THE IMPACT OF MAC CONTENTION AND ROUTE FAILURES ON BURST LOSSES IN SATURATED AD HOC NETWORKS

A Thesis by

Shahid Zaman

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The following faculty members have examined the final copy of this thesis for form and content, and recommends that it be accepted in partial fulfillment of the requirement for the degree of Master of Science with a major in Computer Networking.

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Ravi Pendse, Committee Chair

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Neeraj Jaggi, Committee Member

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Krishna Krishnan, Committee Member

DEDICATION

To my parents for their prayers and wishes which gave me strength and support.

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ABSTRACT

Medium access contention and route failures in the saturated wireless ad hoc networks increase the average number of transmission attempts required to successfully deliver a frame. MAC contention and route failures also cause burst packet losses at the network layer, which results in the degradation of real-time application quality. In this thesis, the effect of MAC contention and route failures is analyzed on stationary and mobile saturated ad hoc networks for variation in the network size and data rates. In stationary networks, MAC contention has a dominating effect on burst losses. In mobile networks, route failure causes a similar effect. An increase in the data transmission rate affected the burst losses in both stationary and mobile networks.

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LIST OF ABBREVIATIONS

MAC	Medium Access Control
DCF	Distributed Coordination Function
PCF	Point Coordination Function
RTS	Request To Send
CTS	Clear To Send
DIFS	DCF Inter-Frame Spacing
SIFS	Short Inter-Frame Spacing
PIFS	PCF Inter-Frame Spacing
AIFS	Arbitration Inter-Frame Spacing
EIFS	Extended Inter-Frame Spacing
NAV	Network Allocation Vector
RREQ	Route Request
RREP	Route Reply
RERR	Route Error
ARP	Address Resolution Protocol
CBR	Constant Bit Rate
DSSS	Direct Sequence Spread Spectrum
RET	Maximum Retransmission Count

CHAPTER 1

INTRODUCTION

Real-time applications such as VoIP, IPTV, and video conferencing have increased in demand. The variations in packet delay, jitter, and packet losses in packet switched networks degrade voice and video quality. Burst packet losses have severe impact on such transmissions as the voice and video codec cannot interpolate the loss information from the received packets when a high number of consecutive losses occur in the network. The severity of these problems further increases in saturated wireless ad hoc networks because of medium access control (MAC) contention and the dynamic nature of the network resulting in route failures. Hence, a thorough analysis of such networks and the impact of MAC contention and route failures on burst losses is required to design efficient mechanisms for regulating such losses.

This thesis presents a quantitative analysis of the impact of medium access control (MAC) contention and route failure on burst losses in saturated ad hoc networks. There has been no research done in the past to study burst losses in wireless ad hoc network with respect to MAC contention and route failure. This thesis discusses how burst losses occur and the effect of MAC contention and route failure on burst losses. The dependence of burst losses on maximum retransmission count (RET) at MAC layer is the basis for this research. When a saturated wireless node exceeds the RET and still cannot successfully deliver the frame, the node discards the current packet and also the subsequent packets required to pass through the same next hop node, which results in burst losses. As the MAC contention and route failures increase, the burst losses and the average number of attempts required to successfully transmit a packet also increase. The network size and data transmission rate further affect the burst losses. Simulations were performed to study the effect of MAC contention and route failure on burst losses in stationary and mobile networks. The network size and data rate were varied to analyze their impact on burst losses. Simulation results show that MAC contention has a dominating effect on burst losses in saturated stationary networks, whereas mobility has a dominating effect on burst losses in mobile networks. An increase in network size and data rate has a similar effect on burst losses in both categories of networks.

The remainder of this thesis is organized as follows: Chapter II presents the related work, Chapter III discusses the analytical model, and Chapter IV discusses the simulations, their results and analysis. Chapter V concludes the thesis.

