorithm in the CSMA/CA protocol, the backoff delay avg_{bt} of the n^{th} retransmission $(0 \le n \le 5)$ is given by the following equation:

$$\text{avg}_{\text{bt}} = \sum_{n=0}^{4} P^i_{\text{idle}}(\text{slot}) * \left(1 - P^i_{\text{idle}}(\text{slot})^n * 2^{n-1} * \text{CW}\right) \\ + \left(1 - P^i_{\text{idle}}(\text{slot})\right)^5 * 2^4 * \text{CW}. \tag{7}$$

In ad hoc networks, the propagation delay is more driven by the distance between node N_i and node N_{i+1}. Since the value of the propagation delay is negligible compared to the other delays, we ignore the effect of the propagation delay.

3.5. Multipath Node-Disjoint QoS Aware Routing. In this section, we propose a cross layer based on-demand routing protocol CADM, which exploits the MAC layer information to discover congestion-free routes.

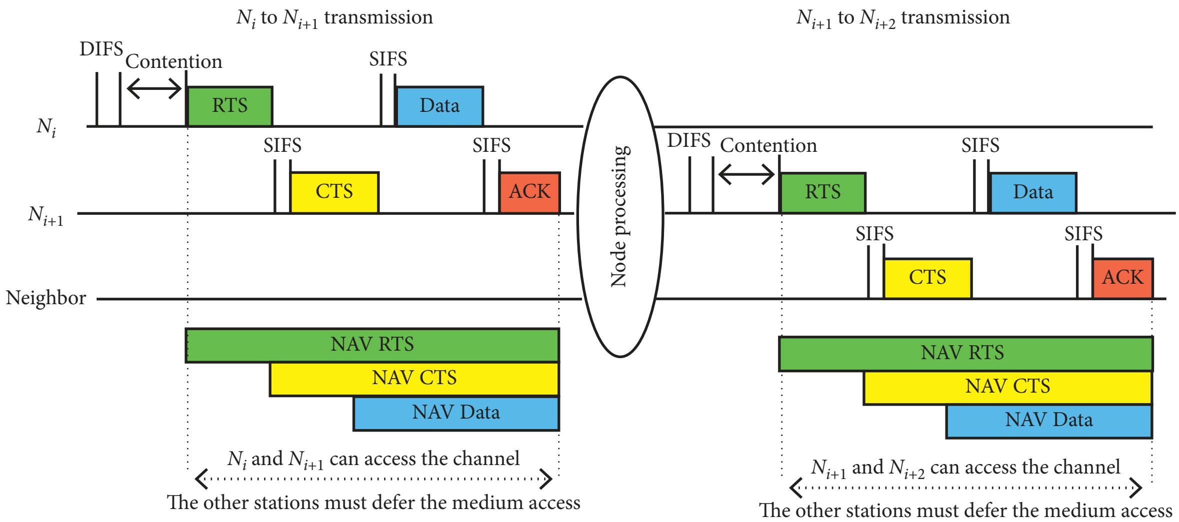

Figure 3: Forwarding process in contention-based DCF.

Traditional routing protocols in ad hoc networks, such as AODV, DSR, and DSDV, are mainly focused on discovering the single route between the source node and the destination node. In such a scenario, when the primary route breaks, the intermediate nodes drop the packets as alternate routes are not available to reach destination until the next route discovery process is initiated. Multipath routing protocol provides the advantage by discovering multiple routes in the single route discovery phase. These multiple paths between source and destination can be useful during dynamic and unpredictable topology changes in MANETs. The multipath routing effectively drops the rate of route discovery. Hence, the latency for rediscovering another route is reduced when existing used route fails. When multiple routes are available in advance, it can be lead in improving the effective bandwidth of the communication channel by dealing with congestion and frequent link failures.

3.6. Route Discovery. This section proposes a distributed algorithm, which select the appropriate data rates between the links and determine a delay-efficient route from source to destination for admitting a new flow. A route should satisfy the delay requirement of the requesting service, i.e., the end-to-end delays for data packets should be smaller than the required delay.

