A Cross Layer Solution to Address TCP Intra-flow Performance Degradation in Multihop Ad hoc Networks

Ehsan Hamadani and Veselin Rakocevic

***Abstract*— Incorporating the concept of TCP end-to-end congestion control for wireless networks is one of the primary concerns in designing ad hoc networks since TCP was primarily designed and optimized based on the assumptions for wired networks. In this study, our interest lies on tackling the TCP instability and in particular intra-flow instability problem since due to the nature of applications in multihop ad hoc networks, connection instability or starvation even for a short period of time can have a negative impact on the Quality of Service and may not be acceptable for the end user. Through a detailed analysis, it will be shown that the main causes of TCP intra-flow instability lies in overloading the network by sending more packets than the capacity of the channel. Based on this, the paper proposes a novel cross layer solution called "TCP Contention Control" that dynamically adjusts the amount of outstanding data in the network based on the level of contention experienced by packets as well as the throughput achieved by connections. The simulation results show TCP Contention Control can drastically improve TCP stability over 802.11 multihop ad hoc networks.**

***Index Terms*– Contention, intra-flow instability, multiple ad hoc networks, TCP.**

I. INTRODUCTION

Multihop ad hoc networks are collection of wireless nodes dynamically forming a temporary network without the use of any preexisting network infrastructure or centralized administration. Consequently, ad hoc networks are fundamentally different from conventional stationary wireless and wired computer networks. During recent years, ad hoc networks have attracted considerable commercial and research interest. In particular, the de facto adoption of the popular IEEE 802.11 standard [1] has further fuelled the deployment of wireless transceivers in a variety of computing devices such as PDAs by ensuring inter-operability among vendors thereby aiding the technology's market penetration. However, as initially the deployment of these wireless technological advances came in the form of an extension to the fixed LAN infrastructure model, the 802.11 standard was mostly evolved and optimized for infrastructure-based wireless LANs rather than ad hoc networks.

To enable seamless integration of ad hoc networks with the Internet, TCP seems to be the natural choice for users of ad hoc networks that want to communicate reliably with each other and with the Internet. Here also, despite the fact that in theory TCP should not care whether the network layer is running over wired or wireless connections, in practice, this does matter because TCP has been carefully optimized based on assumptions that are specific to wired networks. For instance, since bit error rates are very low in wired networks, nearly all TCP versions assume that packet losses are due to congestion and therefore invoke their congestion control mechanism in response to such losses. On the other hand, because of wireless medium characteristic and multihop nature of ad hoc networks, such networks exhibit a richer set of packet losses, including medium access contention drops, random channel errors and route failure where in practice each are required to be addressed differently. Ignoring these properties of wireless ad hoc networks can obviously lead to poor TCP performance as shown in previous research studies (e.g. [2]–[10]).

Not surprisingly, multihop ad hoc networks exhibit serious performance issues when TCP runs over IEEE 802.11 as neither TCP nor IEEE 802.11 MAC have been designed based on the properties of such networks. Therefore, during the recent years, a number of research studies have highlighted some of the problems TCP encounters in ad hoc networks [4], [10]–[18]. However, one of the key areas that has not attracted enough attention and needs to be addressed further in the deployment of TCP in ad hoc networks is the problem of TCP instability where the receiver (data sink) does not receive any packets for a period of time and therefore the connection throughput drops to zero or fluctuates rapidly. In particular, due to the nature of scenarios in which ad hoc networks are used (e.g. emergency operation and battlefield communication), disconnectivity or starvation even for a short period of time can have a devastating impact on the Quality of Service and may not be acceptable for the end user. In other words, ad-hoc network users are likely more willing to receive a continuous and stable flow of data rather than sending/receiving large bulk of data instantly.

In general, TCP instability can be broken down into two broad categories named as *TCP intra-flow* and *TCP inter-flow* instability, where the former is caused by the interaction of nodes belonging to the same TCP connection, while the latter happens when nodes belonging to different connections interact. Due to complexity and different nature of TCP intra-flow and inter-flow instability, this paper only investigates the TCP intra-flow instability problem in fine details.

The rest of this paper is organized as follows: section II analyzes the underlying cause of TCP intra-flow instability by giving a number of simple but yet important examples that will shed lights on the roots of the problem. Some of the most important related work are reviewed in section III. Section IV presents the details of the proposed cross layer solution that aims to alleviate the intra-flow instability issue in multihop ad hoc networks. The simulation results and analysis are given in section V. Finally, section VI summarizes the paper and outlines the future direction in this area.

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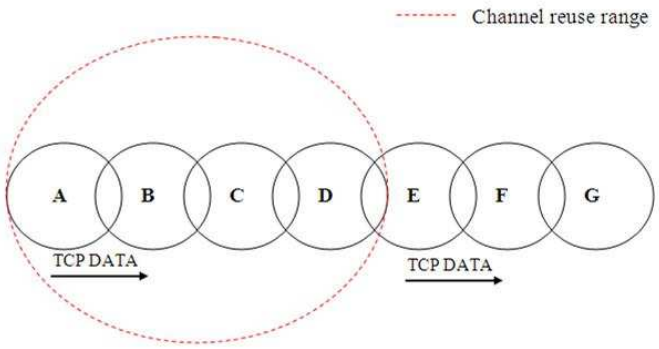


Fig. 1: TCP packet self interference.

II. Intra-flow Instability

A. Description

TCP intra-flow instability refers to the situation where the successive transmissions of packets in a single TCP flow, interfere with each other (link layer intra-flow interference) and result in large number of contention related packet drops and hence TCP instability in the network. Therefore, we begin our discussion of TCP intra-flow instability by reviewing different types of intra-flow interference and their impact on TCP instability.

B. Intra-flow Interference

As mentioned earlier, the intra-flow interference refers to situations where transmission interference in a single TCP flow causes packet drop in the network. In particular, when TCP runs over 802.11, the intra-flow interference can be broken down into the following categories:

1) Interference of TCP packets with each other
2) Interference between TCP packets and 802.11 control packets
3) Interference of 802.11 control packets with each other

Here, TCP packets refer to either TCP DATA or TCP ACK packets and 802.11 control packets include MACK (802.11 acknowledgements) and Request To Send/Clear To Send (RTS/CTS) if used.

To investigate and explain each category, in all subsequent analysis it is assumed one TCP flow is running on a 6 hop chain from node A (as data source) to node G (as data sink) and the transmission range of nodes is shown by a circle around them.

