Multipath Load Balancing & Rate Based Congestion Control for Mobile Ad Hoc Networks (MANET)

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Abstract

In mobile ad hoc network (MANET), congestion is one of the most important restrictions that deteriorate the performance of the whole network. Multi-path routing can balance the load better than the single path routing in ad hoc networks, thereby reducing the congestion by dividing the traffic in several paths. This paper presents a new approach Multipath Load Balancing and Rate Based Congestion Control (MLBRBCC) based on rate control mechanism for avoiding congestion in network communication flows. The proposed approach contains an adaptive rate control based technique in which the destination node copies the estimated rate from the intermediate nodes and the feedback is forwarded to the sender through an acknowledgement packet. Since the sending rate is adjusted based on the estimated rate, this technique is better than the traditional congestion control technique. Simulation results show that proposed technique has better packet delivery ratio and improved throughput and also controls the congestion in more effective manner.

Keywords: MANET, Congestion, Multipath Routing, Load Balancing, Rate Control

I. INTRODUCTION

1.1. Mobile Ad hoc Networks (MANET)

The network comprising mobile nodes with no infrastructure is termed as mobile ad hoc networks (MANET). The self-organization of the mobile nodes results in a network over radio links. For establishing routes among nodes, some specific routing protocols are used. [1] High mobility nature of the ad hoc networks resulted in new challenge that initiates a concept of dissimilarity between wireless characteristics and nature of the network topology. The exclusive nature of the mobile ad hoc network makes it to be deployed in any place and at any occasion. [3]

The application of the mobile ad hoc network includes military battlefield circumstance, disaster relief, and rescue during emergency, discovery etc. The group related application of mobile ad hoc networks include teleconferencing, data dissemination services etc. [2]

1.2. Congestion Control

It is essential to adjust the data rate used by each sender in order not to overload the network, where multiple senders compete for link bandwidth. Packets are dropped when they arrive at the router and cannot be forwarded. Many packets are dropped while excessive amount of packets arrive at a network bottleneck. The packets dropped would've traveled long way and in addition the lost packets often trigger retransmissions. This intimates that even more packets are sent into the network. And so, network throughput is still more worsened by the network congestion. There are chances of congestion collapse where almost no data is delivered successfully if no appropriate congestion control is performed. [4]

Shared broadcast medium is used in mobile ad hoc networks. Medium capacity which is very inadequate is shared within all the nodes in a collision domain. While delivering data to multiple destinations, multicast communication is of great concern in these networks, since it helps saving resources. Group communication which is an inherent feature of many proposed applications in MANETs is added to this broadcast medium. So, it is important to avoid congestion collapse in wireless multihop networks in order to perform efficient congestion control. [5]

1.3. Rate Control

The protocol of rate control is proposed for distinctive flows of characteristics user which takes in two components such as end-host congestion control layer among IP and TCP/UDP and every router upholds a single fair share for each link. [6] The issues such as exploding nature of the network and self-locking scenario while ACK packets arrive can be avoided by rate based transmission technique. [7]

In order to meet the bandwidth and delay requirements of real time traffic, rate control is done in a localized manner at each mobile node in entirely scattered and decentralized way. The rate control is designed for restricting the best effort traffic for creating the necessary bandwidth. Rate control also permits the best effort traffic to make use of the bandwidth in efficient way that is not used by the real time traffic during any particular situation. The total rate of all best effort traffic and real-time traffic distributed over each load shared media channel is sustained below a certain threshold, for minimizing the excessive delay. [8]

The method of identifying the approved data rate is facilitated by the explicit technique of rate control mechanism and thus the flows responds quickly to modulation in bandwidth and re-routing events. [9] The requisite transmission rate for the available resource facility can be adjusted with the help of controlling the flow which avoids congestion. [10]

1.4. Problem Identification and Solution

An important challenge in congestion control technique in MANET is that how the sender learns about network congestion and adjusts its rate.

In our previous paper [15], we have proposed congestion controlled adaptive multi-path routing protocol to achieve load balancing and avoid congestion in MANETs. The selected fail-safe multiple paths include the nodes with least load and more battery power and residual energy. When the average load of a node along the route increases beyond a threshold, it distributes the traffic over disjoint multi-path routes to reduce the traffic load on a congested link.

