QOS BASED BANDWIDTH SATISFACTION FOR MULTICAST NETWORK CODING IN MANET

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ABSTRACT-*Wireless links are lossy due to network traffic and fading channel. To avoid the packet loss in this method duplicate packet is forwarded in the link to get the acknowledgement from a receiver. Multipath routing trees are constructed for the transmission which providing the bandwidth guarantee. With the randomized network coding, the probability of overhearing the same packet transmission by different forwarder is very low. The proposed multicast routing protocol is designed based on contention-based media access control (MAC) protocols. Scheduling algorithm is not separately needed for distributing packets among the constructed trees. Estimation of the bandwidth before transmission is obtained using a variable bit rate. To improve the accuracy of system and earlier estimation of the bandwidth, multicast routing protocols are used. This method reduced the delay and retransmission rates and improve the bandwidth usage in the MANET*.

Key Words: *MANET, QOS, HRP, HMRP.*

1. INTRODUCTION

A mobile ad hoc network (MANET) enables wireless communications between participating mobile nodes without the assistance of any base station. Two nodes that are out of one another's transmission range to need the support of intermediate nodes, which a relay messages to set up a communication between each other. The most fundamental role in MANET is the broadcast operation because of the broadcasting nature of radio transmission: When a sender transmits a packet, all nodes within the sender's transmission range will be affected by this transmission. The advantage is if one node transmits a packet, all its neighbors can receive this message. In the negative side, one transmission may interfere with other transmissions, creating the exposed terminal problem where an outgoing transmission collides with an incoming transmission and the hidden terminal problem where two incoming transmissions collide with each other.

Some existing QoS-guaranteed multicast protocols for MANETs assume ideal links. However, links are lossy in real wireless networks. The packet loss ratio, which represents the probability that a packet is lost or erroneous, depends on the channel quality. When the packet loss ratio of a lossy link increases, more bandwidth consumption for retransmission is required .Since real wireless links are lossy, it is practically important to consider lossy MANETs when we are designing a bandwidth-satisfied multicast protocol. It is a great challenge to design a multicast routing protocol for lossy MANETs that can provide bandwidth guarantees, avoid redundant packets, and reduce the bandwidth consumption at the same time. Providing bandwidth guarantees to traffic flows may incur lots of redundant packet due to lost packets. No bandwidth-satisfied unicast/multicast routing protocol for lossy MANETs was proposed MANETs suffer from a high transmission error rate because of the high transmission contention and congestion. Furthermore, it is a major challenge to provide high reliability for broadcasting operations under such dynamic MANETs.

2. EXISTING SYSTEM

In RACC mechanism, the receiver not only performs the function of flow control, but also participates in the congestion control. It first measures the bandwidth, and then computes an appropriate congestion window size based on the measured bandwidth and the RTT. To perform these functions, the receiver has to maintain two timers: one timer for recording the packet inter-arrival interval and the other for measuring the RTT. The sender makes use of this information about its receiver to adjust the congestion window (Xu 2006).

2.1 FUNCTION OF RECEIVER

The bandwidth is measured from the receiver according to the packet inter-arrival interval. This method can remedy the oscillation in the estimation of bandwidth of TCP Westwood (Casetti 2002).

Let B_w be the measured bandwidth, L be the data packet size, and t_{int} be the packet inter-arrival interval. Then, one can estimate the available bandwidth by

$B_w = L/t_{int}$ for each packet arrival. Moving average method is used within each congestion window. Let B_w be the i-th measured bandwidth. Then, the bandwidth can be continuously updated by below the equation (3.1).

$$B_w=\alpha B_w+(1-\alpha)B_{wi} \quad (3.1)$$

Where α is an exponential filter co-efficient, α =0.9 is a good value because the former averaged values should have a higher weight (0.9 in this mechanism) to lower the measure variations. To reduce the consequence of cross traffic, a low pass filter can also be adopted.

The receiver measures the RTT based on the data packet arrival time. A variable rcv.rtt is used to record the RTT. The algorithm can be described as follows.

i) When an ACK is sent, if rcv.rtt is zero, let rcv.rtt equal 1 and record the corresponding sequence (rtseq) of data packets (equals to the sum of the ACK sequence and the current congestion window). Otherwise, just send the ACK.

ii) If a data packet of a larger sequence number than "rtseq" is arrived in order and the new measured packet inter-arrival interval is larger than two times of pre-estimated packet inter-arrival intervals, set the new measured RTT to the value of rcv.rtt, and let rcv.rtt be zero.

iii) On each clock cycle, if rcv.rtt is not zero, rcv.rtt is added by 1. Then, the retransmission timer is setting based on the measured RTT.