1) **TCP self interference**

   In principle, the TCP self interference is caused by two effects. One is the interference caused between TCP DATA (TCP ACK) packets transmission with each other which prevents concurrent transmissions within a neighborhood area. For instance, as shown in figure 1, a transmission from node A interferes with node C, which cannot simultaneously communicate with node D. Similarly, a transmission by node D may cause a collision at node B.

   This type of interference can harm TCP mainly in two ways. Firstly, it greatly decreases the TCP throughput in ad hoc networks since in most occasions, very few simultaneous packet transmissions can occur in the network. For instance, in the above example, links A-B and E-F represent maximum possible concurrent channel usage while if link D-E is active, only one simultaneous transmission is possible. The other impact of such interference is on increasing the end-to-end delay. This is also because a successful transmission can occur only if nodes within the spatial channel reuse of that node are silent during the entire transmission. This means packets have to wait for a relatively long period of time in the node's buffer before the node can get a chance to access the channel. Therefore, the packets in multihop connections experience longer queuing delay and hence larger end-to-end delay.

   The second part of the TCP self interference is caused by interference between TCP DATA and TCP ACK packets along the forward and return paths, respectively. In essence, this interference can specially result in TCP ACK drop as there are larger number of TCP DATA frames on the forward route compared to the smaller number of the TCP ACK packets in the return path. So, the medium will be on average mostly accessed by TCP DATA frames and as a result significant amount of ACKs will be lost because of collisions while accessing the channel.

2) **TCP and 802.11 control packets interference**

   The other type of intra-flow interference in the link layer happens between the TCP packets (TCP DATA or TCP ACK) and one of the 802.11 control packets (RTS, CTS, or MACK). However, it is important to note that regarding 802.11 MAC timing specification, the DCF protocol ensures that CTS frame transmission will be successfully received at its destination (the one who sent the RTS), if the CTS frame has been issued in response to the RTS. This is because successful RTS frame transmission silences all the nodes in the neighborhood of the source either for a duration specified in the duration field of the RTS or for EIFS time (if collision occurs) which is large enough to transmit a CTS. Therefore, the CTS frame cannot collide with any frame at the source. Using a similar argument, it can be concluded that a successful TCP frame transmission ensures a successful MACK frame transmission. Thus, there cannot be CTS and MACK frames drop at the intended destination (the node who is waiting to receive the packet) because of medium contention.

   Figure 2 reviews one the most common scenarios of TCP packet drop due to 802.11 control and TCP packets collision.

   Here, station D has TCP DATA to send to E and it sends its data after a RTS/CTS handshake with node E. Meanwhile B has a TCP DATA to send to C, thus starts its own RTS handshake. However, due to ongoing TCP DATA transmission between D and E, the B's RTS is dropped at C. Although B resends the RTS after performing an exponential backoff, in most of the cases all its RTS retransmissions (7 by default) are collided at node C and therefore node B drops the TCP packet[1].

3) **802.11 control packets self interference**

   The last type of intra-flow interference happens between 802.11 RTS, CTS control packets if the RTS/CTS handshake is used prior to data transmission. We should note that despite the small size of RTS and CTS packets

[1]This is due to the relatively large amount of time the channel is occupied when sending TCP DATA.

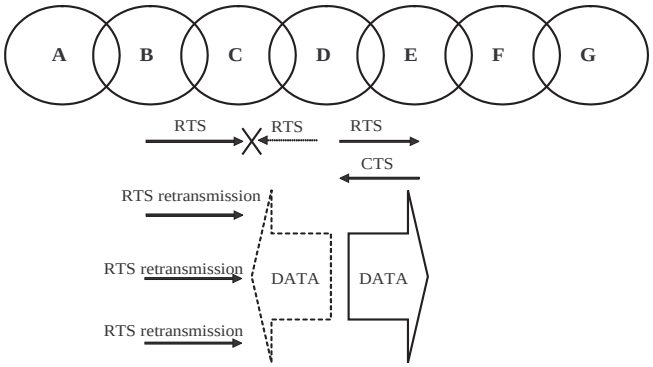


Fig. 2: RTS & TCP DATA collision.

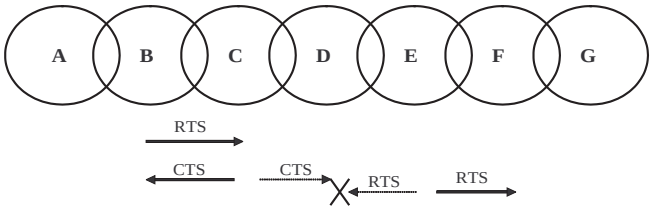


Fig. 3: 802.11 control packets collisions.

compared to data packets, the frequent losses of control packets can waste channel resources and have undesirable impact on the performance of higher layers. On the other hand, the self interference between the RTS and CTS control packets can fail the channel reservation scheme and lead to a loss of data packets as well. Figure 3 depicts a typical scenario of control packets self interference and loss of TCP packet as a result. Here B starts an RTS-CTS handshake with C before transmitting a TCP packet. The CTS reply from C is received by B correctly, but it is not received by D, which is hidden from B, due to a collision with an RTS packet sent from E to F. This happens because E, being far away from both B and C, does not hear either the RTS or the CTS packet and is unaware of the communication between B and C. Node B assumes that the channel is successfully reserved and proceeds with transmission of the data packet to C. Therefore, the TCP transmission from B is vulnerable to interference from D, which has not been able to set its Network Allocation Vector (NAV) accordingly, and may initiate a transmission to any of its neighbors before the data transmission is over.

C. Intra-flow Interference and Intra-flow Instability

Having reviewed the three types of intra-flow interferences, let us explain in more detail how such link layer interferences and packet drops can create TCP intra-flow instability. However, before that and to show the impact of intra-flow interference on TCP stability, let us review figure 4 that shows the change of congestion window (cwnd) and the instances of TCP retransmission in a static 6 hop chain topology (similar to figure 1) using 802.11 MAC.

Here, to confine the packet losses to contention drop, it is assumed the channel is error-free, no routing messages are exchanged between the nodes and all nodes have infinite buffers. Therefore, the packet losses and retransmissions are restricted to intra-flow interference related drops. It is clear from the result in figure 4 that intra-flow interference can trigger a large number of TCP retransmissions/TCP congestion window fluctuation and therefore TCP instability.

To explain further how intra-flow interference can cause TCP instability, we should note that according to 802.11 MAC standard, if a node cannot reach its adjacent node within the limited number of allowed retries (MAC-Retry-Limit), it will drop the packet. These packet drops are wrongly perceived as congestion by the TCP and result into false trigger of TCP congestion control algorithm, frequent TCP retransmissions and therefore TCP instability.