CHAPTER 2

LITERATURE SURVEY

And ad hoc wireless network is a self-organizing network of wireless nodes connected to each other using wireless links. Nodes in an ad hoc wireless function as a host as well as a router. In host mode, nodes operate as a source and destination, while in router mode, nodes route the packets and behave as intermediate nodes [1]. Unlike the infrastructure mode, ad hoc wireless networks do not have central authority, and nodes talk to each other in peer-to-peer fashion [2]. In ad hoc wireless networks, the shared and broadcast nature of medium, mobility, variable capacity links and node energy (as it mostly relies on node battery) are major issues which require consideration while designing MAC and routing layer protocol. Because of these issues, traditional routing protocols used for wired networks are not suitable for wireless ad hoc networks. Multiple unicast and multicast routing protocols are designed for ad hoc wireless networks which mainly include two categories: proactive (table driven) and reactive (source-initiated or demand-driven) protocols. Proactive (table driven) routing protocols try to maintain up-to-date routing information, while reactive protocols initiate route discovery on-demand and do not try to maintain the routing table all the time [4]. Both categories of protocols have their own merits and demerits; proactive protocols involve less latency because the route is already available, but they use more energy and bandwidth in order to maintain the routing table even when the route is not needed. On the other hand, reactive protocols use less bandwidth and energy and sit idle during the period of inactivity. Research is ongoing for performance analysis of different routing protocols and investigations are made under different conditions [4] [5] [6]. Dealing with packet loss and delay in ad hoc wireless networks is an important consideration. MAC contention and route failure are the primary causes for packet drops in saturated wireless ad hoc network [8]. In a saturated ad hoc wireless network the shared and broadcast nature of medium collisions are common. With an increase in contention, the nodes have to wait more in order to access the medium, and therefore, average time for successful transmission increases. When a node experiences multiple collisions for a single frame transmission, then, after a defined number of attempts (retry count), the MAC layer gives up and drops the frame and passes this information to the routing layer. Due to the presence of mobility, a node may move at any time which causes wireless link failure. Any link failure which is part of an already established path will result in path loss [7]. Mobility may cause frame drops at the MAC layer and the network layer. When a packet is delivered to the network layer from the upper layer, then the network layer forwards the packet if it has the route information. Otherwise, the network layer has to buffer the packet and the packet drop may occur if the buffer is not free or the packet sits in the buffer for the maximum allowed time because a node's implementation can't keep the packet in the buffer for an infinite time, otherwise it will cause a buffer full situation if the route is lost and the node has packets to send. On the MAC layer, mobility results in frame drop when a neighboring node, which can be a destination or intermediate hop, has moved out of range and the source node attempts to send traffic, resulting in no response from the neighbor. Consecutive packet drops are termed as burst losses; this thesis mainly focuses on burst losses due to contention and route failure. In a saturated ad hoc network, collisions may occur because of contention. In the IEEE 802.11 specifications, the distributed coordination function (DCF) is a fundamental access method of medium access control (MAC), and it shall be implemented in all mobile stations (STAs). It is also intended for use in both the independent basic service set (IBSS) and the infrastructure networks. DCF is based on carrier sense multiple access with collision avoidance (CSMA/CA), which describes both the physical and virtual carrier sensing for medium access. Physical carrier sensing mechanism is provided by the physical layer while MAC is responsible for virtual carrier sensing which involves an exchange of RTS-CTS frames. The use of RTS-CTS exchange is optional and depends upon an attribute called RTS threshold, because a high overhead will be introduced if the RTS-CTS exchange is used for small size packets. IEEE 802.11 also specifies the MAC level acknowledgement which ensures the correct reception of the frame [9].

Inter-frame spacing (IFS) refers to a time interval between two successive frames transmissions by any station. IEEE 802.11 specification defines five IFSs to provide priority levels for accessing the medium:

- Short inter-frame space (SIFS) is shortest of all intervals and has the highest priority. SIFS are used for control frames like ACK, CTS.
- PCF inter-frame spacing (PIFS) is used by STA operating in point coordinated function (PCF) mode.
- DCF inter-frame spacing (DIFS) is used by stations operating in DCF mode to transmit data frames.
- Arbitration inter-frame spacing (AIFS) is used by QoS STA to transmit data frames.
- Extended inter-frame spacing (EIFS) is the longest interval and has the least priority. It is used for resynchronization whenever a physical layer detects incorrect MAC frame reception.

