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Cross-Layer Congestion Control with Efficient Bandwidth Allocation and Dynamic Window Adaptation in Mobile Ad Hoc Network

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Abstract: Congestion in the network is a severe problem that affects the performance of MANET applications. When, a packet loss occurs in the wired medium, it clearly signs congestion in the network and suitable measures are taken to alleviate the effect of congestion. In a wireless environment, a packet can be lost for the factors such as wireless link error probability, link breakages, energy depletion, MAC contention fading and interference, etc. Due to various reasons contributing to the packet loss, Transmission Control Protocol (TCP) is unable to efficiently exploit the available network capacity in MANET. Across-layer congestion control algorithm with efficient Bandwidth Allocation and Dynamic Congestion Window Adaptation (CC-BADWA) is proposed in this study that uses TCP receiver's prediction of bandwidth availability in the network and the receiver advertised window size back to the sender which then adjusts sender window. The timeout and the delayed acknowledgement can differently able the sender to predict the level of congestion in the network. When the acknowledgement is not received due to congestion in the network, the sender can estimate the transmission probability, adopt the dynamic congestion window size and transmit the packets. This improves the network throughput since, the sending rate is adjusted consistent to the currently prevailing network status. The bandwidth allocation uses the currently available bandwidth based on the present active connections and the route optimization proposed in this research selects the optimal path for data forwarding. The receiver's timeliness helps to improve the network performance parameters throughput, end to end delay, packet delivery ratio of the existing method. The proposed algorithm is analyzed through ns 2 simulation tool.

Key words: Congestion control, cross-layer information, Ad Hoc networks, bandwidth estimation, congestion window adaptation

INTRODUCTION

The increasing demand for the wireless communication necessitates seamless network connectivity among the mobile devices forming an ad-hoc communication network. These networks consist of wireless multi-hop mobile nodes that are able to organize and configure themselves. Mobile Ad Hoc network does not have any fixed infrastructure and every node in MANET possess the capability of routing in itself. The applications in battlefield communications, disaster rescue and environmental monitoring makes use of MANETs where fixed infrastructure is unavailable (Chen *et al*., 2004). These applications need reliable and less delay communication. The MANETs differ significantly from the wired network of its dynamic nature and unpredictable interconnections among the nodes.

Transmission Control Protocol (TCP) is a reliable, stream orientedend to end protocol between the source and destination processes over the unreliable network. The reliability is achieved through TCP's sequence numbers, acknowledgement packet and packet retransmission in case of packet loss. The TCP ascertains flow control between source and the destination nodes at the transport layer. Congestion control at the transport layer is also performed by TCP. In wired network, TCP infers packet loss as an indication of buffer overflow in the path and appropriate control mechanisms are invoked. However in awireless environment, node mobility, shared wireless channel, unreliable channel result in delayed packet delivery to the receiver and packet losses. Perhaps, these interruptions are due to non-congestion. But, the traditional TCP does not distinguish the cause of these losses and interprets the scenario as congestion (Lochert *et al*., 2007). The delay and the timeouts cause unnecessary packet retransmissions which in turn pumps more packets into the network. Consequently, these retransmitted packets consume the network resources that are scarce through its network travel and processing.

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Ultimately, the mispredicted congestion degrades the network performance. Thus, it is essential to enhance TCP to be aware of the congestion and non-congestion packet losses inwireless environment.

TCP congestion control consists of slow start, congestion avoidance, fast retransmit and fast recovery phases. The TCP sender initiates data transmission in slow start phase wheresize of sender congestion window (s_cwnd) increases exponentially for every ACK received till it equals the slow start threshold (ss_thresh). When s_cwnd>ss_thresh, s_cwnd expands linearly. For every TCP segment transmitted, an associated TCP retransmission timer is started expecting an ACK before its expiration. If the ACK is not received before time out of this timer, the TCP segment is retransmitted. The TCP sender uses cumulative acknowledgement for the TCP segments received. When TCP receiver suffers from out-of-order segmentreception or segment loss, it immediately sends aduplicate acknowledgement to indicate the missing sequence number it actually expects. Upon reception of 3 consecutive duplicate acknowledgements, TCP sender immediately sends the expected segment without waiting for retransmission time-out. This phase is known as fast retransmit. After retransmitting the segment in fast retransmit, fast recovery adjusts the congestion window to maintain the data flow stream by not going to the slow start phase.

Upon timeouts or reception of three duplicate acknowledgements, asmisinter pretation of congestion, the TCP sender process reduces congestion window (Stevens, 1997). But, packet loss in a wireless setup could be caused by other factors also. Thus, the unnecessary reduction in congestion window results in performance degradation of wireless networks in case of non-congestion losses. The standard TCP employs (AIMD) Additive Increase Multiple Decrease congestion avoidance schemes. When there is no congestion, the congestion avoidance simply increases the congestion window size by one maximum segment size (1 MSS) for every Round Trip Time (RTT). When packet drop is detected by three duplicate acknowledgements, congestion window is reduced by half and in the case of timeout, the congestion window becomes one. If non-congestion factors result in packet drop then unnecessary congestion window size reduction yields.