Having shown the impact of intra-flow interference on TCP instability, the next question is how it is possible to minimize the intra-flow interference in multihop ad hoc networks? The answer to this question is not straightforward as each type of intra-flow interferences discussed above are different in nature and therefore needs to be addressed separately. For instance, TCP packets or 802.11 control packets self interference can be best eliminated by designing a smart decentralized link layer schedule that coordinates the concurrent transmission between different pairs to maximize the channel utilization and minimize the number of collisions. On the other hand, TCP with 802.11 control packets interference is best to be addressed by reconsidering the link layer timing specifications, packet transmission coordination and prioritization. However, such schemes can be quite topology dependent and confined to specific scenarios. This is obviously hard to achieve in dynamic multihop ad hoc network environments where the topology of the network is changing rapidly and it is not feasible to propagate global topology information to individual nodes. More importantly, due to scarce channel resources, it is simply unrealistic to broadcast information regarding the topology and the current activity of nodes across the network.

The next alternative solution is to alleviate all types of intra-flow interferences discussed above by controlling the amount of outstanding data in the network. It should be noted that, though this approach does not fully solve the individual

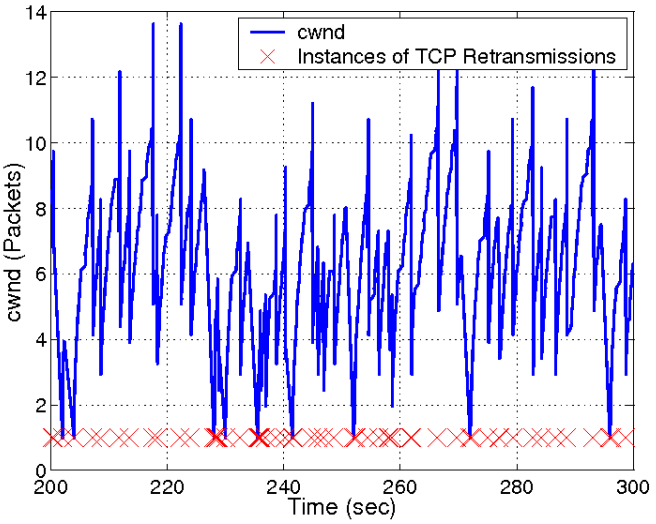


Fig. 4: Illustration of TCP congestion window change in a 6 hop chain topology.

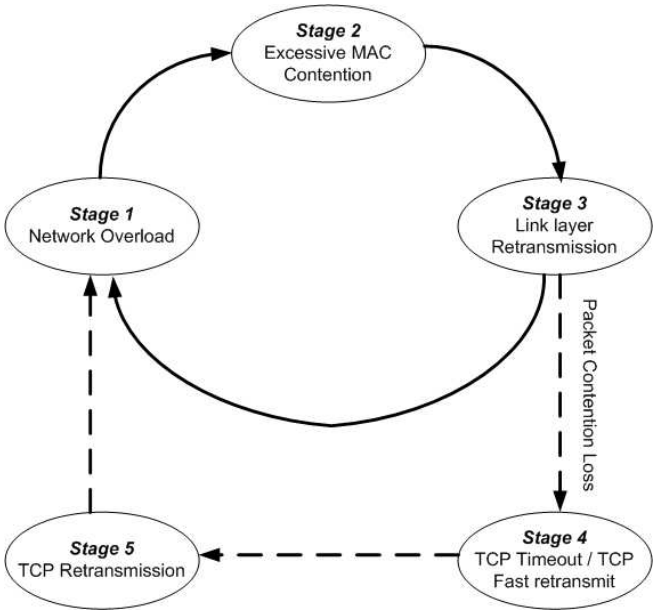


Fig. 5: Network overload and intra-flow instability cycle.

intra-interference scenarios explained before, it addresses the problem by minimizing the unnecessary intra-flow interference and its adverse effects on TCP instability. To better understand this, we should note that the performance of TCP directly depends on its flight size (swnd) which its optimal value should be proportional to bandwidth-delay product (BDP) of the entire path of the data flow [6], [19]. The excess of this threshold does not bring any additional performance enhancement, but only leads to increased queue size in intermediate nodes along the connection. On the other hand, as shown in [4], [6], [14], [20], the BDP of a TCP connection over multihop 802.11 networks tends to be very small. This is mainly because in 802.11, the number of packets in flight is limited by the per-hop acknowledgements at the MAC layer. Such property is clearly quite different from wire-line networks, where multiple packets can be pushed into a pipe back-to-back without waiting for the first packet to reach the other end of the link [21]. Therefore, as compared with that of wired networks, ad hoc networks running on top of 802.11 MAC, have much smaller BDP. However, as shown in [4], [14], TCP grows its congestion window far beyond its optimal value and overestimates the available BDP.

Figure 5 explains the chain of events that occur following a network overload and lead to TCP intra-flow instability.

The intra-flow instability cycle initially starts when increasing the network overload (stage 1) causes more contention among nodes as all of them try to access the channel (stage 2). On the other hand, when the level of contention goes up, more packets need to be retransmitted as the probability of collision increases with the increasing level of contention (stage 3). This in turn introduces extra network overload and therefore closing the inner part of the cycle (stage 1 $\rightarrow$ stage 2 $\rightarrow$ sage 3 $\rightarrow$ stage 1). This cycle is continued until one or more nodes cannot reach its adjacent node within a limited number of tries (specified by the MAC Retry Limit in 802.11 MAC standard) and drop the packet (packet contention loss). This packet loss is then recovered by the TCP sender either through TCP fast retransmit or through TCP timeout (stage 4). In both cases, TCP drops its congestion window resulting in a sharp drop in number of newly injected packets to the network (stage 5) and therefore giving the network the opportunity to recover. However, soon after TCP restarts, it creates network overload again by overestimating the available BDP of the path, and the cycle repeats.

III. Related Work

During recent years, many studies have shown that limiting the amount of outstanding data in the network can greatly improve TCP stability. In an early paper by Gerla et.al. [3], the authors showed by simulations that TCP performance degrades for congestion window greater than 1 packet when the MAC layer offers no ACK protection; They further showed that, with the link-layer ACK (MACK) protection, certain performance gain can be realized by allowing a slightly larger congestion window (2-3 packets).

To limit the amount of outstanding data in the network, [22] proposed to adjust the maximum window size parameter to 4 packets as this is the smallest value of window size for facilitating the fast retransmission scheme for TCP connections running over IEEE 802.11 based ad-hoc networks.