When a source has a data to communicate with the destination node, it checks its routing table for a valid route to the destination. If found, it sends the packet to the next hop node in the route towards the destination. However, if the valid route is not found in the routing table, the source initiates the route discovery process. During the route discovery process, the source creates a route request (RREQ) packet. Compared to the traditional RREQ packet, the proposed protocol uses two additional fields $\text{Delay}_{\text{Thresh}}$ and $\text{Delay}_{\text{Remaining}}$. The $\text{Delay}_{\text{Thresh}}$ field contains the end-to-end threshold time requirements to be satisfied for the packets. $\text{Delay}_{\text{Remaining}}$ is the remaining time to reach the destination from the current node and is updated at every intermediate node. At the source node, both fields are initialized to the $\text{Delay}_{\text{Thresh}}$ value. Source address, destination address, and sequence number of the modified RREQ packet header uniquely identify a RREQ packet.

It is essential to carefully select an appropriate data rate while broadcasting a RREQ packet. For example, if RREQ packets are forwarded at the lowest rate (2 Mbps), the discovered routing path will consist of long-range links over which higher data-rate communication will not be successful. It means the data rate selected for the RREQ packets restricts the maximum possible data rate for the link of the discovered route. Thus, the source node selects data rate DRi for the RREQ packet as a highest rate, which it had used during last communication with neighbor hop. If the node that forwarded the RREQ packet did not receive the response within the stipulated time from its neighbor node, it chooses the next lower data rate DR$i-1$ and resends the RREQ packet. If neighbor node receives the RREQ packet, it verifies if it is the destination node. If not, it analyses the queue utilization as per equation (2) before inserting the packet in its interface queue. If the Q_{util} of a node is already in the drop level, the node does not forward the packet forward and instead drops the packet. This is because forwarding the RREQ packet to other nodes will intensify the congestion in two aspects. Firstly, transmission of this RREQ packet increases the use of the medium around the congested area. Secondly, route discovery across the congested area results in additional transmission burden. If the Q_{util} of node is in advanced, or the alert level, the node inserts the packet in interface queue to be forwarded to next neighboring nodes. Subsequently, the node computes the packet-forwarding delay D^i_{delay} given by equation (3). The node then executes the forwarding eligibility test as per the following equation:

$$\text{Delay}_{\text{Remaining}} = \left(\text{Delay}_{\text{Thresh}} - D^i_{\text{delay}}\right). \tag{8}$$

If $\text{Delay}_{\text{Remaining}} \leq 0$, the intermediate node drops the packet. Otherwise, node updates the $\text{Delay}_{\text{Remaining}}$ field in the RREQ packet and starts the timer T_{wait} to receive multiple RREQ packets traversed through the disjoint path. After timer T_{wait} expires, the intermediate node selects the RREQ packet with maximum value of the $\text{Delay}_{\text{Remaining}}$ field. The intermediate node updates the routing table with an additional information of $\text{Delay}_{\text{thresh}}$ corresponding to the route entry. When a MAC layer passes the received packet to the network layer, it informs the data rate at which it is received. It also updates the link status of its communicating neighbors and their data rates in the routing table. The intermediate nodes then broadcasts the RREQ packet to their neighbors until it reaches the destination. On receiving the first RREQ by the destination node, it does not reply immediately but waits for T_{wait} time. The route request process of the CADM protocol is explained in Algorithm 1.

3.7. Route Reply. Route reply (RREP) packets are generated only by the destination node to the corresponding RREQ packets that are cached and arrived from the source node via loop-free and node-disjoint route to the destination. The intermediate nodes are prohibited from generating RREP packets to route request to avoid stale route information. In the proposed protocol, the traditional RREP packet is modified with additional field $\text{Max}_{\text{queue delay}}$, which holds the maximum value of queuing delay among the intermediate nodes on the downstream route. When the destination node prepares the route reply for all RREQs received, it initializes $\text{Max}_{\text{queue delay}}$ field to zero.