Hence, the TCP performance has to be improved in wireless scenarios. The Cross-layer Bandwidth Estimation and Dynamic Congestion Window Adaptation scheme (CC-BADWA) proposed in this study aims to adjust the congestion window dynamically to carry out the packet transmission in consistent with the current network condition. The TCP receiver performs the bandwidth estimation and RTT calculation and accordingly predictsthe receiver window size. In the ACK, returned to the TCP sender, the advertised window is embedded and considered by the TCP sender to adjust its congestion window. When there is no acknowledgement due to congestion or no notification of non-congestion instead of reducing simply the congestion window, the proposed method calculates the transmission probability using the previous history based on which the segments are transmitted. The congestion window size is dynamically ascertained on the values using spatial-temporal correlation. The congestion window is adjusted to the current network capacity instead of blind reduction. It is shown that the transmission probability uses the currently prevailing network parameters to adjust the packet sending rate. The bandwidth fair sharing and the optimized route selection further improve the throughput of the network. The simulation results confirms that the CC-BADWA outperforms the network performance parameters Packet Delivery Ratio (PDR), throughput, end toend delay, jitter and normalized overhead.

Literature review: Shi *et al.* (2010) proposed a congestion control (RACC-Receiver Assisted Congestion Control) mechanism to maximize the TCP throughput in Wireless network. They used the receiver assisted bandwidth measurement and the advertised window of the receiver to adjust the sending rate and the congestion window at the sender. The receiver uses a timer to record the arrival of the next packet. If there is packet loss, then time out will indicate the packet drop to the receiver earlier than the sender. Now the receiver immediately sends a message to the sender requesting to retransmit the packet where it requires three duplicate acknowledgements. This mechanism still reduces the congestion window to 1 when timeouts at either sender or receiver. After one RTT, the source node adjusts the window to the receiver window size if the congestion mitigates. Chandran *et al.* (2001) address the congestion control with a feedback based solution called TCP-Feedback (TCP-F). The moment the network layer of any node in the current path experiences the route failure, it triggers a link failure message explicitly to the TCP source. Immediately the TCP sender enters the snooze state freezing all TCP state variables, timers and the congestion window size. When the route is reestablished, Reestablishment Notification is sent by the intermediate node to the sender making the sender to resume the interrupted session.

Holland and Vaidya (1999) presented a notification message for the link failure to improve the TCP performance that works on Dynamic Source Routing protocol (DSR). This message notifies the TCP sender about the link failure in the path and makes the TCP sender to enter the standby mode. During the standby mode TCP sender periodically checks the networkfor the route reestablishment. When the route is reestablished the sender restarts the transmission from the previously frozen state. Both TCP-Feedback and explicit link failure Notification, upon route failure notification stops transmission making the TCP sender to enter the snooze or standby mode. It avoids unnecessary retransmissions and reduction in congestion window size. However, both these mechanisms have not considered the congestion, out-of-order packet reception and wireless errors in wireless networks.

Ad hoc TCP (ATCP) (Liu and Singh, 2001) uses the Explicit Congestion Notification (ECN) messages as the feedback from the network layer. Along with ECN, it uses destination unreachable ICMP messages. Apart from route failures, ATCP addresses the packet losses from wireless errors. Between the TCP and IP layers, a new thin layer known as ATCP Layer is introduced. When ATCP layer receives the ICMP message, it changes the TCP sender to persisting state. Frozen TCP sender transmits no packets in this state until anew path is found. When ATCP gets an ECN message, TCP sender enters congestion state where congestion control is invoked. When a packet is lost and no ECN then ATCP assumes the loss is of wireless error and immediately it retransmits the packet. Thus, ATCP addresses, link failure, network partition and wireless error rate. Since, ATCP is transparent to TCP, nodes with and without ATCP can be compatible with this method. Yu (2004) proposed two mechanisms to keep routing protocols aware of the data and ACK packet losses occurring in between route failure and route reestablishment. The First mechanism, Explicit Packet Loss Notification (EPLN) notifies the TCP sender about the lost data packet that could not be salvaged and the second, Best Effort Acknowledgement Delivery (BEAD) attempts to retransmit the ACK packet either from the intermediate node or the TCP receiver.

TCP Westwood (Caseti *et al*., 2002) uses an end-to-end congestion control measure that involves the ACK arrival rate at the sender to compute the bandwidth. The values for congestion window and slow start threshold are set using the estimated bandwidth when receiving three duplicate acknowledgements or time out. If the ACK uses adifferent path than the data path, then the bandwidth estimate is affected. It considers not only the packet loss to set the congestion window but also the currently available bandwidth measures. In TCP Vegas (Brakmo and Peterson, 1995) congestion avoidance, the sender exploits the changes in Round Trip Time (RTT) to infer level of congestion in the network. If the observed RTT is higher than the threshold, the sourceshrinks its congestion window (cwnd), thus reducing its transmission rate. TCP Vegas calculates the difference between actual and expected sending rate. Two thresholds α and β are used to set the value of cwnd. If the diff is less than α, then the cwnd is increased, greater than β, the cwnd is decreased linearly. If α<diff<β, cwnd is not changed. TCP Vegas relies on the propagation delay which changes when nodes move. Hence, inaccurate estimation of baseRTT will decrease the network throughput.

Chen *et al.* (2003) suggested an adaptive method to set the limit for congestion windowdepending on the current Round Trip Hop Count (RTHC) of the route. The window limit should not exceed hop count of forward and backward path between source and destination. They derived an upper bound on bandwidth delay product that is 1/5th of RTHC. This upper is used to set the congestion window limit dynamically. The congestion control presented in Zhai *et al.* (2005) is an end to end, rate based design that depends on the feedback carried in the ACK packet. The congestion header in every packet stores the sending rate of the intermediate nodes calculated on the channel busy ratio and the destination copies this into the ACK packet sent back to the sender. Then, leaky bucket algorithm is used by the source to regulate the sending rate of the sender. The sender sets the CWND to the value of receiver's advertised window.