In DCF mode, if a node has some data to transmit, then it senses the medium and accesses the medium for transmission. If it finds the medium idle for the DIFS duration of time, then in that case of a very low volume of traffic, the medium access delay for data frame transmission will be equal to DIFS. In case of a busy medium, a node starts random backoff and defers channel access for a random time within the contention window (between CW_{min} and CW_{max}). The size of contention window affects the collision probability. If the CW is very small, then the random values will be close to each other, thus having high probability of collision and with large size of CW there are chances of unnecessary delay. The CW value should reflect medium contention and therefore the technique used is the binary exponential backoff such that, with an increase in collision count, the window size increases. The initial value of CW is set to a random value between (0, CW_{min}) and CW doubles each time there is a collision and goes to $CW_{max.}$ When backoff counter reaches zero, the node can access the medium, but during backoff if the node finds a busy medium, then it freezes the backoff counter and resumes when the medium is sensed idle for DIFS duration. For a busy medium, when it becomes idle after a busy period, then there is a high probability of collision because multiple nodes may be waiting for transmutation. Therefore, this situation is tackled by using random backoff. If the RTS-CTS exchange is used then first sending node sends the RTS frame, and, upon reception of CTS, from the destination node it sends the data frame. The receiving node sends ACK back to ensure the delivery of the frame. Figure 1 and 2 shows the RTS/CTS/Data/ACK/NAV settings and DCF access method respectively.