II. Related Work

S.Karunakaran et al [1] proposed a cluster based congestion control (CBCC) protocol that consists of scalable and distributed cluster-based mechanisms for supporting congestion control in ad hoc networks. The clusters autonomously and proactively monitor congestion within its localized scope. The present approach improves the responsiveness of the system when compared to end-to-end techniques. After estimating the traffic rate along a path, the sending rate of the source nodes is adjusted accordingly. Thus this protocol look forward the injection of dynamic flows in the network and proactively adjusts the rate while waiting for congestion feedback.

S.Venkatasubramanian et al [8] proposed QoS architecture for Bandwidth Management and Rate Control in MANET. The proposed QoS architecture contains an adaptive bandwidth management technique which measures the available bandwidth at each node in real-time and it is then propagated on demand by the QoS routing protocol. The source nodes perform call admission control for different priority of flows based on the bandwidth information provided by the QoS routing. A rate control mechanism is used to regulate best-effort traffic, whenever network congestion is detected.

Kai Chen et al [9] proposed an explicit rate-based flow control scheme (called EXACT) for the MANET network. In EXACT, flow's allowed rate is explicitly conveyed from intermediate routers to the end-hosts in each data packet's special control header. As a result, EXACT reacts quickly and precisely to re-routing and bandwidth variation, which makes it especially suitable for a dynamic MANET network.

Kazi Chandrima Rahman et al [11] proposed explicit rate-based congestion control (XRCC) for multimedia streaming over mobile ad hoc networks. XRCC addresses the problems that TCP faces when deployed over ad-hoc networks, and thus shows considerable performance improvement over TCP. Although XRCC minimizes packet drops caused by network congestion as compared to TCP congestion control mechanism, it still suffers from packet drops.

Hongqiang Zhai et al [12] proposed a novel rate based end-to-end Congestion Control scheme (RBCC). Based on the novel use of channel busyness ratio, which is an accurate sign of the network utilization and congestion status, a new rate control scheme has been proposed to efficiently and reliably support the transport service in MANET. In RBCC, a sub-layer consisting of a leaky bucket is added under TCP to control the sending rate based on the network layer feedback at the bottleneck node.

Emmanuel Lochin et al [13] proposed a complete reliable rate-based protocol based on TCP-Friendly Rate Control (TFRC) and selective acknowledgement (SACK) mechanisms. This design also introduces a flow control variable, which regulates the sender to avoid packet loss at the receiver due to a slow receiver. In this mechanism, there is no packet loss due to flow control, at the receiver, and applies a smoothness criterion to demonstrate that the introduction of the flow control inside TFRC does not alter the smoothness property of this mechanism.

Yuedong Xu et al [14] proposed a fully distributed congestion control algorithm to balance throughput and fairness for TCP flows in multihop ad hoc networks. The interactions between the hidden nodes and network congestion are mainly focused. A distributed algorithm to improve the end-to-end throughput, and at the same time, provide per-flow fairness by exploiting cross-layer information is proposed. In the link layer, each node uses a proportional controller to determine the ECN marking probability for the purpose of notifying incipient congestion. Then the rate based TCP sender adjusts its sending rate according to the feedbacks from the link layer.

III. Rate Based Congestion Control Technique

3.1. Overview

The source node forwards the data packet to the destination through the intermediate nodes. On reception of the data packet at the intermediate node, percentage of channel utilization and queue length are estimated and node is verified for congestion status. This process is repeated at every intermediate node, and finally the packet reaches the destination node. After the reception of the data packet, the destination node checks for the rate information in the packets IP header fields. Along with other essential fields, estimated rate is copied to an acknowledgement packet and sent as a feedback to the sender. The sender performs rate control according to the estimated rate obtained from the destination.

3.2. Estimation of Metrics

3.2.1 Rate Estimation

The source node forwards the data packet to the destination through the intermediate nodes. Based on the buffer status of the intermediate nodes, the packet arriving or leaving at the intermediate node is either discarded from the node or updated with the status of the node.

The rate of the incoming and outgoing packet [16] is estimated as follows.

The rate of the incoming packet (τ_i) is estimated as the reciprocal of the arrival time interval of the incoming packet.

$$\text{i.e. } \tau_i = 1/T_i \qquad (1)$$

T_i is defined as the time interval of two consecutive packets received at the node.