Kliazovich (2006) in Cross Layer Congestion control C3TCP used two metrics bandwidth and delay of the path as the feedback to the sender to adjust its sending rate. The collected feedback is stored in the link layer header and forwarded by the intermediate nodes. It deploys a module called Congestion Control Module (CCM). The CCM at the destination copies the bandwidth and delay information into the outgoing ACK packets. The CCM at the sender makes use of the end-to-end measurements bandwidth and RTT (forward path) from the link layer header to set the congestion window size. It is set to the minimum of calculated congestion window and the capable windowsize imbeddedby the receiver in the ACK packet.

Cluster-Based Congestion Control (CBCC) proposed by Karunakaran and Thangaraj (2010) consists of scalable and distributed clusters in mobile ad hoc network. Clusters monitor the congestion level in its localized area and this information is communicated through the cluster heads that reduces the control overhead. The nodes in the cluster communicate its current load and sends to cluster head. Upon collecting load estimate from nodes, cluster head estimates the traffic rate estimate IE. The aggregate estimate for the path can be calculated by exchange of IE among the cluster heads over the path. After computing the traffic rate over the path, datatransmission rate at the source is varied proactively.

Kumaran and Sankaranarayanan (2012) presented the routing method for MANET centered on monitoring network congestion with the aid of average queue length at the node level. The route discovery mechanism employs Congestion Free Set (CFS) to discover a path with less or no congestion between source and destination. The nodes periodically check data packet transmission and calculate congestion status of its own. The congestion status is exchanged with Congestion Status Packet (CSP). When the primary path becomes congested, an alternate path will be chosen from the one hop CFS set.

Thilagavathe and Duraiswamy (2011), the researchers suggested a combined method to overcome congestion at the transport and the MAC Layer. When the source node gets congestion information from both layers, then an alternate path which is congestion free is determined to reach the destination node.

Kazuya recommended a routing method that uses two agents known as routing agent and message agent. The routing agent gathersdata about congestion and link breakages in the network. Routing table at every node is then updated by the routing agent with this data.The gathered information is then operated by the message agent to get to their destination.

In various protocols were proposed for routing to find the path between the source and destination to lessen the impact of congestion during data forwarding in MANET (Narasimhan and Santhosh, 2009; Tamilarasan and Eswariah, 2013; Kim *et al*., 2014).

Barreto (2015) described two cross layer congestion control mechanism XCP-Winf and RCP-Winf that exploits the MAC layer information. The link capacity and the bandwidth information is obtained by the exchange of RTS-CTS-DATA-ACK to control the congestion in the network. Zhao *et al*. (2014) have presented an optimization design that yields maximizing multicast utility and minimizing energy consumption. This algorithm controls congestion by regulating the sender's flow rate and removes the bottleneck at the intermediate node by giving power.

Zhang *et al*. (2014) proposed Heuristic Evaluation Postdecision State (HE-PDS) algorithm for packet transmission. According to the feedback of the delay, throughput and channel state information obtained at the receiver, the transmission rate and transmission power are adaptively adjusted at the transmitter.

Chunsheng *et al*. (2014) discussed about directed cooperative path that exploits the reasons for congestion in three aspects, i.e., node, link and channel. According to the congestion degree function, congestion processing is dynamically adaptive. They have also applied QRD (Queue and Rate Dispatch) and QPCG (Queue Transmission Path of Channel Game) to enhance the fairness of the wireless network resource competition, improve the resource utilization of wireless network and increase the wireless network transmission.

Ahmad *et al*. (2014) classified the causes of congestion into NLC (Node Level Congestion) and link level congestion. They proposed a protocol TSEEC (Time Sharing Energy Efficient Congestion Control) based on STDMA (Statistical Time Division Multiple Access) to improve the performance in NLC and LLC which ultimately enhances the energy efficiency of the network. Sheeja and Pujeri (2013) discussed a cross layer based congestion control method that has congestion detection and congestion control phases. They have also used path gain and buffer tenancy fraction to identify the route to forward the packets.

Yuzhou *et al*. (2015) formulated a framework for rate allocation as a Network Utility Maximization (NUM) model based on end to end cross-layer congestion control design. Li *et al*. (2011) recommended a congestion control strategy in which the congestion state dynamically adjusts the probability offorwarding nodes and the rate of generating packets to the network. Karthikeyan *et al*. (2009), the researchers discussed about the clustering mechanism in wireless network to reduce the broadcast overhead. Everymobile node that receives Route Request (RREQ), rebroadcast the request to reach the target during route discovery. To reduce the routing overhead, cluster based flooding method is used by the researchers. They have improved the packet delivery ratio and reduced the broadcast overhead. Density based Probabilistic Flooding Protocol has been applied in (Karthikeyan *et al*., 2010) to reduce the routing and MAC overhead in MANET environment.

From the above-mentioned research works, it is understood that it is vital to differentiate the congestion loss from non-congestion losses. Setting the value for the congestion window plays a major role in utilizing the network capacity. Subsequently, the current network condition is to be known accurately for treating the congestion and non-congestion losses. Packet loss cannot be treated as the only reason for the sign of congestion. In this study, a novel scheme cross layer congestion control with efficient Bandwidth Allocation and Dynamic Window Adaptation (CC-BADWA) is proposed in which when timeouts happen, instead of reducing the congestion window and sending rate, the immediate history of the network parameters are analyzed to set the current network parameters. The link failure is notified to the TCP sender by the intermediate node. When the sender receives this notification message, sender stops packet transmission for a period of time. For successful packet transmission, ACK is expected within the retransmission timer expiration. In the case of no notification or no ACK is received by the sender before the RTO expiration, then the transmission probability used in our method to transmit the packets to be injected into the network. When the receiver window size is optimal compared to receiver window size, then the packet transmission continues with aprobability of 1. The dynamic adjustment of TCP congestion window using spatial-temporal relationship allows a reasonable window size instead of setting to a small value. Bandwidth allocation employs the number of current active flows and its bandwidth usage of a node. The bandwidth allocator assigns the optimum bandwidth based on the current bandwidth of a node. And, the route selection employed in this scheme considers the path with maximum residual energy, maximum bandwidth, minimum mobility for data forwarding. Compared to the other techniques our method dynamically changes the transmission probability, congestion window, bandwidth and path selection according to the network conditions.