The authors in [8] showed that due to the spatial reuse and transmission interference property of the IEEE 802.11 MAC layer protocol in a chain topology, a sensible choice is to set TCP congestion window to $\frac{1}{4}h$, where h is the length of the chain. They further showed that TCP tends to overshoot this optimal value and operates in larger window size as we explained earlier. In the same paper, they proposed Link-layer Random Early Dropping (LRED), which aims to control the TCP window size by tuning the link-layer dropping probability according to the perceived channel contentions. In essence, similar to the RED algorithm [23] with a linearly increasing drop curve as the queue size exceeds a minimum value, LRED increases its packet dropping probability when the link-layer contention level, measured by the retransmission counts, exceeds a minimum threshold. To this aim, in LRED the link layer maintains a moving average of the number of packet retransmissions and the head-of-line packet is dropped/marked with a probability based on this average retransmission count. In particular, at each node, if the average retransmission count is smaller than a minimum threshold, the head-of-line packets are transmitted as usual. When the average retransmission count becomes larger, the dropping/marking probability is set as the minimum of the computed dropping probability and an upper bound. It was shown there that LRED can provide an early sign of network overload to the transport layer protocol and therefore force the TCP sender to reduce its transmission rate.

Finally, to decrease the amount of channel contention in the network, the authors in [24] present a cross layer approach named adaptive TCP that adaptively adjust the TCP maximum window size according to the number of RTS retransmissions at the MAC layer at the TCP sender to control the number of data packets in the network and thus decrease the channel contention.

IV. TCP Contention Control

A. Description

As discussed earlier, a high percentage of intra-flow interference and therefore contention drops can be eliminated by decreasing the amount of traffic load in the network. However, this can be a very challenging task. On one hand, if the amount of data in the network is reduced beyond the bandwidth delay

product (BDP) of that flow, the channel resources are under-utilized. On the other hand, any increase above the BDP does not bring additional performance enhancement, but only leads to increased queue size in intermediate nodes along the connection and other consequences as discussed earlier. Therefore, the main question in limiting the traffic load is how to set properly the amount of outstanding data in the network to achieve maximum throughput while minimizing the queueing delay experienced by individual packets.

In this section, the above issue is addressed by introducing a cross layer solution called TCP ConTention Control (TCTC). In simple words, TCTC adjusts the TCP transmission rate to minimize the level of unnecessary contention in the intermediate nodes. To this aim, during fixed probe intervals, TCP receiver monitors both the achieved throughput and the level of contention experienced by packets during that interval. Then, based on these observations, the receiver estimates the optimum amount of traffic to get the maximum throughput and the minimum contention delay for each connection. Finally, the TCTC propagate the information back to the sender to adjust its transmission rate.

Using this information, the TCP sender now sets its transmission rate not merely based on the level of congestion in the network and the available buffer size at the receiver but also on the level of medium contention experienced by intermediate nodes. More precisely, while TCP congestion control adjusts the TCP transmission rate to avoid creating congestion in the intermediate network buffers, TCP contention control adjusts the TCP transmission rate to avoid creating queue build up in the intermediate network buffers. Therefore, the main advantage of TCTC over previous algorithms (e.g. [3], [8], [22]) is its ability to adjust the TCP transmission rate dynamically based on the current level of network load.

Since the key element in TCTC is optimum TCP flight size, in the following we explains how TCTC estimates the optimum TCP flight size.

B. Optimum Load Estimation

To estimate the optimum amount of traffic that should be sent by the sender to get the maximum throughput while keeping the contention delay minimum, TCTC defines a new variable called *TCP Contention Window (ctwnd)*. The value of the *ctwnd* is determined according to the TCTC stages as defined below:

- **Fast Probe**
  When a TCP connection is established, the TCTC enters the *Fast Probe* state where the *ctwnd* is increased exponentially. This is very similar to the TCP slow start phase in TCP congestion control where the TCP sender probes the available bandwidth in a short time. The Fast Probe is also entered after the network is recovered back from Severe Contention stage explained shortly.
- **Slow Probe**
  Slow probe is entered when the receiver realizes that both the achieved throughput and the packet contention delay have decreased compared to the last probe interval. In this situation, the receiver concludes the network is being under-utilized and tries to gradually increase the amount of newly injected data into the network by adding one MSS to *ctwnd* in every probe interval (additive increase)
- **Light Contention**
  If after changing the amount of injected data to the network, both the throughput and the level of packet contention delay are increased, the TCTC enters *Light Contention* stage. In Light Contention stage, the TCTC slowly decreases the $ctwnd$ by one MSS per probe interval to control the amount of outstanding data in the network while avoiding unnecessary reduction in TCP throughput by implementing additive decrease. In other words, the Light contention stage is entered when the network is in early stages of overload.
- **Severe Contention**
  Severe Contention stage is entered whenever the receiver sees an increase in the level of contention delay while the achieved throughput has been decreased. This situation is a clear sign of network overload since it shows the push of more data into the network has just increased the amount of contention experienced by individual packets without increasing the throughput seen by the receiver. This situation can also happen if suddenly the level of contention in the network increases (e.g. a second connection starts using the intermediate nodes). To combat this, the TCTC sets its *ctwnd* to 2*MSS to force the sender to minimize its transmission rate.

The pseudo code in Algorithm 1, summarizes the calculation of *ctwnd* in different stages.

It is important to note that because of TCP Delayed ACK algorithm [25], the minimum *ctwnd* in TCTC is set to 2*MSS to make sure at least 2 segments are in the network and can trigger the transmission of TCP ACK at the receiver without waiting for maximum ACK delay timer to expire.

As it can be seen in Algorithm 1, the condition on which stages are entered is according to the value of two parameters named $Delta_{Throughput}$ ($\Delta_{Throughput}$) and $Delta_{Contention}$, ($\Delta_{Contention}$).
$Delta_{Throughput}$, which is calculated as in formula 1, simply compares the amount of data received by the receiver (in Bytes) in the current probe interval (probe-new) and the last probe interval (probe-old)

$$\Delta_{Throughput} = \frac{(\text{data received})_{\text{probe-new}} * (\text{probe-old})}{(\text{data received})_{\text{probe-old}} * (\text{probe-new})} \quad (1)$$

$Delta_{Contention}$ on the other hand compares the average amount of contention delay experienced by all packets during the current probe interval with the average contention delay experienced by packets during the last probe interval. In the next subsection, we explain how $Delta_{Contention}$ acquires the required information on contention delay values.