Let $Q_{\text{delay}}(t)$ be the weighted moving average queuing delay of any node for time interval δt and is given by the following equation:

$$Q_{\text{delay}}(t) = \eta \times Q_{\text{delay current}}(t) + (1-\eta) \times Q_{\text{delay}}(t-1), \tag{9}$$

$$\eta = \frac{Q_{\text{size}} - Q_{\text{load}}}{Q_{\text{size}}}, \tag{10}$$

where η acts as a weight parameter to regulate network congestion and $Q_{\text{delay current}}(t)$ is the queue delay at current time t.

When the RREP packet reaches the intermediate node, it compares $Q_{\text{delay}}(t)$ with $\text{Max}_{\text{queue delay}}$ specified in the packet. If $Q_{\text{delay}}(t)$ is greater than $\text{Max}_{\text{queue delay}}$, then $\text{Max}_{\text{queue delay}}$ is replaced by the $Q_{\text{delay}}(t)$ value and the RREP is forwarded to the downstream node of the route. The source node may receive several RREP packets from different routes. On receiving the first RREP packet, source node waits for T_{wait} duration. At the end of T_{wait} duration, the route table of the source node may have multiple route entries to the destination. The source node then chooses the primary route that has the minimum $\text{Max}_{\text{queue delay}}$ value. The route reply process of the CADM protocol is explained in Algorithm 2.

3.8. Route Maintenance. In mobile ad hoc networks, the network topology frequently changes due to continuous movement of mobile nodes, thus causing frequent link interruption. In the proposed protocol, any node which detects either a QoS violation in terms of delay or a link failure informs the source node by sending a route error packet (RERR). During the data transmission, each node computes the packet-forwarding delay according to equation (3). The intermediate node then performs the forwarding eligibility test as per equation (8). If $\text{Delay}_{\text{Remaining}} \leq 0$, the intermediate node notices the QoS violation, and it sends the RERR packet to the source node. If the forwarding eligibility test is successful, the intermediate node updates the $\text{Delay}_{\text{Remaining}}$ in the data packet and forwards it to the next node in the route. In a case, if the source node itself moves away from the neighbor nodes, it reinitiates the route discovery process to find route to the destination node. If the intermediate node in the route moves away, the upstream node in the route sends a link failure notification message to each of its active upstream neighbors through RERR until it reaches the source node.

3.9. Data Transmission and Rate Adaptation. Use of high data rates provided by IEEE standard has some trade-offs. High data rate requires a high signal to interference noise ratio (SINR), which is achieved through an efficient modulation scheme. The physical layer of the 802.11 protocol stack uses several modulation and coding schemes, namely, DSSS, FHSS, and OFDM. As the channel is volatile and frequent topological changes in case of ad hoc networks, the current state of the transmission channel should be used to decide the modulation scheme to be applied to support appropriate IEEE 802.11 protocols.

The range of the link depends on the data rate, i.e., higher the data rate, the shorter the range. High data rates also cause frequent link failures in case of mobility. This leads to an increase in routing overhead. Therefore, to support higher data rates, the number of hops between the source and destination node has to be increased. Achieving a high data rate and long range simultaneously is practically infeasible. The effectiveness of the rate adaptation scheme depends on how fast it can respond to transmission failures due to the channel error, node mobility, or packet collisions.

When a data packet is ready at a node N_i, it searches for the active primary route in the route table. If the route is found in the routing table with the data rate DRi, the node uses it to send the packet at data rate DRi. In order to improve network performance, in this paper, we adopt the rate adaptation mechanism proposed in [23], based on the underlying network parameters. The novelty of this rate adaptation mechanism is that it accurately estimates the channel condition despite the presence of various dynamics caused by fading, mobility, and hidden terminals and effectively selects the appropriate data rate.