Problem definition and motivation: The problem of congestion affects the network performance drastically both in wired and wireless network. While the loss of packet is considered as the sign of congestion in wired network, the picture is not the same in wireless networks. Since the causes like mobility of node, energy drain at the node, interference, link error, MAC contention can cause the packet loss, the reason for loss is to be identified suitably. When, there is no congestion, unnecessary reduction in the congestion window reduces the TCP packet sending rate into the network. The dynamic adjustment of the congestion window, suitable packet sending rate through stable path is required to improve the network performance.

MATERIALS AND METHODS

Proposed method (CC-BADWA): The proposed scheme works as follows:

- Loss of packets due to link failure and congestion is differentiated to adjust the sender side congestion window. Transmission probability is used to forward the packets into the network in case of congestion
- Congestion window is adjusted using spatial-temporal correlation
- Bandwidth allocation is performed based on the current requirement and availability of the bandwidth
- Optimal path selection is made on the energy, mobility and bandwidth of the path

Identification of packet loss from link failure and congestion: Using hello messages and neighbor update timer, every node in the MANET constructs its neighborhood table in its transmission range. When the node is ready to send the data, it checks the availability of the path to the intended destination in its routing table. If the path is readily available, then the data forwarding starts with the available path. If the path is unavailable the source node starts disseminates route request (r_req) messages in the network. Intermediate nodes rebroadcast the messages until the r_reqmessage finds the destination. Upon receiving the r_req message, the destination performs the best path selection and the best path information to the sender is carried in route reply (r_rep) as the unicast message. When the route reply message passes through the intermediate nodes, they create an entry in its routing table. Once the path is established, the data transmission starts flowing from the source to the destination.

The MANET suffers the problem of mobility. When the intermediate nodes move in between an active data flow then the link failure causes the packet loss in the network. In case of link failures, the intermediate nodes try to find an alternate route to the destination to forward the data packet. Further, the acknowledgment message will not reach the sender, making the sender to reduce the congestion window unnecessarily. Upon link failure, if the alternate path to the destination is unavailable the intermediate node starts route discovery process with a timer triggered. If the route reply from the destination or from any of the intermediate nodes is not received before the timer expires then a Notification message is sent to the sender to hold the data transmission for some time. Neither notification nor ACK is received by the TCP sender before the retransmission timer expires, it uses transmission probability to transmit the packets (Table 1).

Calculation of tranmission probability: When the incoming traffic exceeds the available network bandwidth, it leads to congestion. When there is congestion, the packets coming excess of the node's buffer capacity are dropped. In the proposed method, the required bandwidth is calculated and based on this the receiver advertises the receiver window size (AD_WND). The AD_WND received in the ACK packet used by sender toadjust its congestion window.

The bandwidth is measured from the DATA packet interarrival time at the receiver. Then moving

Table 1: Notations

Values	Parametrs
B	Measured bandwidth
S	Packet size in bits
Δt	Average inter-arrival time
Δt_k	Current time
Δt_{k-1}	Time at which previous DATA packet received
B_i	The ith measured bandwidth
α	Exponential filter coefficient (value is 0.9) from TCP-Westwood
AV_BUF	Available buffer at receiver
RCWD	Receiver's congestion window
S	Source node
D	Destination node
IN	Intermediate Node
D1, D2	Data packet
ACK	TCP acknowledgement packet
BW	Recent bandwidth advertised by the receiver in TCP ACK
maxBW	Maximum bandwidth used
BLen	Recent buffer length of the path
maxBLen	Maximum buffer Length
currentwindow	Current congestion window
maxWND	Maximum window size
maxADwnd	Receiver advertised window
AD_{WND}	Maximum receiver's advertised window
α_1-α_4	Random variables (0,1)
BL	Buffer length at intermediate nodes
rwnd	Receiver advertised window
n	Time duration
CW[i]	Congestion window at ith iteration
ECN	Explicit congestion notification
avg	Average
std_dev	Standard deviation
var	Variance
expval_SR	Expected value of spatial relation
expval_TR	Expected value of temporal relation
STC	Spatial temporal correlation
CUR_CWND	Current congestion window
β_1, β_2	Random variables (0,1)
BW_i	Bandwidth at time i
average_BW	Average bandwidth
max_BW	Maximum bandwidth
border_BW	Border bandwidth
CE	Current energy of the node
IE	Initial energy of the node
PMBW	Path maximum bandwidth
maxBW	Maximum bandwidth
max_speed	Maximum speed of a node
speed	Current speed
RREQ	Route request
RREP	Route reply

average method is used to continuously update the available bandwidth to find the congestion window for the sender:

$$B = \frac{S}{\Delta t} \quad (1)$$

$$B = \alpha B + (1-\alpha) B_i \quad (2)$$

At the receiver side, the Round Trip Time (RTT) is calculated as the time interval between the last acknowledgement sent and the time at which the DATA packet arrival time after the last ACK as shown in Fig. 1.