C. Contention Delay Measurement

The main objective of measuring contention delay is to reflect the current level of contention in the network. To this aim, the contention delay is measured from the time a node places the first fragment of that packet at the beginning of a buffer until the packet leaves the buffer for actual transmission on the physical layer. Also, the contention delay timer inside each node only resets to zero when the node receives a MACK. In this manner the contention delay includes the period for the successful RTS/CTS exchange, if this exchange is used for the packet. In addition, the contention delay for a retransmitted packet will start from time the original packet was placed in the head of the buffer for the first time until the corresponding MACK is received. If after reaching maximum retry limit, the

Algorithm 1 PSEUDO CODE OF CALCULATING CTWND IN DIFFERENT STAGES

if $\Delta_{Throughput} \geq 1$ **then**
  if $\Delta_{Contention} > 1$ **then**
    $TCP\_Contention = TCP\_Contention - \frac{MSS*MSS}{TCP\_Contention}$ // Light Contention
  else
    $TCP\_Contention = TCP\_Contention + MSS$ // Fast Probe
  end if
else
  if $\Delta_{Contention} > 1$ **then**
    $TCP\_Contention = TCP\_Contention + 2 * MSS$ // Severe Contention
  else
    $TCP\_Contention = TCP\_Contention + \frac{MSS*MSS}{TCP\_Contention}$ // Slow Probe
  end if
end if
if $TCP\_Contention \leq 2 * MSS$ **then**
  $TCP\_Contention = 2 * MSS$
end if

packet cannot be transmitted, the value of contention delay is added to the contention delay of the next packet.

The value of measured contention delay is then inserted inside the Contention Delay Field (CDF) using the optional field in 802.11 MAC. More precisely, each packet records the contention delay it experienced in each node and add the new contention delay to the CDF. In this manner, the Cumulative Contention Delay (CCD) experienced by each packet along the path are delivered to the final receiver (TCP receiver). The TCP receiver then calculates the value of Contention Delay per Hop (CDH) by dividing the CCD by total number of hops traversed by that specific packet. The main property of CDH is its independency from number of hops traversed by the packet. Finally the receiver derives the Mean Contention Delay per Hop (MCDH) by calculating the mean value of CDH received during each probe interval.

Having the value of MCDH, the $Delta_{Contention}$ as shown in equation 2 is derived by comparing the MCDH received by the receiver in current probe interval (probe-new) and the last probe interval (probe-old)

$$\Delta_{Contention} = \frac{MCDH_{probe-new}}{MCDH_{probe-old}} \quad (2)$$

D. Contention Delay Field

To carry the value of CDF inside a packet, we use the extended IEEE 802.11 MAC protocol packet format with optional fields inside the MAC header as suggested in [26]. In essence, a new field called "options" is proposed as a variable length field which extends standard MAC header. To perform separation of the data encapsulated into the frame from the MAC header, the option contains a Header Length field which specifies the entire length of the MAC header, including the list of options. In addition, each option consists of option type, length and data as shown in figure 6. The Header Length field is required to handle the case when a node does not support the corresponding option and therefore the knowledge of the option's length makes skipping the current option easier, jumping to the next one for processing.

Since in the proposed model, the only option being used is the CDF field, the 802.11 data packet supporting contention delay field is similar to figure 7 with the Frame Control value of 101000.

Note that here we have introduced a new type of data packet using the available reserved types in the Frame Control field of the MAC header of data frames. In particular, our frame "type" is equal to "10" (Data packet) and the "subtype" is set to "1000" as an indication of CDF-enabled data frame. This is to provide backward support within the existing IEEE 802.11 standard specification, since a CDF-enabled data frame should be of a different type with respect to a normal data frame.

E. Propagating the Contention Delay Information

Having calculated the optimum value of network overload over the next period of the probe interval, the next question is how to propagate this information (which is stored in *ctwnd*) back to the sender so the TCP sender can adjusts its transmission rate accordingly. To answer that, we should note that the TCP sender cannot have a number of outstanding segments larger than the *rwnd* which is advertised by its own receiver. By default, the TCP receiver advertises its available receiving buffer size, in order to avoid saturation by a fast connection (flow control). We propose to extend the use of *rwnd* to accommodate the value of *ctwnd* in order to allow the receiver to limit the transmission rate of the TCP sender also when the path used by the connection exhibits a high

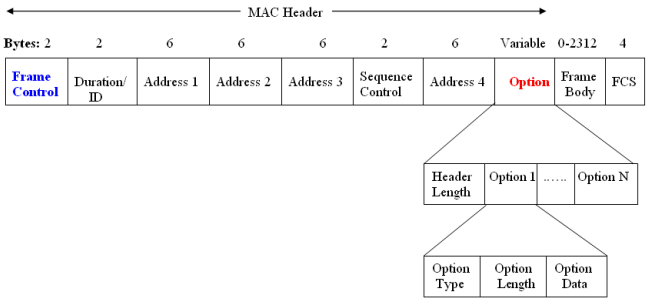


Fig. 6: Options-enabled IEEE 802.11 data frame.

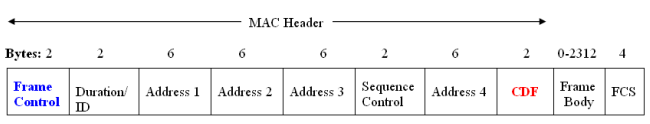


Fig. 7: CDF-enabled IEEE 802.11 data frame.

contention and frame collision probability as well as its default flow control mechanism. In other words, when TCTC is used, the new value of *rwnd* becomes the minimum of the available buffer size of the receiver (available receiver buffer) and the current value of *ctwnd* as shown below in equation 3.

$$rwnd = Min(\text{available receiver buffer}, ctwnd) \quad (3)$$

F. Choice of Probe Interval

It is clear from the above discussions that the choice of probe interval at the receiver can affect the performance of TCTC. Too large probe interval means that TCTC responds too slowly to contention changes in the network while too small probe interval will make TCTC sensitive to contention delay experienced by individual packets and therefore leads to *ctwnd* fluctuation. Heuristically, the probe interval of one RTT seems to be the natural and reasonable choice as it gives the receiver enough time to monitor the packets it received during one RTT and then sets its recommendation of sender's transmission rate (using *ctwnd*) for the next RTT. However, to provide the receiver with the correct and up to date information on the impact of previous *ctwnd* on the level of network contention, the *ctwnd* should be updated in every other RTT. Therefore, we recommend the *ctwnd* gets updated every 2*RTT and remains fixed between two updates. In this way, the receiver can make sure the packets it has received are sent into the network after the sender has applied the changes imposed by the receiver in the last probe interval. Having the value of the TCTC probe interval, the other major issue is that TCP receiver (which is running the TCTC algorithm) is unaware of connection's RTT[2]. This problem can be solved by using the fact that according to [27], in a small to medium size ad hoc networks the congestion window size (i.e. number of packets sent per RTT) does not grow beyond 5 packets. Therefore, considering this estimation together with the discussion given earlier, the probe interval in receiver can be defined as the period in which 2*5=10 packets are received. Note that although this is a rough estimation, it clearly solves our design objectives of probe interval which is merely the frequency of updating TCTC.