4. Simulation of CADM Protocol

The performance of the proposed CADM protocol is evaluated using the simulation experiment carried out using the NS2 [24] simulation tool. In the simulation, the 802.11b network is assumed at the MAC layer and configured with 80 mobile nodes uniformly distributed over an area of $1500 * 1500$ m. The random waypoint model is used to model

```
(1) N ⟵ node, S ⟵ source node, D ⟵ destination node
(2) I ⟵ intermediate node
(3) if S has data to send then
(4)    if S has path to D then
(5)       Start data_transfer()
(6)    else
(7)       Create RREQ packet
(8)       Select appropriate DRi
(9)       Initiate RREQ_flooding()
(10)      if no reply received within threshold time then
(11)         Decrease DRi to DRi − 1
(12)         Repeat step in 11
(13)      end if
(14)      if N ≡ I then
(15)         if Q_util = drop level then
(16)            Drop the packet
(17)         end if
(18)         while T_wait > 0 do
(19)            Hold the RREQ packet in received_rreq_packet table
(20)            Compute Delay_Remaining as given in equation (8)
(21)            if Delay_Remaining < 0 then
(22)               discard_RREQ()
(23)            end if
(24)         end while
(25)         Select RREQ packet with min {Delay_Remaining}
(26)         relay_RREQ
(27)      else
(28)         N ≡ D
(29)         Receive first RREQ packet start a timer T_wait
(30)         while T_wait > 0 do
(31)            Hold the RREQ packet in received_rreq_packet table
(32)         end while
(33)      end if
(34)   end if
(35) end if
```

ALGORITHM 1: Route request algorithm.

the node's mobility. Twenty nodes were randomly selected as a constant bit rate (CBR) real-time sources, generating 512 bytes of data packets to be sent to the randomly chosen destination nodes. The MAC layer is based on IEEE 802.11 DCF. The interface queue at the MAC layer can hold a maximum of 50 packets. The duration of each simulation is 900 seconds, and each data point is calculated as an average of 10 simulation runs. The key performance metrics evaluated are packet delivery ratio, average end-to-end delay, normalized routing load, and average data rate used by the nodes to transmit data packets. The simulation parameters assumed at the MAC layer is mentioned in Table 1.

4.1. Performance Evaluation of CADM Protocol. The performance of the proposed CADM protocol is compared with the CRP protocol. For the purpose of fair comparison, identical traffic and mobility scenarios are used for CADM, CRP, and CARA protocols. In the simulation, three protocols are compared and evaluated by varying two parameters, as shown in Table 2.

The key performance metrics used for evaluating CADM are as follows:

(i) *Packet Delivery Ratio.* It is the ratio of the successful packets delivered at the destination node to the total packets generated at the source node. It is a vital metric to measure the packet losses which directly affects the network throughput.

(ii) *Average End-to-End Delay.* It is the delay incurred in successfully transmitting the packets from the source to the destination. It includes the delay which caused packet buffering in the queue, channel contention delay, transmission delay, retransmission delay, and propagation delay. It is considered to be one of important metrics to analyze the QoS routing performance.

(iii) *Normalized Routing Load (NRL).* It is the number of routing packets exchanged per data packets delivered at the destination node. NRL metric measures the efficiency of the routing protocol over low bandwidth and congested wireless networks.

$\text{Max}_{\text{queue delay}} \longleftarrow 0$
if $N \equiv D$ **then**
    Prepare RREP for each RREQ packet received
    relay_RREP ()
end if
if $N \equiv I$ then
    Compute Q_{delay} as in equation (9)
    if $Q_{\text{delay}}(t) > \text{Max}_{\text{queue delay}}$ **then**
        $\text{Max}_{\text{queue delay}} = Q_{\text{delay}}(t)$
    end if
    relay_RREP () to the downstream node along the path
end if
if $N \equiv S$ **then**
    Receive first RREP and start a timer T_{wait}
    while $T_{\text{wait}} > 0$ **do**
        Hold the RREQ packet in the received_rrep_packet table
    end while
    Choose the route with minimum $\text{Max}_{\text{queue delay}}$
    send_data ()
end if

Algorithm 2: Route reply algorithm.

Table 1: Simulation parameters in the MAC layer.