The rate of the outgoing packet (τ_o) is estimated as the reciprocal of the service time of the packet.

$$\text{i.e. } \tau_o = 1/T_o \qquad (2)$$

T_o is defined as the time interval between the time that a packet arrives at node and time that it is transmitted

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successfully. It is also defined as summation of the time for queue, collision, back off and transmission.

The estimation of T_i and T_o is performed using exponential weighted moving average (EWMA) algorithm as follows [16].

$$T_{ic} = (1- \sigma)\, T_{ip} + \sigma\, (T_l - T_p) \quad (3)$$

$$T_{oc} = (1- \varepsilon)\, T_{op} + \varepsilon\, T_s \quad (4)$$

where T_{ic} = time interval of the currently arrived packet

T_{oc} = service time of current packet

T_{ip} and T_{op} are time interval of lastly arrived packet and service time of last packet respectively

T_l and T_p are the arrival time of the last packet and penultimate packet so T_l-T_p is the packet arriving interval

T_s is the service time of the last outgoing packet.

σ and ε are the constants for weighing T_i and To with value between 0 and 1

3.2.2 Estimation of Percentage Channel Utilization

In the wireless networks, when the transmission channel is being completely utilized, it is concluded that the network congestion has occurred. The channel utilization for time interval t is estimated using channel busy-time (T_c) metric on a percentage scale. The channel busy-time can be computed based on the category of control frame and the rate and data frame size. [18]

Let DIFS denote the distributed inter-frame spacing

Let SIFS denote the short inter-frame spacing.

Let α (t) denote the request to send (RTS) frame.

Let β (t) denote the clear to send (CTS) frame.

Let γ (t) represent the acknowledgement (ACK) packet.

Let δ (t) represent the beacon frame.

Let α_{delay}, β_{delay}, γ_{delay} and δ_{delay} denotes the delay component of RTS, CTS, ACK and beacon signals respectively.

Let D_{delay} and S_{delay} be the delay components of DIFS and SIFS respectively.

In order to compute the busy-time of a data frame (T_{cd}), D_{delay} interval is used. For a frame of size Z bytes transmitted at a rate τ, the channel busy time (T_{cd}) is computed using Eq (5).

$$T_{cd} = D_{delay} + d\,(Z)\,(\tau) \quad (5)$$

When RTS frames arrive at the data set, the T_{cr} for the respective frame is computed using Eq (6).

$$T_{cr} = \alpha_{delay} \quad (6)$$

Following the reception of RTS frame, when CTS frame is encountered in the data set, it is transmitted with S_{delay}. The respective T_{cc} is given using Eq: (7)

$$T_{cc} = S_{delay} + \beta_{delay} \quad (7)$$

Following the reception of CTS frame, when ACK frame is encountered in the data set, it is transmitted with S_{delay}. The corresponding T_{ca} is given using Eq: (8)

$$T_{ca} = S_{delay} + \gamma_{delay} \quad (8)$$

When a beacon frame is encountered in the data set, it comes first by the D_{delay} interval. Its T_{cb} value is computed using Eq: (9)

$$T_{cb} = D_{delay} + \delta_{delay} \quad (9)$$

Tc can be computed as the sum of the time utilized by the transmission of all data and control frames in the network and the total number of delay components at time t (explained in Eq (10)). The delay is taken into account while estimating T_c as the medium remains unshared among the stations in the network at time t.

$$T_c(t) = (\alpha(t)*T_{cr}) + (\beta(t)*T_{cc}) + (\gamma(t)*T_{ca}) + (\delta(t)*T_{cb}) + \left(\sum_{0}^{d(t)} T_{cd}(Z_i)(\tau_i)\right) \quad (10)$$

The percentage channel utilization at time t, CU (t) [18] is given using Eq (11)

$$\%\ CU\ (t) = \frac{T_c(t)}{10^6} * 100 \quad (11)$$

3.2.3 Estimation of Queue Length

The queue length describes the total traffic load in a mobile node. In general, when excess traffic flows through the mobile node, then there will more number of packets in the interface queue. Thus average queue size (L_Q) [19] is defined as the node's traffic in a long term. L_Q is given using Eq (12)

$$L_Q = \psi * L_{Qold} + (1-\psi) * L_{Qc} \quad (12)$$

Where L_{Qc} = current value of the queue length

ψ = constant in the range [0, 1]

C. Algorithm for Detecting Congestion

1. The source sends the data packets to the destination through the intermediate nodes.

2. Let L_{Qth} be the predefined threshold value of queue length.

Let % CU_{th} be the predefined threshold percentage channel utilization

Upon reception of the data packets, intermediate node verifies both the queue length and channel utilization (explained in section 3.2.2 and 3.2.3) and further assigns value to C_b as per following cases.