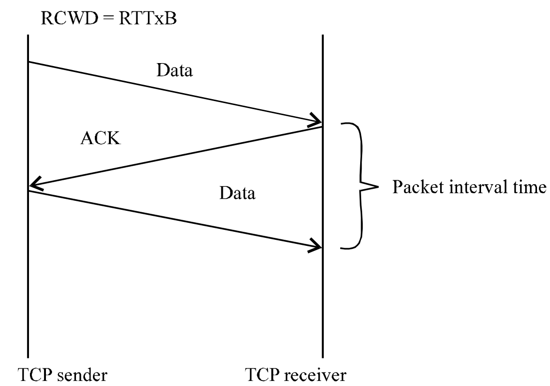

Fig. 1: Packet interarrival time

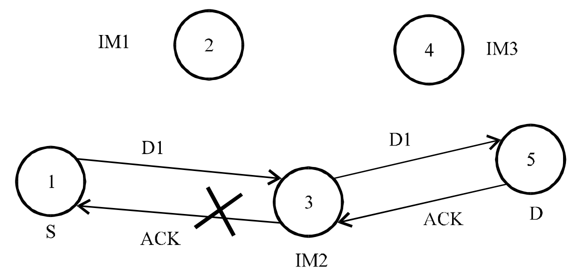

Fig. 2: Loss of ACK packet

When the transmission utilizes the full network capacity (bandwidth delay product) and the receiver is capable of receiving all the packets, then the receiver congestion window size is given by:

$$RCWD = RTT \times B \quad (3)$$

The receiver now computes the advertised window as the minimum of receiver's Available Buffer size (AV_BUF) and the Receiver Window (RCWND) and sends this value in ACK packet to the sender:

$$AD\_WND = \min(AV\_BUF, RCWD) \quad (4)$$

The TCP sender compares this AD_WND with the current window (CUR_CWND). If the CUR_CWND is less than the AD_WND, it representsthe network is not congested. When the advertised window is less than the current window, then the TCP's sending rate is reduced. (Flow and congestion control).

In Fig. 2, the source node S sends packets through the intermediate node IM_2 to the destination node D. For the data received, the destination calculates the bandwidth using the packet inter-arrival time and the receive packets in bits. Using the bandwidth and the

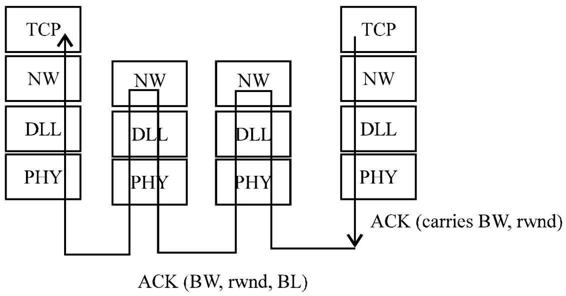

Fig. 3: The ACK from receiver to sender through layered architecture

RTT the destination calculates the receiver advertised window. The estimated bandwidth and AD_WND is sent to the sender using the ACK packet. As shown in Fig. 3, the ACK packet is lost in IM_2 due to congestion. Eventually, S does not receive the ACK for packet when the retransmission timer time outs. Now, the source node estimates the transmission probability TP as given below.

The transmission probability depends on the usage of bandwidth, queue length at the intermediate nodes, window size at the sender, the receiver advertised window. Transmission Probability (TP) is given by:

$$TP = \alpha_1\left(\frac{BW}{\max BW}\right) + \alpha_2\left(\frac{BLen}{\max BLen}\right) + \alpha_3 + \exp\left(\frac{\text{current window}}{\max WND}\right) + \alpha_4 \exp\left(-\left(\frac{AD_{WND}}{\max ADWND}\right)\right) \quad (5)$$

The bandwidth is measured by the receiver and it returned back to the sender in the ACK packet along with the receiver advertised window. Also, the average buffer length of the path is determined by the ACK packet. The ACK packet travels through the intermediate nodes of the path collecting the buffer length at them and finally, estimating the average buffer length at the sender (Fig. 3).

If AD_WND is greater than the CUR_CWND, then the probability of packet transmission TP = 1. Otherwise, the transmission probability is used to determine the probability for packet transmission. When the time-out happens at the receiver, the receiver sends the ACK with ECN flag set and send it to the sender. Then, the proposed method dictates the sender either to hold the packets or to transmit packets based on the severity of the congestion in the network.

CW_0	CW_1	CW_2	CW_3	...	CW_{n-2}	CW_{n-1}

Time interval t (sec)

Fig. 4: Congestion widow values over time period t

The parameters chosen for transmission probability ascertains the level of congestion in the network because the recent values of the above network parameters clearly specify the degree of congestion in the network. Hence, instead of considering the packet loss as anindication of congestion, these parameters are considered as the valid indications.

Dynamic congestion window adaptation in CC-BADWA: As the transmission probability is calculated, the size of the sender congestion window is dynamically adapted using Spatial-Temporal Correlation (STC). The Congestion Window (CW) size over a time period (t) is used to calculatethe STC value. Using the average of congestion window, the expected value of spatial and temporal relations are estimated. The CUR_CWND size is set accordingly. Figure 4 depicts the array that stores the values of congestion window over a time interval t.

$$\begin{aligned} avg &= \frac{\sum_{i=0}^{n-1} CW[i]}{n} \\ std\_dev &= \frac{\sqrt{(avg - CW_i)^2}}{n} \\ var &= (std\_dev)^2 \\ \exp val\_SR &= avg + var \end{aligned} \quad (6)$$

The expected value for the temporal relation of the congestion window is estimated by considering the contiguous sets of congestion window values. Here, the set value is set to 10. The n_sets = Tot_window/set_value set_value->10:

$$\begin{aligned} std\_dev &= \frac{\sqrt{(avg - CW_i)^2}}{n\_sets} \\ var &= (std\_dev)^2 \\ \exp val\_SR &= avg + var \\ STC &= \beta_1 SR + \beta_2 TR \end{aligned} \quad (7)$$

$$CUR\_CWND = STC \quad (8)$$

The current CWND at the TCP sender is set to the value of the calculated Spatial-Temporal Correlation (STC). The STC provides a reasonable value to the CUR_CWND instead of reducing it to 1 in thecase of timeout at the sender side.