Next, the receiver uses RTT to convert the bandwidth to the receiver congestion window value rwnd using the below equation (3.2)

$$rwnd=B_w*RTT \quad (3.2)$$

Then, it analyses the rwnd with the available receiver buffer, and deposits the lesser value

of the advertised window field of an ACK going back to the sender. It is shown in the below equation (3.3)

$$adv\_wnd=\min(r\_abuf,rwnd) \quad (3.3)$$

In other words, the receiver advertised window not only has the original flow control function, yet also takes on the congestion control function. When the receiver detects a timeout, it will send an ACK to and inform the sender that the network is congested. In this ACK, the receiver advertised window is set to one packet, meaning that the sender must retransmit the dropped packet. As the receiver must have detected this timeout earlier than the sender, it will help the sender to reduce the waiting time and confirm the packet loss.

2.1.1 MANAGING THE RTO TIMER

An implementation should manage the retransmission timer(s) in such a way that a segment is never retransmitted too early, i.e. less than one RTO after the previous transmission of that segment.

The following algorithm is recommended for managing the retransmission timer:

1. Every time a packet containing data is sent, if the timer is not running, start it running so that it will expire after RTO seconds (for the current value of the RTO).
2. When much outstanding data has been acknowledged, turn off the retransmission timer.
3. When an ACK is received that recognizes new data, restart the retransmission timer so that it will expire after RTO seconds.

When the retransmission timer expires, do the following:

4. Retransmits the earliest segment that has not been acknowledged by the TCP receiver.
5. The host must set RTO <- RTO * 2 ("back off the timer"). The maximum value may be used to provide an upper bound to this doubling operation.
6. Start the retransmission timer, such that it expires after RTO seconds (for the value of RTO after the doubling operation.

The TCP implementation may clear SRTT and RTTVAR after backing off the timer multiple times as it is likely that the current SRTT and RTTVAR are bogus in this situation. Once SRTT and RTTVAR are cleared they should be initialized with the next RTT sample taken.

2.2 REACTION OF SENDER

As the receiver can timely help the sender to increase the congestion window according to the instantaneous available bandwidth, the sender only needs to maintain the AIMD mechanism in the congestion avoidance stage, and the slow start stage can be eliminated. Upon a timeout event (whether it is detected by the sender's timer or informed by the receiver's ACK), the sender will decrease the congestion window to one in consideration that the network is in congestion (Srinivas 2010) and it will have some time to recover. If congestion is mitigated after one RTT, the sender will recover to adjust the congestion window in the next window by using the receiver advertised window. During fast retransmission, the sender sets the congestion window size of the lesser value of the receiver advertises window size and the current size. Since the packet loss may also indicate congestion, we should reduce the congestion window. On the other hand, if the congestion window is less than the receiver advertised window, the network may only be in a mild congestion (Afanasyev 2010).

Therefore, it is unnecessary for the sender to reduce the congestion window. In a normal state, when the sender receives an ACK, it will compare the current congestion window with the receiver advertised congestion window. If the receiver advertised window is larger than the sender's congestion window, and the difference is larger than a predefined threshold, the sender will set the congestion window size to the receiver advertised window. Or else, the sender will ignore the receiver advertised congestion window by just performing the additive increase mechanism. The threshold value of TCP Vegas (Brakmo 2001) is used.

3.METHODOLOGY

A multicast routing protocol for lossy MANETs is proposed. By means of constructing a multiple multicast trees (trees for short) and transmitting coded packets of randomized network coding the proposed multicast routing protocol can provide bandwidth guarantees, avoid redundant packets, and reduce the total bandwidth consumption to a certain degree. With the randomized network coding, the probability that two forwarders overhearing the same packets transmit the same coded packets is very low.

The proposed multicast routing protocol is designed based on the contention-based media access control (MAC) protocols. Scheduling algorithm is not separately needed for distributing packets of the constructed trees. In order to provide bandwidth guarantees to multicast destinations, the bandwidth that each source-to-destination route can provide must be aware. The forwarders can provide the minimum bandwidth for the route. To determine the bandwidth that a forwarder can provide, its residual bandwidth and bandwidth consumption are estimated by the proposed multicast routing protocol. Moreover, its one-hop and two-hop neighbors are also examined in order to prevent the hidden route problem (HRP) and the hidden multicast route problem (HMRP)

3.1 NEW BANDWIDTH-SATISFIED MULTICAST ROUTING PROTOCOL

In this section, we intend to propose a multicast routing protocol for lossy MANETs that can reduce the total bandwidth consumption to a certain degree while providing bandwidth guarantees to a requested flow and ongoing flows. Multiple multicast trees are constructed so that the residual bandwidth can be fully utilized and the bandwidth consumption can be reduced. In addition, the randomized network coding is applied so that redundant packets can be avoided and each destination can receive innovative coded packets of distinct routes in the proposed multicast routing protocol.