2.1 If $L_Q > L_{Qth}$ and %CU > % CU_{th}, Then

Set $C_b = 1$

End if

2.2 If $L_Q = L_{Qth}$ and %CU = % CU_{th}, Then

Set $C_b = 0$

End if

After the detection of congestion, the intermediate node estimates the new rate information using equations (13) and (14). The source updates its packet sending rate with this estimated rate. This is done after getting acknowledgement packet from receiver as feedback regarding the nodes congestion status which is described in section 32.4.

3.2.4. Algorithm for rate based congestion control

We assume the following metrics for the rate control technique

Let C_b = congestion bit set in the packets IP header

τ_s = current rate of the sender (or) sending rate

τ_c = current rate of the node

τ_e = rate estimated in the intermediate node

τ_p = estimated rate of the previous node

δ = factor which normalizes the rate value.

ACK = acknowledgement packet

The algorithm for rate based congestion control is described below.

Step 1

The current rate of the node [11] is taken as inverse of its queue length.

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Current rate, $\tau_c = 1/q$ (13)

Thus the rate estimated at the intermediate node (τ_e) [11] is as follows

$$\tau_e = \delta . \tau_p + (1- \sigma) . \tau_c \quad (14)$$

where τ_p is the estimated rate of the previous node. The factor σ is used in order to get the smoothed value of the rate.

Step 2

If $\tau_e > \tau_p$ Then

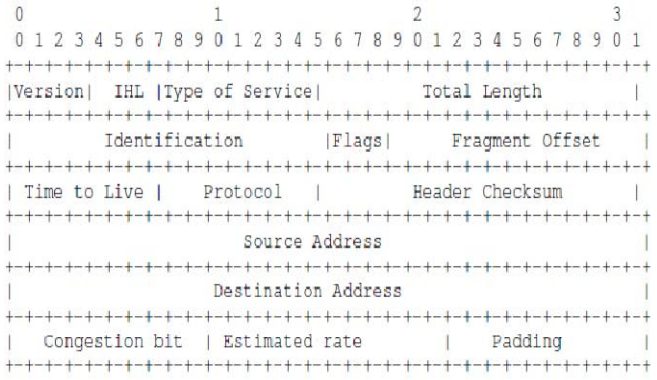
Set the packet with τ_e

End if

Step 3

The intermediate nodes insert the information of congestion bit (Cb) and the estimated rate (τ_e) into the options field of standard IP header.

The modified IP header format as defined in RFC 791 is given below:



Step 4

The process of rate estimation is repeated in every intermediate node and packets are updated with estimated rate value as per the above cases and finally the packet reaches the destination node.

Step 5

After the reception of the data packet, the destination node checks for the value of C_b and rate information in the packets IP header fields.

Step 6

Along with C_b, τ_e is also copied to an acknowledgement (ACK) packet and feedback to the sender by the destination node. The ACK packet contains the source address, destination address, and time stamp fields. [Shown in Figure 1]

Step 7

After receiving an ACK packet, sender checks the value of C_b. If $C_b = 1$, then it updates the current sending rate to the estimated rate τ_e.

(i.e) $\tau_s = \tau_e$ (15)

Since the sending rate is adjusted based on the estimated rate from the intermediate nodes, this technique is better than the traditional congestion control technique.