Bandwidth Allocation: The CC-BADWA deploys a bandwidth allocator table at every mobile node to satisfy the bandwidth requirement of the current flow. This table contains the bandwidth usage details of the transmission flows. The attributes of the table are <flow ID, bandwidth_usage, time>. Flow ID specifies the identifier of the flow of packets, bandwidth_usage specifies the bandwidth allocate for a flow and the time designates the time instance of the bandwidth usage.

The sender assigns the available bandwidth to multiple users in a rational manner. It calculates the average bandwidth usage of all flows and assigns the available bandwidth based on the usage requirement of the applications. The sender determines the current usage of bandwidth based on the previous usage which is stored in the table:

$$\begin{aligned} \text{avgerage\_BW} &= \frac{\sum_{i=0}^{n-1} BW_i}{n\_sets} \\ \text{Max\_BW} &= \max\left(BW_0^{n-1}\right) \\ \text{border\_BW} &= \frac{(\text{average\_BW} + \text{Max\_BW})}{2} \end{aligned} \tag{9}$$

For any node i, if BW_i<average_BW, then its current bandwidth usage is less than the average bandwidth. These nodes are allocated bandwidth in the range of aaverage_BW and the border_BW. For the node i, if BW_i>average_BW, then its bandwidth requirement is more. These nodes are allotted bandwidth in the range of border_BW and Max_BW:

if BW_i<average_BW then
random.uniform (average_BW, border_BW)
else
    random.uniform (border_BW, max_BW)

Thus, the bandwidth allocator proposed in this method provides the available bandwidth to the current flow in an efficient manner. Because, the node can provide the bandwidth ranging from average_BW to max_BW. If the current requirement is less than the average_BW, then the node can give more than sufficient bandwidth and also if therequirement is greater than average_BW, then it can support bandwidth upto max_BW.

Route optimization: The path between the sender and the receiver is affected by the frequent mobility of the nodes, bandwidth availability, the residual energy of the intermediate nodes, etc. An optimized route in CC-BADWA is chosen with minimum mobility, high bandwidth availability and the high residual energy among the identified paths.

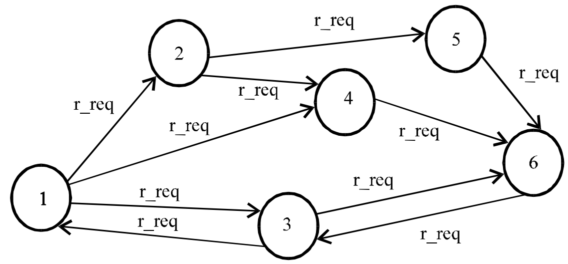

Fig. 5: Route discovery phase

When the route request messages are broadcast in the network, the intermediate nodes append its energy level, available bandwidth and the speed. The destination after collecting the route request (r_req) messages calculates the param value for the received request messages. The highest param implies more optimized route among multiple request messages. The destination node unicasts the route reply (r_rep) with the highest param value determined. The intermediate nodes receiving the reply message will make an entry in the routing table for the data transmission.

In Fig. 5, the source node 1 initiates a Route Discovery by disseminating the route request (r_req) packets in the network. The r_req when traverses the intermediate node adds the node's residual energy, available bandwidth and the speed of the node on its path to the destination node 6. The destination node upon receiving multiple r_reqs, calculated the param value as follows:

$$\text{Path param} = \max\begin{bmatrix} \alpha_1 \dfrac{CE}{IE} + \alpha_2 \dfrac{PMBW}{maxBW} + \\ \alpha_3 \exp\left(\dfrac{max\_speed}{speed}\right) \end{bmatrix} \tag{10}$$

Using the above Eq. 10, the path with maximum energy, maximum bandwidth and minimum mobility is chosen for data transmission with the mentioned transmission probability.

Proposed Algorithm (CC-BADWA): The proposed algorithm estimates the transmission probability and the spatial temporal correlation for the sender congestion window, when there is no notification of the link failure or no ACK packet.

The transmission probability TP is set to 1 when AD_WND>CUR_CWND; otherwise the calculated TP value is used for transmission. The bandwidth allocation is performed at every intermediate node such that it estimates the average and border bandwidth using the bandwidth usage over a period of time. The bandwidth allocator allocates the required bandwidth consistent with the current bandwidth usage. The optimal route is determined by the receiver using the parameters bandwidth, residual energy and mobility of the node in the path. The algorithm for the proposed scheme is given.