V. Results

A. Simulation Model

All simulations are performed using OPNET simulator [28]. The transmission range is set to 100m according to the 802.11b testbed measurements presented in [13]. In physical layer Direct Sequence Spread Spectrum (DSSS) technology with 2Mbps data rate is adopted and the channel uses free-space with no external noise. Each node has a 20 packet MAC layer buffer pool and in all scenarios, the application operates in asymptotic condition (i.e., it always has packets ready for transmission). The scheduling of packet transmission is FIFO. Nodes use DSR as the routing protocol. In transport layer, TCP NewReno flavor is deployed and the TCP advertised window is set its maximum value of 64KB so the receiver buffer size does not affect the TCP congestion window size. TCP MSS size is assumed to be fixed at 1460B. RTS/CTS message exchange is used for packets larger than 256B (therefore no RTS/CTS is done for TCP-ACK packets). The number of retransmission at MAC layer is set to 4 for packets greater than 256B (Long_Retry_Limit) and 7 for other packets (Short_Retry_Limit) as has been specified in IEEE 802.11 MAC standard. All scenarios unless otherwise stated, consist of nodes with no mobility.

[2]Assuming the receiver is not sending any data packets towards TCP sender or there is a route asymmetry between the forward and return path.

B. Simulation Results

To evaluate the efficiency of the proposed algorithm, let us first review figure 8 which shows the underlying cause of TCP packet drops in the link layer under varying number of hops in a chain topology. The importance of the results in this figure is to find out the roots of TCP retransmissions triggered from the link layer.

As it can be seen, a majority of higher layer packets are dropped because of excessive contention between competing nodes in the default algorithm. This clearly confirm the role of intra-flow interference on TCP performance. Meanwhile, a considerable number of packets are dropped due to large number of outstanding data in the network and congestion. These two issues are clearly addressed using the proposed algorithm where the TCTC almost drops the TCP packet drops to zero by limiting TCP flight size and therefore controlling both congestion and contention in the network.

To evaluate the TCP stability, let us consider the results of the TCP segment delay shown in figure 9.

As it is obvious from the above results, TCP segments in the proposed scheme experience a lower and less fluctuating delay in contrary to the default algorithm. More specifically, while using the proposed scheme, the TCP segments delay fluctuation is on average 3% in each scenarios, the figure rises up to 44% when the default algorithm is deployed. Furthermore, while the TCP segment delay is bounded between 8 to 40 ms depending on number of hops traversed by the packet in the proposed algorithm, the TCP packets segment delay can be as high as 800ms in the default algorithm which implies a high sensitivity of packet delay to hop counts. It is also interesting to note that the TCP instability in the default algorithm becomes more serious, as the number of hops increases.

While the above results shows the improvement of the TCP stability when the proposed schemes are deployed, it is very important to make sure that such stability has not been achieved in the cost of compromising TCP goodput. To this aim, figure 10 presents the aggregated TCP goodput achieved across different number of hops.

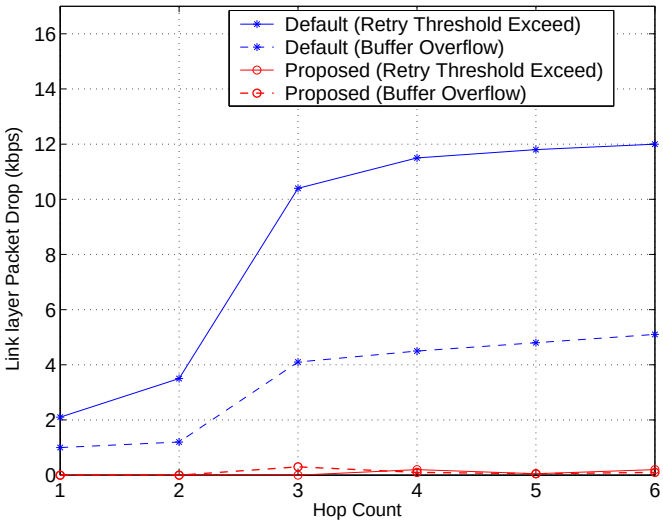
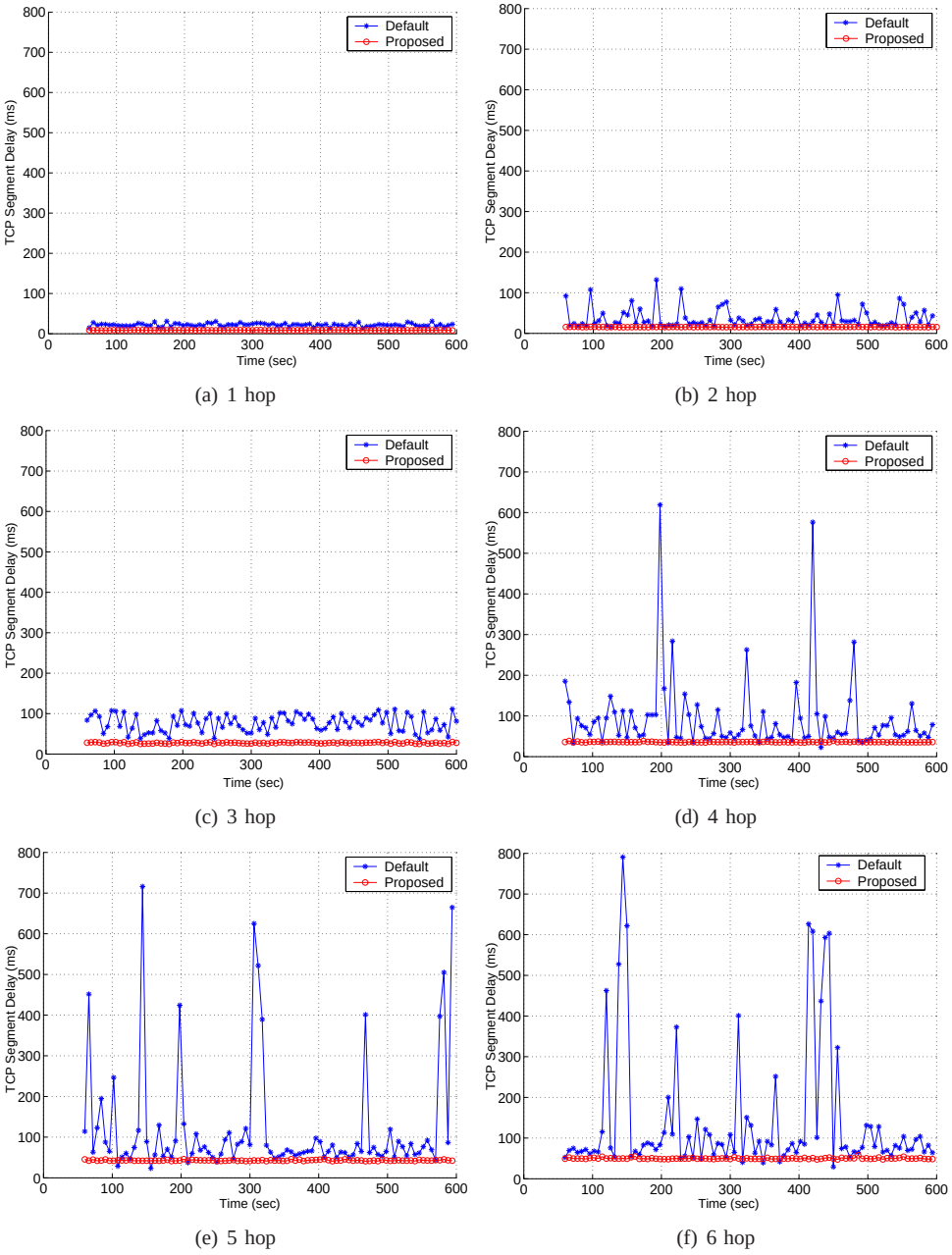