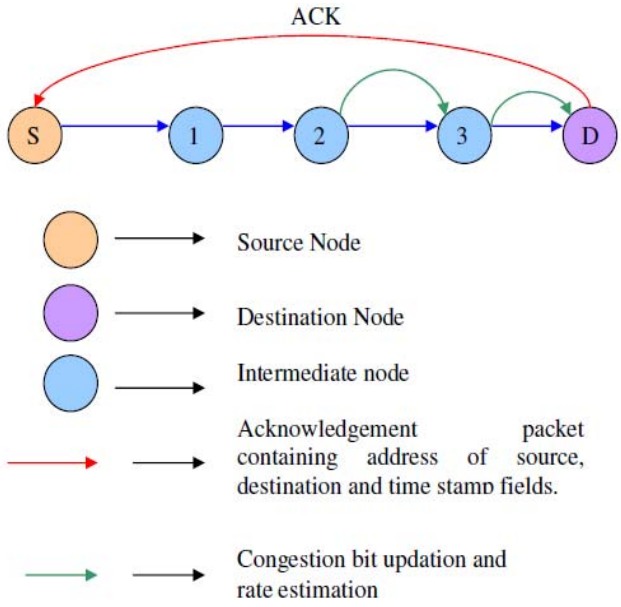


Figure 1. Congestion Control

IV. Simulation Results

A. Simulation Parameters

The proposed algorithm is experimented in the simulated environment with NS2 [17]. In this simulation, the channel capacity of mobile hosts is set to the value of 2 Mbps. We use the distributed coordination function (DCF) of IEEE 802.11 for wireless LANs as the MAC layer protocol. It has the functionality to notify the network layer about link breakage.

In our simulation, the number of nodes is fixed as 110. The mobile nodes move in a 1250 meter x 1250 meter square region for 50 seconds simulation time. We assume each node moves independently with the same average speed. All nodes have the same transmission range of 250 meters. In our simulation, the speed is fixed as 10 m/s. For mobility, Random Way Point mobility model is used. The simulated traffic is Constant Bit Rate (CBR). We vary the traffic rate from 250kb to 1000kb and number of traffic flows from 2 to 8.

B. Performance Parameters

The MLBRBCC algorithm is compared with XRCC [11] based the following parameters.

Average end-to-end delay: The end-to-end-delay is averaged over all surviving data packets from the sources to the destinations.

Average Packet Delivery Ratio: It is the ratio of the number of packets received successfully and the total number of packets transmitted.

Drop: It is the average number of packets dropped during the transmission.

Throughput: It is the number of packets received successfully.

The simulation results are presented in the next section.

A. Based on Flow

Initially we vary the number of flows as 2, 4, 6 and 8 keeping the rate as 250kb.

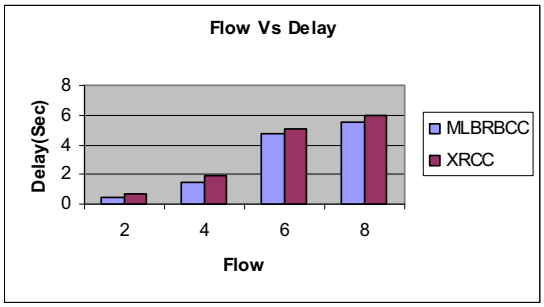


Figure 2. Flow Vs Delay

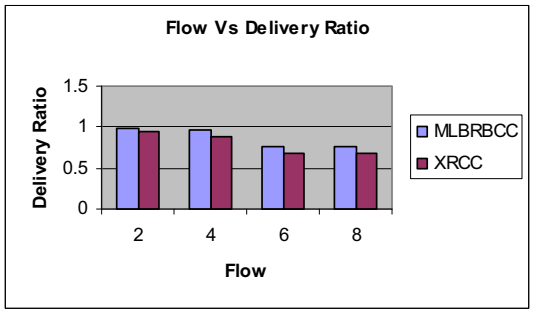


Figure 3. Flow Vs Delivery Ratio

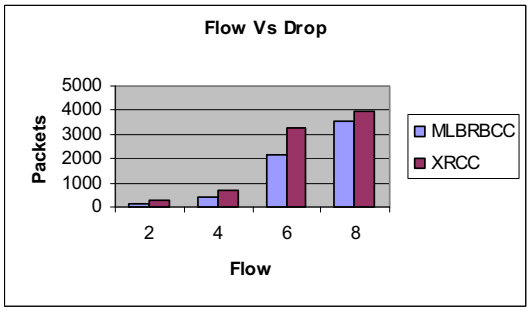


Figure 4. Flow Vs Drop

As the number of flows is increased from 2 to 8, the incoming traffic rate will also increase leading to the overflow of queue size. So the number of packet drops increases leading to the degradation of packet delivery ratio and increased end-to-end delay. Since the rate is adjusted as per the new sending rate at each sender and also multi path routing is used, the performance of MLBRBCC outperforms XRCC in all the metrics.