Algorithm 1; cross layer bandwidth allocation and route optimization:

At the sender

Step 1: TCP sender sends the DATA packet
Step 2: For every DATA packet, TCP sender
    Starts the retransmission timer
        wait (ACK)
Step 3: if Link failure notification received
        stop transmission and
        wait for route reestablishment(predefine time)
            if (no route discovered)
                initiate route discovery
    else if no notification and no ACK
Step 3.1 : calculate TP
    if (rwnd>cwnd)
        TP = 1
    Else
$(TP) = \alpha_1(BW/maxBW)+\alpha_2(BLen/maxBLen)+\alpha_3\exp(current\ window/maxWND)+\alpha_4\exp(-(AD_{WND}/maxAWND))$
Step 3.2: adjust the cwnd using STC
    $cwnd = \beta_1 SR+\beta_2 TR$
Step 3.3: Find the optimal route
$Param = \max[\alpha_1 CE/IE+\alpha_2 PMBW/Max.BW+\alpha_3\exp(-Max\_speed/speed)]$
Step 3.4: Perform the bandwidth allocation
    if BW_i<average_BW
        random.uniform(average_BW, border_BW)
    else
        random.uniform(border_BW,max_BW)
else ACK received
    cur_cwnd= min(cwnd_, rwnd)

At the receiver

Step 1: Calculate bandwidth
    BW= Packet size/time interval
Step 2: Calculate RTT
    RTT = Last_ACK_time – new DATA arrival time
Step 3: rwnd = BW*RTT
Step 4: ad_wnd = min (avail_wnd,rwnd)
Step 5 : Send ACK
    append (BW,ad_wnd) in ACK

Performance evaluation

Simulation parameters: The proposed method is simulated in ns 2 with 50 nodes, MAC type as 802.11, packet size with 1000 bytes in 1000×1000 topology area. The simulation results of CC-BADWA are compared with the existing algorithm Receiver Assisted Congestion Control (RACC).

Performance parameters

Packet delivery ratio: It is the fraction of successfully delivered packets to the receiver to the total number of packets produced by the sender.

End to end delay: It the time taken by the packet to arrive at the destination from the sender.

Throughput: It is defined as amount of data transferred over communication channel in a span of time. Generally measured in bits per sec.

Normalized overhead: Normalized overhead is the ratio of the amount of control packet sent to the amount of data packets received.

Jitter: Jitter is defined as the difference in the time variation of the received packets.

RESULTS AND DISCUSSION

Figure 6 compares the packet delivery ratio of the existing and proposed methods. The proposed method uses the bandwidth allocation based bandwidth usage of the traffic flows over a period of time. The bandwidth allocator assigns available bandwidth efficiently to be utilized by the current incoming flow. Hence, the incoming packets get optimal bandwidth which results in faster transmission along the route. And also, the proposed route optimization provides the path with maximum energy, maximum bandwidth and minimum mobility is chosen for data transmission. Since, the minimum mobility path is chosen, the probability for packet loss due to link failure is reduced which in turn reduces the number of retransmissions. Ultimately, the transmitted packets reach the destinations successfully in the proposed method.

End to end delay: The CC-BADWA uses the route selector that uses the low mobility path to forward the data packets. The selected path results in fewer link

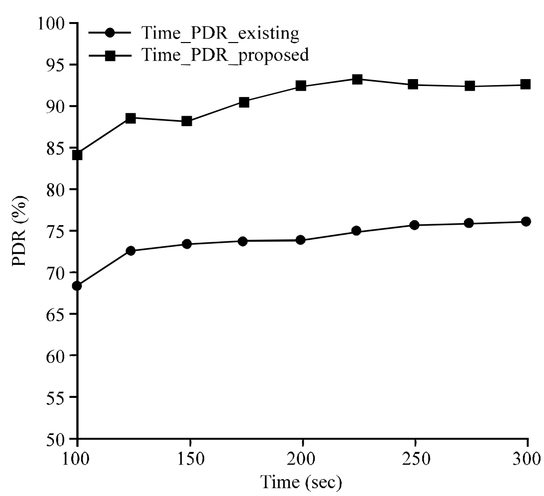

Fig. 6: The PDR analysis

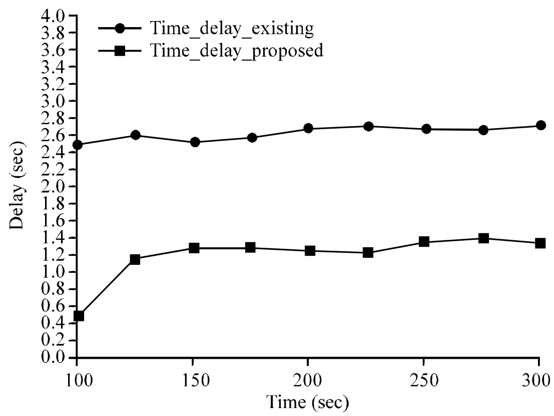

Fig. 7: The delay comparison

breakages. In the case of frequent link failures, more route discoveries are to be performed. In those cases, the packets are queued in the interface queues of the intermediate nodes that increase the delay. But, the proposed method, the selected path has minimum mobility that has less link breakages ultimately minimizes the delay. And, the bandwidth allocator in CC-BADWA suits the current condition of the network the bandwidth requirement for the data transmissions. The sufficient bandwidth allocation for the incoming packets reduces the delay time of the packets at the intermediate nodes. Figure 7 illustrates the delay comparison of existing and proposed methods.

Throughput: Figure 8 analyzes the throughput performance of the existing and proposed methods. Since, the receiver knows the congestion status earlier than the sender, it estimates the available bandwidth, the size of the congestion window. Instead of simply reducing the congestion window in case of no acknowledgment and no notification, the sender uses the proposed packet transmission probability to transmit the packets in the network. The congestion window is also set using the spatial temporal correlation. The dynamic adaptation of the congestion window improves the sending rate of the source. If the recv_wnd is optimal to the maximum recv_wnd over a time period, then the packet transmission is continued with probability 1 because there. Otherwise, the packet transmission uses Eq. (TP) to generate packets. This reduces unnecessary reductions in cwnd and the throughput. Compared to the existing method, the proposed method achieves better performance of throughput.