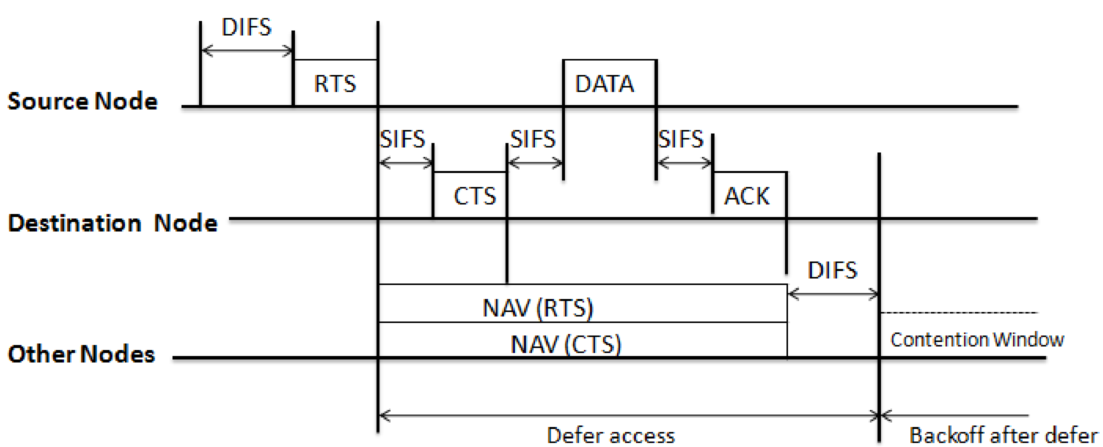


Figure 1: RTS/CTS/Data/ACK/NAV Settings

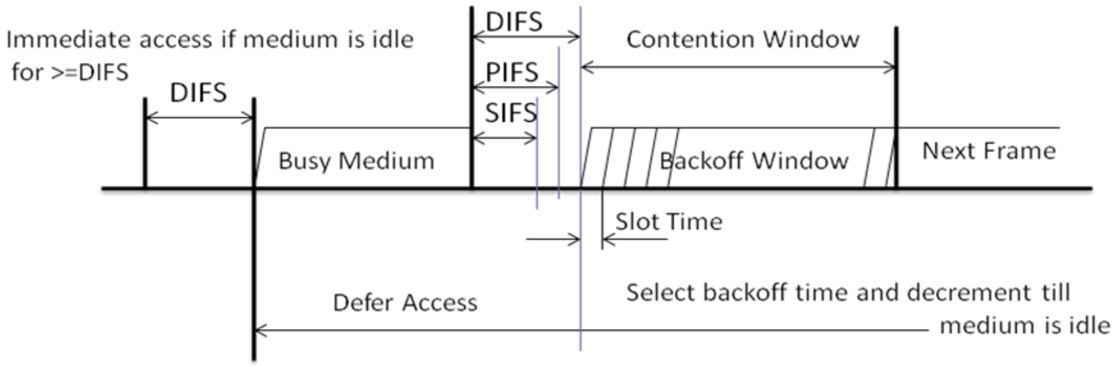


Figure 2: DCF Basic Access Method

RTS and CTS frames also have information about the duration of time for which medium is going to be busy (data transmission time plus the returning ACK time), so that the STAs within the range of the sending device as well as the receiving device will have information about the duration for which medium is going to be busy. Another way of passing this information to other STAs is through use of the Duration/ID field in individual frames [9].

In a saturated ad hoc network, the nodes have to contend more for accessing the medium resulting in a higher delay. If node 'A' wants to send data to node 'B', and after following the above mentioned procedure, it sends the RTS frame, and in case it does not receive CTS back, then node 'A' will start recovery by retrying the transmission of the frame until the maximum retry count is reached. For a node to start carrier sensing, it should first have route information. Then it can start transmission to the next-hop, which will be the destination itself if only two nodes are involved, or destination is at one hop. In case of a reactive protocol, if there is no route available, then the network layer will start route discovery. Simulations for this thesis are done using AODV as routing protocol, which is a reactive protocol. It uses a destination sequence number which ensures a loop free operation and allows having the latest route to the destination [p10]. Figure 3 shows a high-level view of on-demand routing protocol operation.

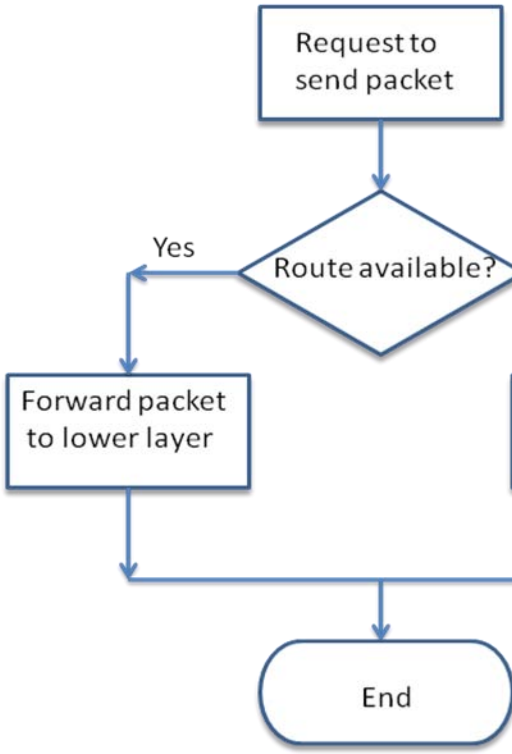


Figure 3: AODV Operation

The basic message set of AODV consists of following messages[10]

- Route Request (RREQ): If node 'A' needs a route for node 'X', then it will broadcast a RREQ message, and the diameter of RREQ can be controlled by the TTL field in the IP header. The route is determined when the RREQ reaches either the destination 'X' or an intermediate node has that has a route to 'X', provided that the route is fresh enough (the sequence number is greater than the sequence number in RREQ).
- Route Reply (RREP): The route reply unicast packet is generated by the node receiving RREQ, if the node is the destination node, or intermediate, having a route fresh enough (a sequence number greater than the sequence number mentioned in RREQ).

- Route Error (RERR): When a link break is detected for an active route, the RERR message is generated by node. One case is when a node receives RERR, which causes one or more of its routes to become unavailable, and then the node will send RERR.
- Hello Message: It is used for link monitoring; the node may send connectivity information using this type of message. The node should use this if it is part of an active route.