Figure 2 shows the results of average end-to-end delay for the increasing number of flows. From the results, we can see that MLBRBCC scheme has 46.8% less delay than the XRCC scheme.

Figure 3 show the results of average packet delivery ratio for the varying flows scenario. Clearly our MLBRBCC scheme achieves 17.8% more packet delivery ratio than XRCC.

Figure 4 shows the results of packet drop versus flows. From the results, we can see that our MLBRBCC has 51% lesser packet drop than XRCC.

B. Based on Rate

In the second experiment we vary the packet sending rate value as 250,500,750 and 1000Kb, keeping the no. of flows as 8.

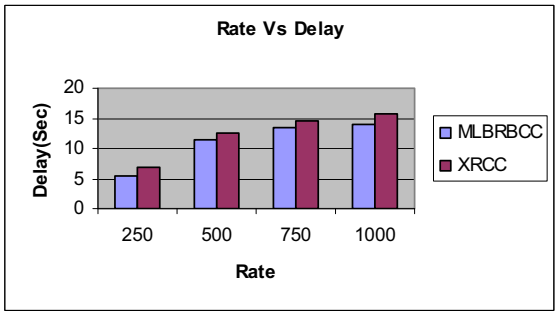


Figure 5. Rate Vs Delay

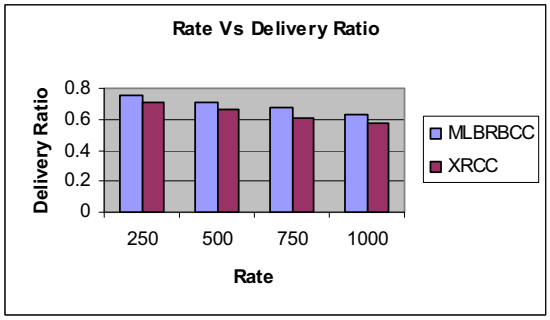


Figure 6. Rate Vs Delivery Ratio

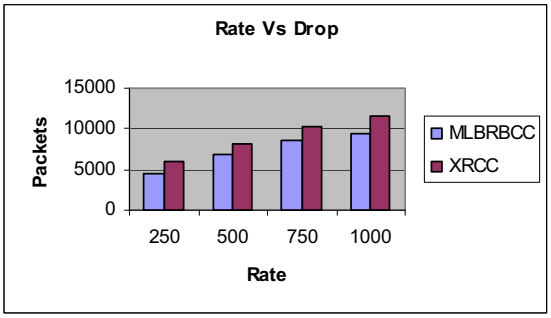


Figure 7. Rate Vs Drop

As the packet sending rate is increased from 250kb to 1000kb, for 8 flows, the incoming traffic rate of all flows increases leading to the overflow of queue size. So the number of packet drops increases (and is more than the previous experiment) leading to the degradation of packet delivery ratio and increased end-to-end delay. In MLBRBCC, Since the rate is adjusted as per the new sending rate at each sender and also multi path routing is used, it reduces the packet drops significantly. So the performance of MLBRBCC outperforms XRCC in all the metrics.

Figure 5 shows the results of average end-to-end delay for the increasing the rate. From the results, we can see that MLBRBCC scheme has 16.11% less delay than the XRCC scheme.

Figure 6 show the results of average packet delivery ratio for the varying rate scenario. Clearly our MLBRBCC scheme achieves 21.4% more packet delivery ratio than XRCC.

Figure 7 shows the results of packet drop versus rate. From the results, we can see that our MLBRBCC has 23.01% lesser packet drop than XRCC.

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V. CONCLUSION

In this paper, a technique for Multipath Load Balancing and Rate Based Congestion Control (MLBRBCC) is presented. In our technique, source node forwards the data packet to the destination node through the intermediate nodes. Upon reception of the data packet, the channel utilization percentage and queue length are estimated at each intermediate node along the destination. Based on these values congestion status and estimated rate are calculated and transmitted towards the destination. By checking the updated values from the intermediate nodes, the destination node determines the estimated rate and it is transmitted as a feedback to the sender. The sender performs rate control based on the estimated rate in the feedback packet. Simulation results shown that MLBRBCC has higher packet delivery ratio and less end-to-end data packet delay.

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