Parameters	Value
MAC header	52 bytes
PHY header	28 bytes
Rate for MAC/PHY header	1 Mbps
RTS	44 bytes
CTS	38 bytes
ACK	38 bytes
Slot time	20 μs
SIFS	10 μs
DIFS	50 μs
CW_{min}/CW_{max}	32/1024

Table 2: Scenario variations.

Varying parameters	Constant parameters
Traffic load = 10–80 packet/sec	Pause time = 5 sec, speed = 5 m/sec, CBR source: 20
Speed = 2–20 m/sec	Pause time = 5 sec, packet rate: 10 packets/sec, CBR source: 20

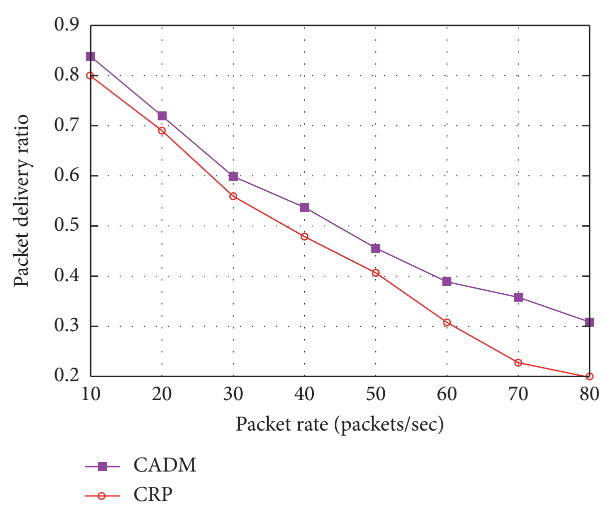

Figure 4: Packet delivery ratio (PDR) with varying packet rates.

4.1.1. Varying Packet Rate. In the first set of experiment to determine the effects of congestion in MANETs, the sending rate of every real-time source is varied from 10 packets per second to 80 packets per second. The number of source connections is kept fixed. The mobility of nodes is also fixed to 5 m/s.

Figure 4 shows the variation of the packet delivery ratio as a function of the packet rate. It can be noticed from the figure that, when the traffic load is less, there is not much difference in the performance of the CADM and CRP protocols. As the traffic load scales up, route links face a higher probability of congestion and the packet drop rate increases due to collisions or buffer overload. This results in retransmitting the packets more than once. However, the CADM outperforms the compared protocol CRP. This is primarily due to two reasons. Firstly, the queue load balancing mechanism through rate adaptation is executed at the node during the congestion. And secondly, due to the availability of multiple routes which were discovered during the initial route discovery phase so that in case the primary route is congested and does not satisfy QoS, it can switch to the alternate route without much delay. The CADM shows an improvement of 10–15% in the delivery ratio to that of CRP when the packet rate is increased from 60 to 80.

Figure 5 shows the average end-to-end delay against the varying packet rate for CADM and CRP. It can be noticed from the figure that, in both cases, the delay increases with the increase in the packet rate. It is evident from the figure that CADM outperforms CRP. This is mainly because

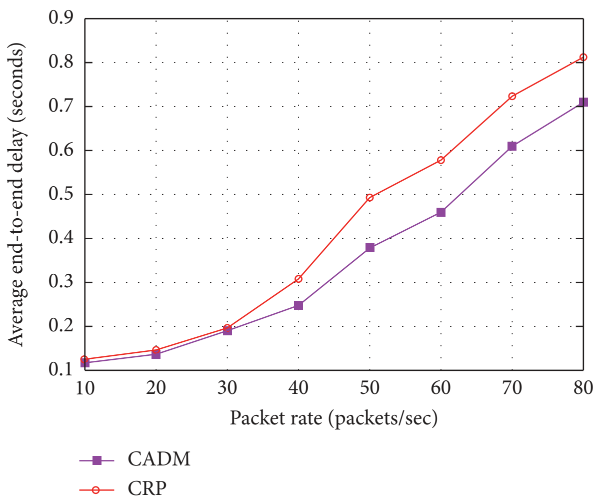

Figure 5: Average end-to-end delay with varying packet rates.