If a node detects that the link to the neighbor node is lost and it has a route through that neighbor node, then it will drop all the packets that were in the queue and waiting to be handed off to that neighbor. This situation may cause burst losses if there are multiple packets waiting in queue. This link lost can be detected when node reaches the maximum limit of retry for RTS and no CTS is received back. In that case, the node will mark its route, going through that next hop as invalid, and will generate an AODV RERR message to notify others. This may cause a similar situation on nodes receiving that RERR message. These burst losses may cause performance degradation for applications, especially for real-time applications which transfer media using a constant bit rate (CBR) instead of a user datagram protocol (UDP). The Voice over IP is one example of a real-time traffic; the voice stream is analog which is converted to digital before transmission over the IP network. The codec is an encoding/decoding algorithm that can be used to digitize voice data, codec compresses and decompresses voice streams, and each codec introduces processing delay [12]. The most used codec are G711, G729 and G723. Among these three codecs G711 requires the most bandwidth for transmission over the IP network and G723 requires the least bandwidth. The mean opinion score (MOS) gives an idea about codec quality and scales that value to a scale from 1-5. Table I summarizes various parameters related to these codecs [13].

TABLE I

CODEC CHARACTERSTICS

Codec	PAKITIZATION DELAY (ms)	PAYLOAD (bytes)	Tx Rate (pkts/sec)	Bit Rate (Kbps)	MOS
G711	20	160	50	64	4.1
G729	20	20	50	8	3.92
G723	30	20	34	5.3	3.65

Payload = Codec Bit Rate x Packetization Delay

Packet/Sec (PPS) = Codec Bit Rate/Voice Payload

Bandwidth = Total Packet Size x PPS

In the above table the total packet size should include the headers of IP, UDP, and Ethernet. For example, in case of G711 with Ethernet as layer-2 will require (160 + 20 + 8 + 12 + 18) x 50 x 8 = 87.2kbps

Payload = 160 bytes, IP header = 20 bytes, UDP = 8 bytes, RTP = 12 bytes, Ethernet header = 18 bytes.

As seen from the table above, the codec normally has 20msec or 30msec VOIP payload, so if there is a burst loss of 30 packets, then it will cause 600msec to 900msec of voice data loss. Later in this report simulation results show that the burst loss length is more than 30 packets on several occasions, resulting in considerable data loss.

Simulations are done for stationary and mobile networks using a network simulator (NS2). The network simulator is a discrete event simulator that is used for simulating network scenarios. Research has been done on NS2 to highlight its advancement and performance

[14][15]. NS2 implementation of mobile node consists of a link layer(LL), an ARP module connected to LL, an interface queue(IFq), a medium access control layer(MAC), and a network interface(netIF), all connected to the channel. The following is a brief description about the node components:

- Link Layer (LL): The Routing agent hands down outgoing packets to the LL which hands down packets to the IFq. For incoming packets, the MAC layer passes packets to the LL which transfers them to the node entry point.
- Address resolution protocol (ARP): The ARP module receives query from the LL, if the hardware address is known, it is written into MAC header, and, if the hardware address is unknown, a broadcast request is generated for address resolution. When the hardware address is known, the packet is placed into an IFq.
- Interface queue (IFq): Priority queue that gives priority to routing protocol packets.
- MAC layer: The Implementation of 802.11 DCF

As discussed above, the MAC contention and route failures cause burst losses, which can degrade performance of real time data to a great extent. The prior research on burst losses in ad hoc networks show that a mathematical model explaining the causes of burst losses in saturated ad hoc networks is required to be built, and the effect of the MAC layer on burst losses requires elaboration. This thesis presents a mathematical analysis of the role of MAC protocol on burst losses in saturated ad hoc networks.

CHAPTER 3

ANALYTICAL MODEL

The consecutive packet drops that occur at the node's queue are known as burst losses. These losses are the result of medium contention, mobility, and buffer overflow. This chapter presents an analysis of burst losses caused by medium contention and mobility in saturated ad hoc networks.