A tree construction algorithm is proposed to construct a multicast tree at a time. It selects hosts of better channel conditions as forwarders to reduce the total bandwidth consumption. Each constructed tree connects some destinations, and each can provide a predefined percentage of the bandwidth requirement of each connected destination. The tree construction will continue until all destinations are bandwidth satisfied. There is a basic procedure contained in the proposed algorithm.

3.2 TREE CONSTRUCTION ALGORITHM

Given a requested flow Γ, the tree construction algorithm can construct one or more trees. Sequentially, to provide bandwidth guarantees to the destinations of Γ and to reduce the total bandwidth consumption. There are four input parameters, i.e., hs, D, b ¨ _req, and b_per, for the algorithm, where hs is the source, D¨ is the set of all destinations of Γ, b_req is the bandwidth requirement of each destination, and b_per is a predefined percentage for the bandwidth requirement (i.e., b_req). There are three sets D, D , D of destinations, one set F of forwarders, and a series of 2-D arrays B1, B2,...,Bϕ, B˜ used in the algorithm, where ϕ is the number of trees constructed. Given a host hi and a destination hd, Bt [i, d] records the bandwidth of hi that is consumed for hd in the t^{th} constructed tree, where $1 \le t \le \phi$, whereas B˜[i, d] accumulates the bandwidth of hi that is consumed for hd in all constructed trees.

3.3 ESTIMATION OF AVAILABLE BANDWIDTH

The available bandwidth estimation is needed in wireless network because there is no control in the flow of data and gets affected due to the interference such as network traffic, link failures and the channel may either over-utilized or underutilized. Therefore, a deterministic approach to provisioning service generally overestimates the actual resource needs, resulting in a low utilization of network resources. The majority of QOS solutions are loaded with capability for measuring the jitter, delay, available bandwidth. The term available bandwidth can be defined as the maximum throughput that can be transmitted on one link or between two neighbor nodes without influence on any existing flow in the network. The bandwidth reservation solutions designed to 802.11 based ad hoc networks use stochastic estimations of available bandwidth. Stochastic bandwidth estimations of random service are based on inferring an unknown bounding function of measurements of variable bit rates (VBR) probing traffic. It is one of the processes to monitor the channel usage based on the CSMA scheme for IEEE 802.11. However, all the existing approaches do not consider the random factors in bandwidth estimations. The proposed system is to improve the accuracy of available bandwidth estimations by considering the random factors such as

4.2 SIMULATION TOPOLOGY

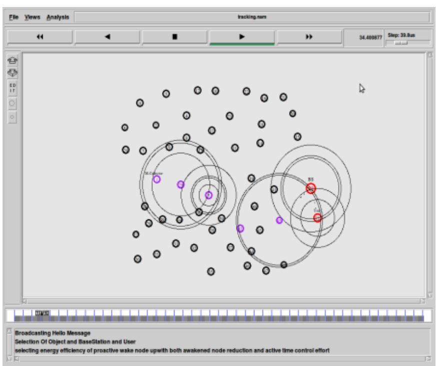
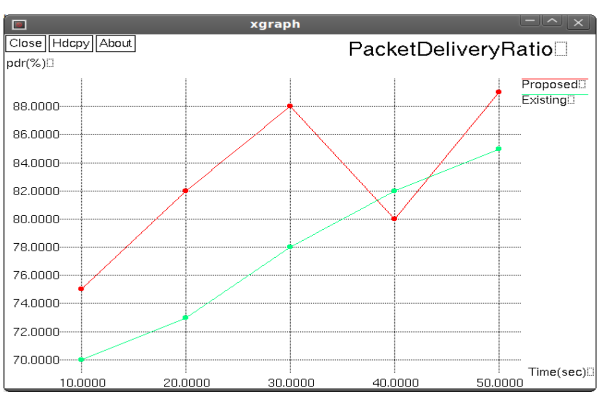


Fig 4.1.1 Link between Source and Destination node

The Fig.6.1.1 shows the simulation topology. Here heterogeneous network is considered. The 5 source nodes are connected to the router by means of wired connection. Similarly the receiver nodes are connected to means of the wired connection. The connection between the routers is wireless. The bursty traffic is considered to analyze the congestion effects.