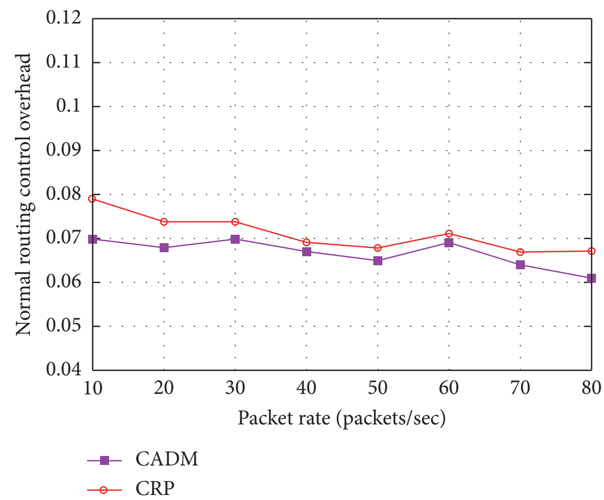

Figure 6: Normalized control overhead with varying packet rates.

CADM uses the route which satisfies delay requirement for the data packet. Also, in case of congestion, it uses variable data rate to balance the traffic load on route to reduce the packet drop and retransmissions. In CRP, on congestion detection, it uses a bypass route from the previous non-congested node to divert the traffic to the destination node. But this does not guarantee the data delivery within the required time period. If the primary route fails, CRP starts the route discovery process all over again, which incurs an additional delay. From Figure 5, it can be noticed that, as the packet rate increases from 30 to 80 packets per second, CADM shows almost 10% improvement over CRP.

Figure 6 shows the normalized control overhead with increasing traffic, for CADM and CRP protocols. It can be noticed from the figure that CADM shows improved performance over CRP and CARA. Increase in traffic load does not affect the performance of CADM severely, as it predicts the congestion at the node and initiates the load balancing mechanism by adapting to the variable data rate. As the protocol discovers multiple node-disjoint route paths during the route discovery process and supports rate adaption, the number of route rediscovery is significantly reduced. In case the congestion or link failure is predicted in the primary route, CADM switches the traffic through the secondary route without much of the packet drop and delay. As CRP does not have an alternate backup route, the rediscovery of routes is very frequent which in turn degrades the performance.

Figure 7 illustrates the variation of the average rate during the entire simulation time with the varying packet rate. We notice that the rates obtained in case of CADM are better than CRP due to the rate adaptation mechanism. The rate varies from 2 Mbps to 11 Mbps. However, in the case of CRP, the rate remains constant, equal to 2 Mbps, due to the lack of rate adaptation mechanisms. In addition, these results show that the nodes have more consecutive successes than consecutive failures, so their rates can rapidly get stabilized. As the traffic load increases, the queue load balancing mechanism in CADM helps to adopt a suitable

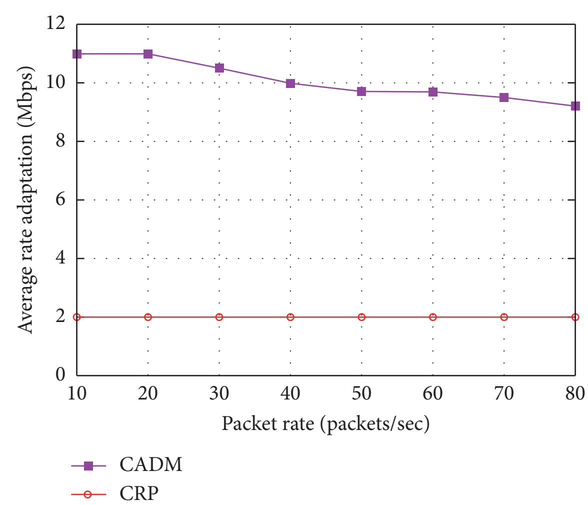

Figure 7: Average rate versus varying packet rates.

data rate in the event of a critical situation without much of the packets drop.