Suppose n_{tr} is the average transmission attempts required for the successful transmission of a frame, RTS_u is the time spent by a node in collisions, and *RTSs* is the time spent by the source or forwarding node in successful transmission, then the MAC frame service-time M_{ST} in ad hoc networks is given by [16] [17].

$$M_{ST} = [(n_{tr} - 1) \text{ x } RTS_u] + RTS_s$$

$$= (n_{tr} - 1)(DIFS + RTS + \mu) + [DIFS + RTS + CTS + DATA + ACK + 4\mu + 3SIFS] + \sum_{j=0}^{n_{tr}-1}(E[BO_j] + E[CT_j]) \quad (1)$$

DIFS is the distributed inter-frame space, *RTS* is the time required for request-to-send (RTS) frame transmission, *CTS* is the time required for clear-to-send (CTS) frame transmission, *DATA* is the time required for data frame transmission, *SIFS* is the short inter-frame space, μ is the propagation delay, $E[BO_j]$ is the average backoff interval of the node for j^{th} transmission attempt, and $E[CT_j]$ is the average time spent by the node in the cross-transmissions for j^{th} transmission attempt.

When a frame is not successfully delivered within the maximum transmission threshold (RET), the current and subsequent packets destined to the same next-hop node or to another destination passing through the same next-hop node are dropped, which results in burst losses.

Average length of burst losses is ‘l’, where ‘l’ represents the average number of packets in the node’s queue. If a node does not receive any response to RTS, it retransmits the RTS frame until the threshold is reached. Major causes behind unanswered RTS include collisions and unavailability of the node. Figure 4 shows a pictorial representation of the node’s behavior of transmitting RTS. ‘T’ represents time, ‘A’ represents source node, ‘B’ represents destination or intermediate node, ‘✕’ represents collision or unavailability of node

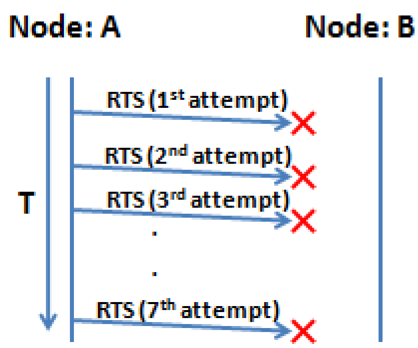


Figure 4: Node Transmitting RTS (After 7 attempts MAC layer will pass information to network layer)

Furthermore, the transmitting node forwards the route error (RERR) packet to its neighbors, which causes these neighbors also to drop all the packets required to be forwarded through the RERR originating node. Figure 5 represents flow chart of the whole process.