4.3 SIMULATIONRESULTS

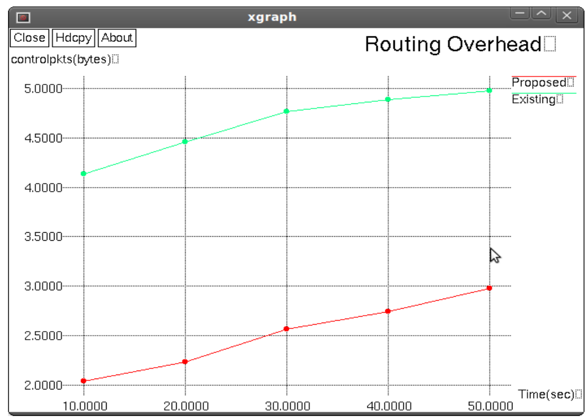


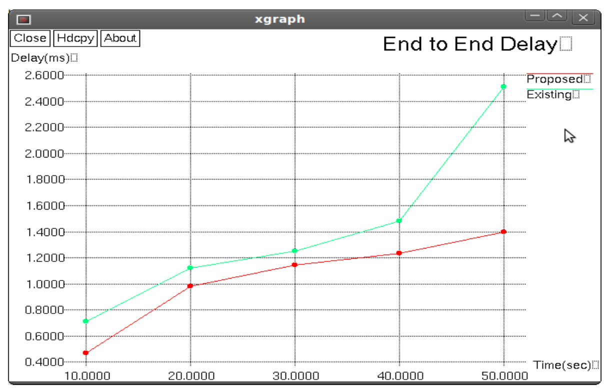
incoming rates; outgoing rate and packets type consideration (e.g., control packets and data packet).

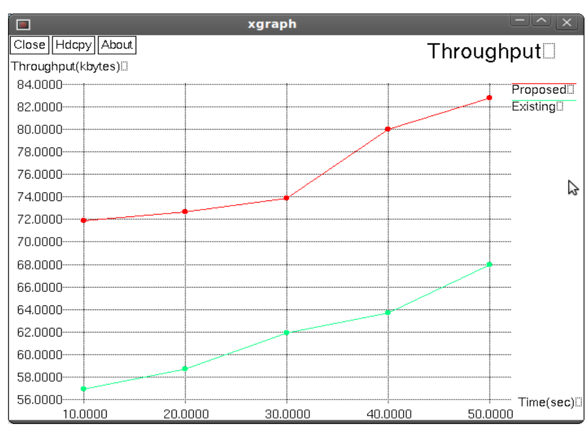
4.SIMULATION RESULTS AND ANALYSIS

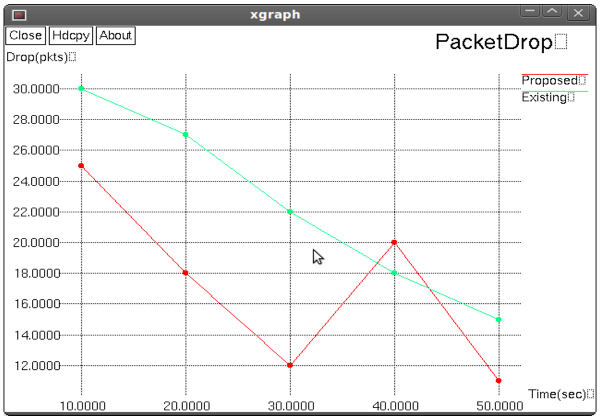
4.1 SENSITIVITY TO NETWORK SIZE

The network has low mobility , where V_{max} is 1 meter per second (m=s), and low transmission error rate (Perr = 1%). The data traffic load CPR is 10 packets .









5.CONCLUSION

Available bandwidth measures the quantity of the maximum throughput of a link. It is used to transmit data without disrupting any existing flows in the networks. The inaccurate estimation of the bandwidth of the link in wireless multi-hop networks leads to wireless networks performance degradation. For lossy MANETs, duplicate packets were transmitted on multiple routes to enhance reliability, which consumed extra bandwidth. The proposed multicast protocol can avoid the HRP/ HMRP while providing bandwidth guarantees to a requested flow and ongoing flows. There is no redundant packet generated by this multicast protocol as a consequence of using the randomized network coding.

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