4.1.2. Varying Node Mobility. In the second set of experiments, the effect of mobility of nodes on the performance of CADM is analyzed. The number of source connection is fixed to 20. The packet sending rate is set to 10 packets/sec. Experiments are conducted by setting nodes speed to 2, 4, 6, 8, 10, 12, 14, 16, 18, and 20 m/s.

Figure 8 depicts the variation of the packet delivery ratio as a function of nodes mobility. When the mobility of the nodes is minimal, both CADM and CRP performs almost similar because there is less probability of link failure in the network which in turn reduces the buffering time in queue, the probability of congestion and collisions. When nodes are highly mobile, link failure probability increases and hence the packet delivery ratio decreases. The CADM shows a better delivery ratio compared to CRP when the node's speed increases, and this is mainly due to node-disjointness

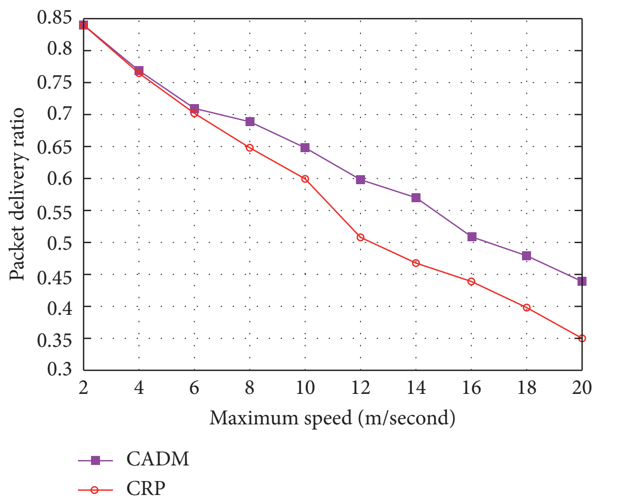

Figure 8: Packet delivery ratio (PDR) with varying nodes speed.

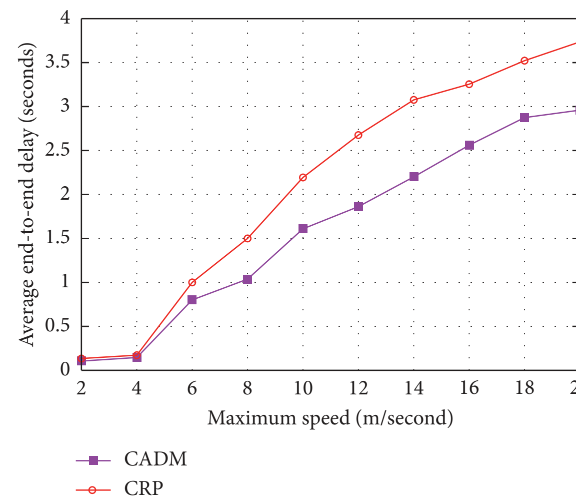

Figure 9: Average end-to-end delay with varying nodes speed.

in multiple routes discovered during the route discovery phase. When an active route is about to break due to the mobility of nodes, the intermediate node predicts the link failure and notifies the source node. The source node at once invalidates the current route in its route table and selects another valid node-disjoint route from its routing table to continue with the communication between source and destination without any interrupt.

The variation of the average end-to-end delay as a function of nodes mobility is as shown in Figure 9. From the figure, it can be observed that both the delay curves increase with the increase in speed of nodes. This is because the high mobility of nodes results in an increased probability of link failure that in turn causes an increase in the number of routing rediscovery processes. This makes data packets have to wait for more time in its queue until a new routing path is found. When the mobility of nodes is less, both the protocols give the same performance. The proposed CADM shows a 5–7% improvement in the average delay. This is because CADM guarantees the minimum delay route for the delivery of packets and switching to an alternate route in case of link failure. However, CRP does not guarantee the delay of packet delivery to the destination node.