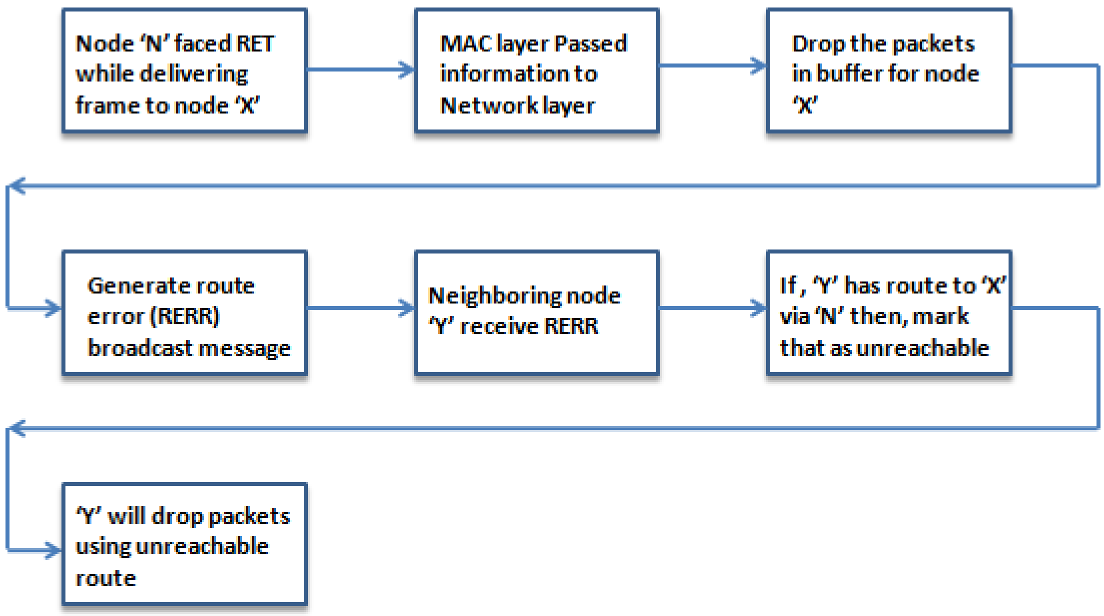


Figure 5: Burst Loss Occurrence (Node 'N' faced RET issue, MAC layer of node 'N' passed information to the network layer which will drop packets in the queue waiting to be transmitted to the next-hop node 'X' and will broadcast RERR message. Node 'Y' upon reception of RERR will mark the route passing through 'N' for destination 'X' as unreachable and will drop its packet in the queue for node 'X')

Hence, the number of packets dropped because of one RET can range from zero through $D_{\mathrm{nei}}=n_{nei}\ x\ l$, where n_{nei} is the average number of neighbors of the source node of the RERR packet which have this node as their next-hop node on the routes to their destination, and *'l'* is an average number of packets in the node's queue.

As the number of nodes in the network increase, the collision and cross transmission also increase. With an increase in collision, the frame retransmission count reaches the threshold leading to RET, which results in a packet drop at the transmitting node, and also, its neighbors re-query to forward packets to the destination. Burst losses severely degrade the quality of real-time applications, such as Voice over Internet Protocol (VoIP), and are required to be regulated. The length of burst losses can be efficiently controlled by analyzing these losses and the factors affecting them. As the MAC contention in saturated ad hoc networks increases with the network size, the number of transmitting nodes in the network directly influences the burst losses in saturated networks.

CHAPTER 4

SIMULATIONS AND RESULTS

This chapter presents the simulations performed to analyze the effect of MAC contention and route failures on burst losses in a saturated DCF network. For this analysis, the MAC contention was represented in the form of network size, whereas route failures were generated by enabling node mobility. The simulations were carried out in Network simulator-2 (NS2). The IEEE 802.11 DCF MAC protocol and AODV routing protocol were enabled in the network. Default parameters used in the simulation are presented in table II.

TABLE II

DEFAULT PARAMETERS FOR SIMULATIONS

PARAMETER	VALUE
Physical Layer Standard	FHSS
Routing Protocol	AODV
DATA (Packet Size)	500bytes
RTS	44bytes
CTS	38bytes
ACK	38bytes
Slot-time	50us
Simulation Time	110sec
Data Rate	30,40,50 (Kbps)
DIFS	128us
SIFS	28us
Channel Capacity	2 Mbps

The stationary network scenarios consisted of several static nodes acting as sources and sending Constant Bit Rate (CBR) traffic to a single destination. Three different data transmission rates of 30, 40, and 50 Kbps were employed in the simulation. All the nodes in the network were in transmission range of each other. The network sizes used in simulation were 10, 20, 30, and

40 nodes. As the source nodes increase, the channel contention and frame collision probability increases. Hence, the average time spent and average number of attempts required by a node for successfully transmitting a packet to its destination increases. After reaching the maximum retransmission count, which was set to 7, if the frame is still not delivered to the next hop node, the MAC layer drops the frame and forwards this information to the network layer. The network layer then drops all the subsequent packets destined for the same next hop node. The number for packet dropped is dependent on the node's queue size, and the packets required to be forwarded through the same next hop node at time '*t*', where *'t'* represents the time at which the MAC retransmission threshold was reached. The network traffic in scenario one was generated in such a way that nodes_1 through node_9 forwarded traffic to a common destination, node_0. Similarly, in scenario two, three, and four, node_0 was the common destination for all the remaining nodes acting as sources. The results of scenarios one through four are presented in table III through V, and Figure 4. Column I in table II through V represents the network size (NS).