Fig. 8: The nature of higher layer drops in a chain topology.



(a) 1 hop (b) 2 hop

(c) 3 hop (d) 4 hop

(e) 5 hop (f) 6 hop

Fig. 9: TCP segment delay in a chain topology.

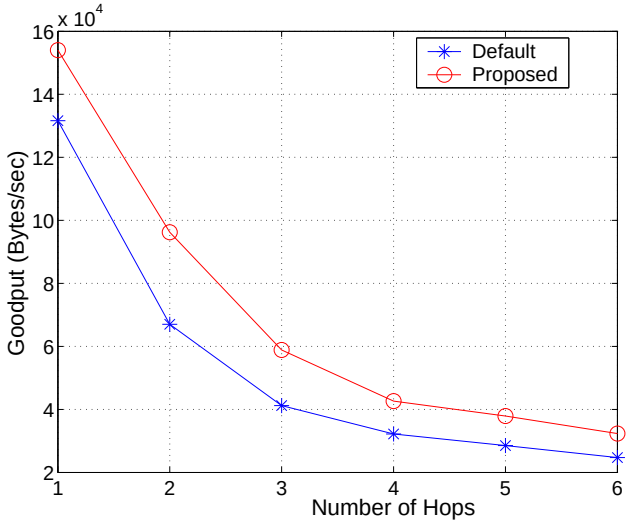


Fig. 10: TCP goodput in a chain topology.

The results are very promising as it shows there is on average 34% goodput improvement in all scenarios when the proposed schemes are applied in comparison to using default protocols.

Finally, to evaluate the efficiency of the proposed algorithm in controlling the level of congestion and unnecessary contention in the network, let us consider the number of TCP retransmissions for both schemes as shown in figure 11.

The results show that in comparison to the default algorithm, the proposed scheme has dramatically reduced the number of TCP retransmissions by controlling the amount of outstanding data in the network. It is important to note that the reduction in TCP retransmissions without any reduction in TCP goodput clearly demonstrate the efficiency of the proposed algorithm in utilizing channel resources and tuning TCP sender transmission rate close to its optimum value.

In addition to transport layer metrics, a number of measurements have been also performed in the link layer to evaluate

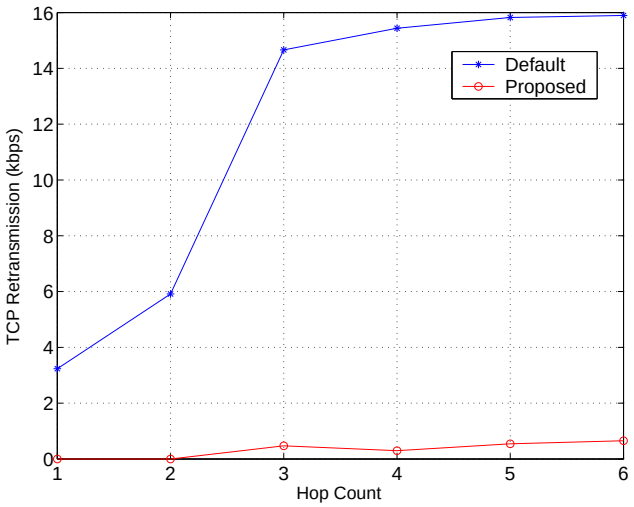


Fig. 11: Average number of TCP retransmissions in a chain topology.

the performance of the proposed algorithms in particular from the power efficiency point of view.

As the first link layer metric, figure 12 shows the average time in milliseconds that packets are kept in queues before transmitted into the network using the default and proposed algorithm.

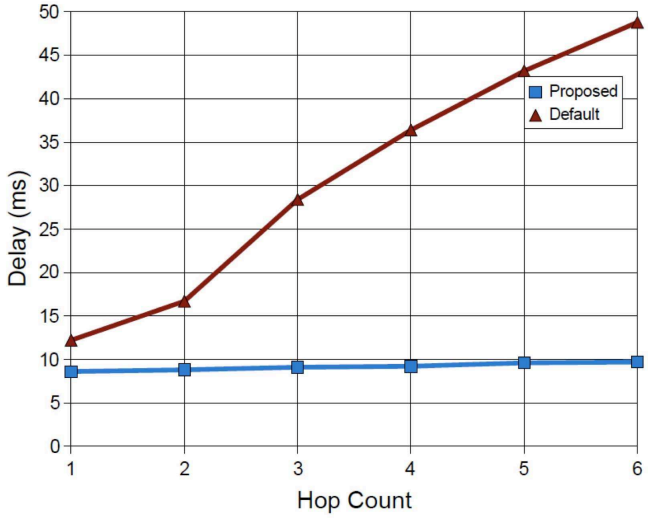


Fig. 12: Average queueing delay in a chain topology.

The results suggest the proposed algorithm keeps the queueing delay and hence the number of packets in the buffer very low and literally fixed during the simulation time and across all number of hops. It is worth mentioning again that such decrease in the queueing delay has been achieved without any compromise in the TCP goodput. In other words, the new algorithm has merely reduced the unnecessary buffering of packets in the network.