Normalized overhead: In the proposed method, the congestion and wireless losses are differentiated with the help of dynamic congestion window adaptation. The less

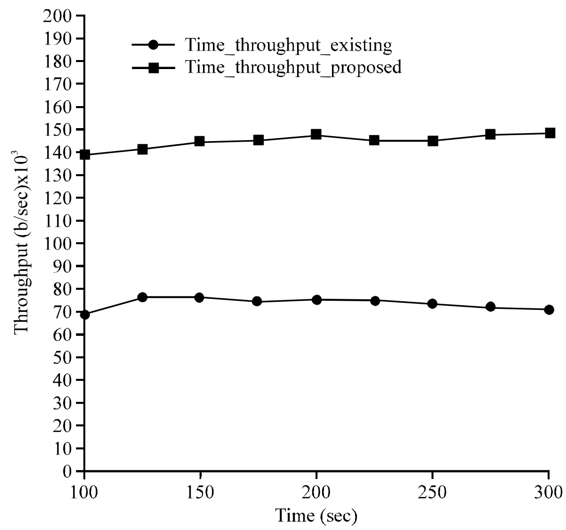

Fig. 8: Throughput analysis

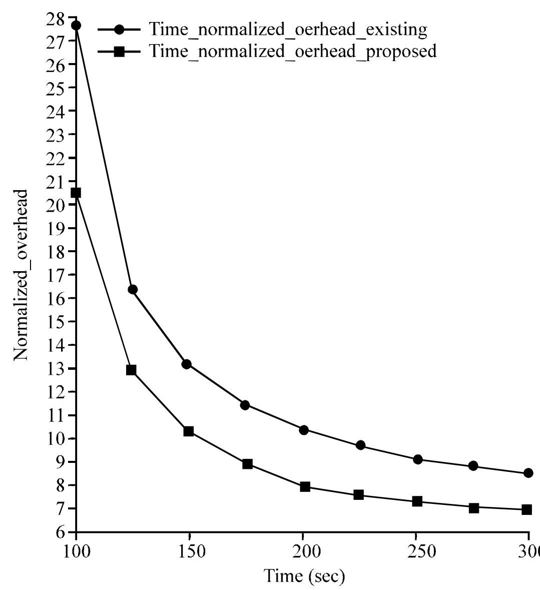

Fig. 9: Normalized overhead

mobility path used in the proposed method results in less route discoveries in the case of route failures. Thus, the control overhead in the network is reduced. In the case of timeout and congestion, the sender uses the transmission probability to forward the packetsto the destination. Subsequently the amount of successfully transmitted packets in the network increases.

The recent values of the network conditions used in the Transmission probability and the route selection predict the current condition of the network and thus improves the rate of successful packet transmissions in the network. Thus the normalized overhead is reduced as shown in Fig. 9.

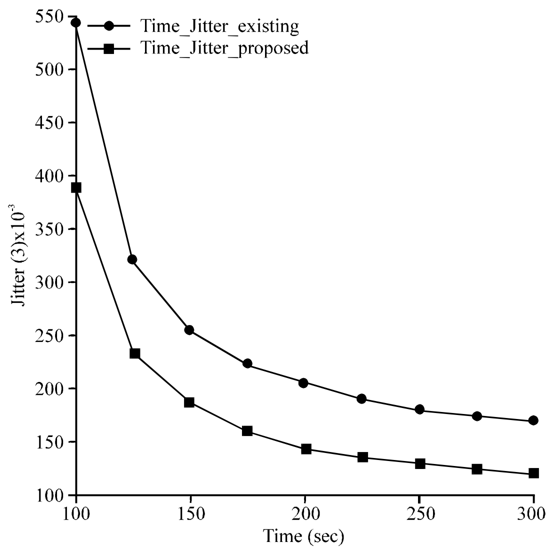

Fig. 10: Jitter analysis

Jitter: Figure 10 compares the jitter delay of the existing and the proposed method. The proposed method clearly identifies thecause of the packet loss and invokes the transmission probability in thecase of no acknowledgment and congestion. The congestion window is set to the spatial-temporal correlation of cwnd values. So, there is aflow of packets in the network and also the proposed method quickly responds to the network congestion and link breakages the selected path. So, the jitter incurred in the received packets decreases compared to the existing algorithm.

CONCLUSION

The proposed cross-layer congestion control and dynamic congestion window adaptation method (CC-BADWA) takes into account the cross layer information from MAC layer and the network layer to improve the network performance. This method uses the receiver measured bandwidth and receiver's advertised window to set the congestion window of the TCP sender. The Link failure notification and the prompt ACK helps the sender to stop the transmission or proceed the transmission successfully. For each data transmission, if the source node does not receive the acknowledgment due to congestion, then it estimates the transmission probability that uses the measured bandwidth, average buffer length of the path, sender window size and the receiver advertised window. During this time, sender's window is set to the value of optimal window value which is calculated using Spatial-Temporal Correlation. The bandwidth allocation is done using the node's average usage of bandwidth and with border bandwidth value. The optimal route selection is carried out by estimating the path with maximum residual energy, maximum bandwidth and low mobility.

The proposed CC-BADWA method predicts the congestion status in the network and accordingly it injects packets into the network. The route selector chooses the path low mobility which reduces the path breaks and ultimately reduces the control overhead packets. The result study shows the performance improvement in CC-BADWAcompared to the existing method.

As future research, the effect of multipath routing and out of order delivery of packets in which the duplicate acknowledgements are sent to the TCP sender can be considered. The fast retransmit sends the missing packet and reduces the cwnd which can be dynamically set depending on the current network parameters. Also, the effect of wireless errors can be considered for the packet loss in wireless environment.

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