Figure 10 shows the normalized control overhead with increasing nodes mobility. From the results, we can clearly observe that CADM performs better than CRP. This is mainly because CADM uses higher data rates for packet transmission whenever possible due to which more data packets are transmitted compared to control packets. CADM shows 5–6% improvement in the routing overhead proving it to be a lightweight protocol.

Figure 11 shows the plot of the average data rate adapted by nodes during the entire simulation time with varying mobility. We observe that the data rates adopted by nodes in CADM are higher and better than CRP due to the rate adaptation mechanism. The rate varies from 2 Mbps to 11 Mbps. However, in the case of CRP, the data rate adapted by the nodes remains static, i.e., 2 Mbps, due to the absence of the rate adaptation scheme. The results show that the

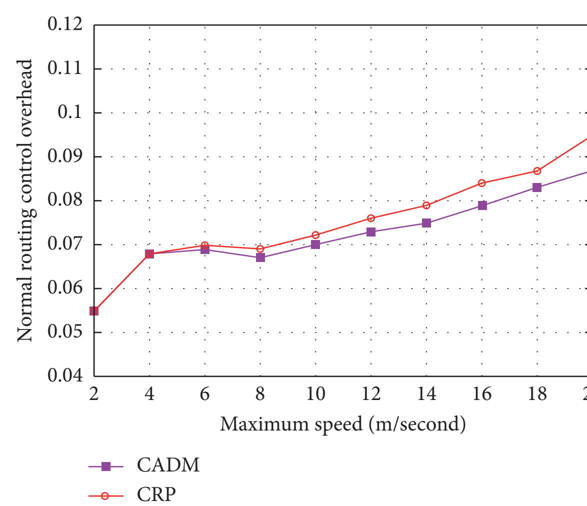

Figure 10: Normalized control overhead with varying nodes speed.

nodes have more repeated successes of packet delivery than repeated failures, which makes the data rate to get stabilized quickly. As the traffic load increases, the queue load balancing mechanism in CADM helps to adopt a suitable data rate in the event of a critical situation without much of the packets drop.

5. Conclusion

In this paper, a novel Cross-Layer Best effort QoS aware routing protocol (CADM) is presented. The proposed protocol is an adaptive and reliable congestion control based on multirate adaptation. It utilizes wireless resources efficiently and provides better QoS support for delay-sensitive communications in multirate MANETs. In CADM, the source tries to discover multiple node-disjoint paths that satisfy QoS requirements in terms of delay and chooses the links with high data rates to achieve high throughput while sending packets. The CADM is a congestion-adaptive routing protocol, where it not only avoids the congested

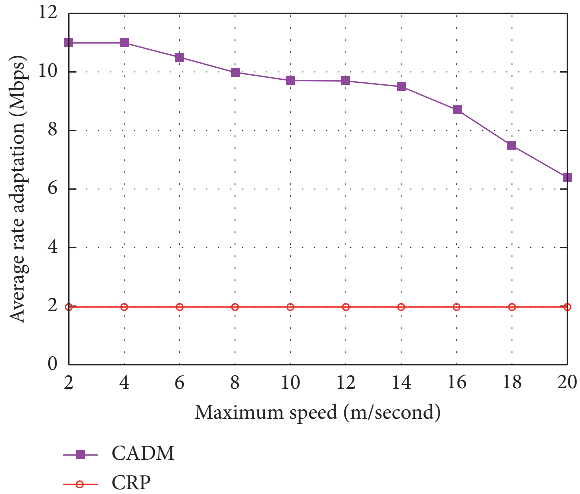

Figure 11: Average rate versus varying nodes speed.

node while constructing a route but also deals with it reactively. A detailed simulation study has been carried out to evaluate the performance of the proposed protocol with a well known CRP protocol. Simulation results show that the CADM protocol significantly reduces both the packet drop ratio and the end-to-end delay without much impact on control overhead. CADM protocol provides better throughput than CRP due to its variable data-rate adaptation technique.

Data Availability

The dataset used to support the findings of this study may be released upon application to the VIT, Vellore, India

Conflicts of Interest

The authors declare that there are no conflicts of interest regarding the publication of this paper.

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