In addition, figure 13 compares the average number of link layer attempts before a packet is successfully transmitted to its next hop. This metric can be used as a indication of relative power consumption efficiency of different algorithms.

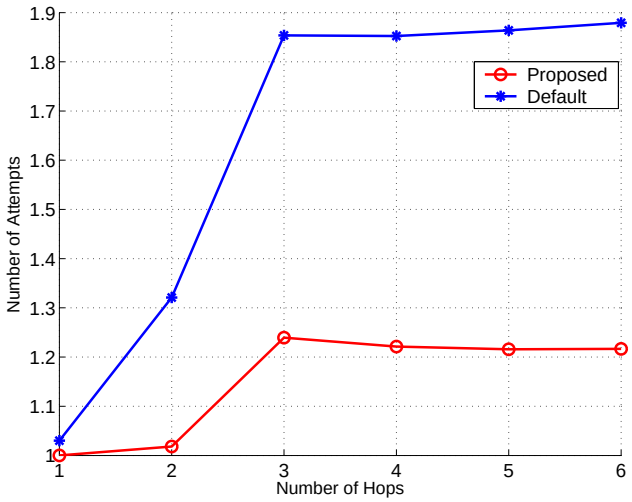


Fig. 13: Average link layer attempts in a chain topology.

From the results presented there, it can be claimed that on average 32% less transmission is required in the proposed scheme compared to the default algorithm to deliver the same amount of link layer data traffic. In other words, the proposed algorithm can reduce the number of unnecessary packet transmission by a factor of one third and therefore can save considerable amount of energy.

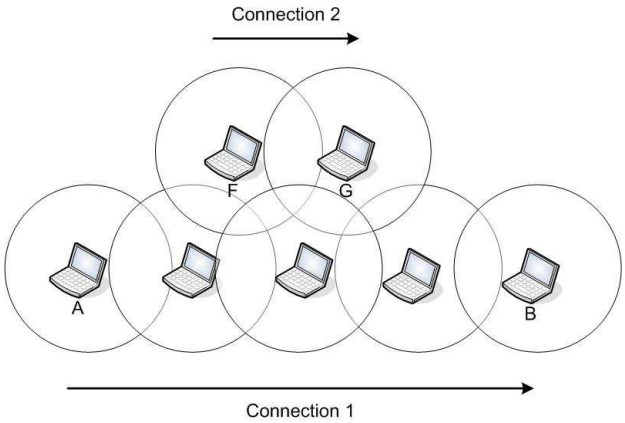


Fig. 14: 4 hop chain topology with interference.

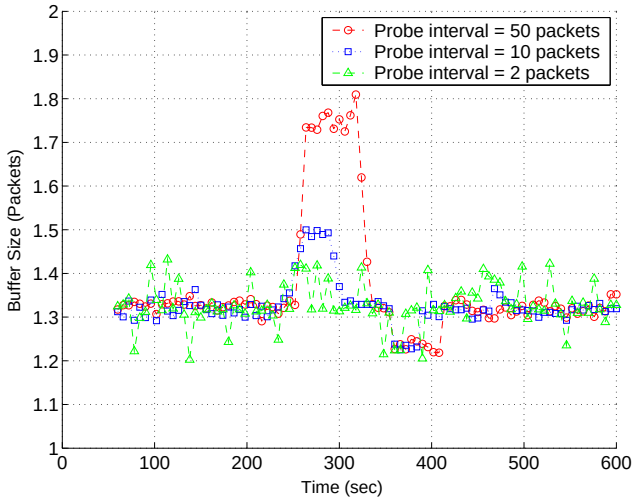


Fig. 15: Impact of probe interval choice on buffer size.

C. Impact of Probe Interval

As discussed in subsection IV-F, the probe interval is the number of samples in which the receiver updates its contention delay window ($ctwnd$). As mentioned there, the main importance of the probe interval is to determine the sensitivity of the receiver to the contention delay information received from individual packets. Note that the choice of the probe interval becomes important when for any reason (e.g. start of new connection in the interference range of one or more of the nodes along that connection), the level of contention experienced by packets changes during the connection period. To simulate such scenario, we used a topology shown in figure 14 where connection 1 (from node A o B) starts at time 0 and runs until the end of simulation while connection 2 (from node F to G) starts at time 250 seconds and lasts for 100 seconds.

To show the impact of different probe interval on system performance, the average number of buffered packets across all nodes has been used as shown in figure 15.

The results confirm first of all a short probe interval (e.g. 2 packets) can lead to fluctuation and therefore instability in the network as the calculation of *ctwnd* becomes very sensitive to the contention delay experienced by individual packets. On the other hand, if a large probe interval such as 50 packets is chosen, the convergence time (i.e. the time it takes for the algorithm to adjusts itself to new situation) can be considerable (as large as 60 seconds). This is definitely unacceptable since if during the probe interval time, the contention level increases (e.g. time 250 seconds in figure 15 when connection 2 starts), connection 1 will experience a severe delay and packet retransmissions as node B does not update its ctwnd before the end of current probe interval. On the other hand, if during the probe interval time, the contention level decreases (e.g. time 350 seconds in figure 15 when connection 2 stops), the channel resources would be underutilized as node B restricts node A from accessing the newly available bandwidth. Although the exact value of the optimum probe interval depends on individual situations, the simulation results suggest values between 10 to 15 packets satisfy the objectives of introducing the probe interval.

VI. Conclusion and Future Work

Improving the performance of TCP over 802.11 multi-hop ad hoc networks is truly a cross-layer problem. To fully benefit from the flexibility and advantages of multihop ad hoc networks, there is a clear need for improvement of the way TCP operates in such networks. This paper studied thoroughly the problem of TCP intra-flow instability in multihop ad hoc networks. In particular, it was shown in this paper that a large number of outstanding TCP packets in the network is one of the primary cause of TCP intra-flow instability. To tackle the problem, we proposed a cross layer algorithm called TCP Contention Control that it adjust the amount of outstanding data in the network based on the level of contention experienced by packets as well as the throughput achieved by connections. The main features of this algorithm is its flexibility and adaptation to the network conditions. Furthermore, TCP Contention Control is compatible with all TCP versions and it does not require any changes in TCP congestion control algorithm since it simply uses the existing TCP to throttle the amount of outstanding data in the network. This can be very useful in heterogeneous networks (wire + wireless) where the same TCP can be used in both networks. Therefore, comprehensive simulations should be conducted to evaluate the capability of the proposed schemes when the connection is expanding from wire to wireless networks or vice versa.

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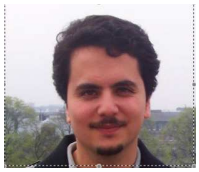

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