Mobile network scenarios consisted of the same number of nodes as discussed for stationary networks. The same data transmission rates were used in these scenarios. The network nodes were enabled with mobility in such a way that at any given point of time only 50% of the source nodes can communicate with the destination node. Network topology was created such that initially nodes were in transmission range of each other, and later node mobility was enabled that introduced path failures leading to burst losses. Mobility can therefore cause packet drop either at MAC layer or network layer. When a packet is delivered to the network layer from the upper layer, it forwards the packet to the next hop node if the routing information is already available. Otherwise, the node has to discover a route to the destination and then forward the packet. During this period of time, the network layer has to buffer the packet, or drop it if the buffer is full. The packet loss can also occur if the packet is stored in the buffer for the maximum allowed time and then dropped since the packet cannot be stored in the buffer for an infinite time. At the MAC layer, mobility results in a frame drop when a neighboring node (destination or next hop node) moves out of the transmission range of the source node. Table III through V and Figure 5 present the results of mobile network scenarios.

TABLE III

TOTAL NUMBER OF BURST PACKET LOSSES (AVERAGE) IN STATIONARY AND MOBILE NETWORKS FOR A DATA RATE OF 30KBPS

NS	STATIONARY NETWORK	MOBILE NETWORK
10	0	201
20	333	1371
30	1309	3275
40	2032	4282

TABLE IV

TOTAL NUMBER OF BURST PACKET LOSSES (AVERAGE) IN STATIONARY AND MOBILE NETWORKS FOR A DATA RATE OF 40KBPS

NS	STATIONARY NETWORK	MOBILE NETWORK
10	0	357
20	372	1696
30	1981	3503
40	2226	4327

TABLE V

TOTAL NUMBER OF BURST PACKET LOSSES (AVERAGE) IN STATIONARY AND MOBILE NETWORKS FOR A DATA RATE OF 50KBPS

NS	STATIONARY NETWORK	MOBILE NETWORK
10	0	537
20	547	2286
30	2231	3930
40	2936	4627

This subsection analyzes the simulation results presented above. Medium contention and route failures affect the transmitting node and its neighbors as discussed below.

Burst Losses at the Transmitting Nodes: Medium contention in stationary ad hoc networks increases with an increase in the number of network nodes. With increased contention, the minimum number of packet transmissions required for successful delivery of packets exceeds (RET). In Tables II, III, and IV, the average of the total number of burst losses in static networks increased with an increase in the data rate and network size. For a small network size of 10 nodes in a stationary network, the nodes did not face much contention to cause the retry count to exceed, so burst losses are recorded as zero. As the network size increased to 20 nodes, the medium contention also increased, and certain nodes reached RET during the packet transmissions resulting in burst losses. The medium contention and the burst losses increased further with a further increase in the network size.

In case of a mobile network, the nodes lose the route to the destination node causing packet drops. The MAC layer of the transmitting node declares the next-hop node as unreachable after reaching RET, and this causes the network layer to mark the routes through this next-hop node as unavailable. The node drops the packet waiting in queue to be handed off to this lost next hop causing burst losses. It also broadcasts a route error (RERR) packet to its neighbors. The number of packets dropped at a node depends upon the number of packets in its queue to be delivered to that next-hop at any time instance *'t'*, where *'t'* represents the time at which the transmitting node experienced RET due to link failure. The burst losses presented in Table II, III, and IV for mobile networks primarily occurred because of route failures, though a very small percentage can be attributed to the MAC contention. Node movement was enabled in the network in such as way that, at any point of time, only half of the source nodes were able to reach the destination. After several packet transmission attempts at the MAC layer, the transmitting nodes declared the next-hop node as unreachable, and dropped the current and subsequent packets destined to the next-hop or the packet which require passing through it. As the network size increased, the number of nodes experiencing burst losses also increased. It may be noted that the mobile networks experienced burst losses even for smaller network sizes because of route failures, which are not observed in stationary networks affected only with MAC contentions. In the 10 node mobile scenario, approximately 5 source nodes experienced burst losses at any given point of time. With an increase in the network size, the medium contention along with route failures increased the burst losses. The following two figures show the graphical representation of burst losses in stationary and mobile ad hoc networks.

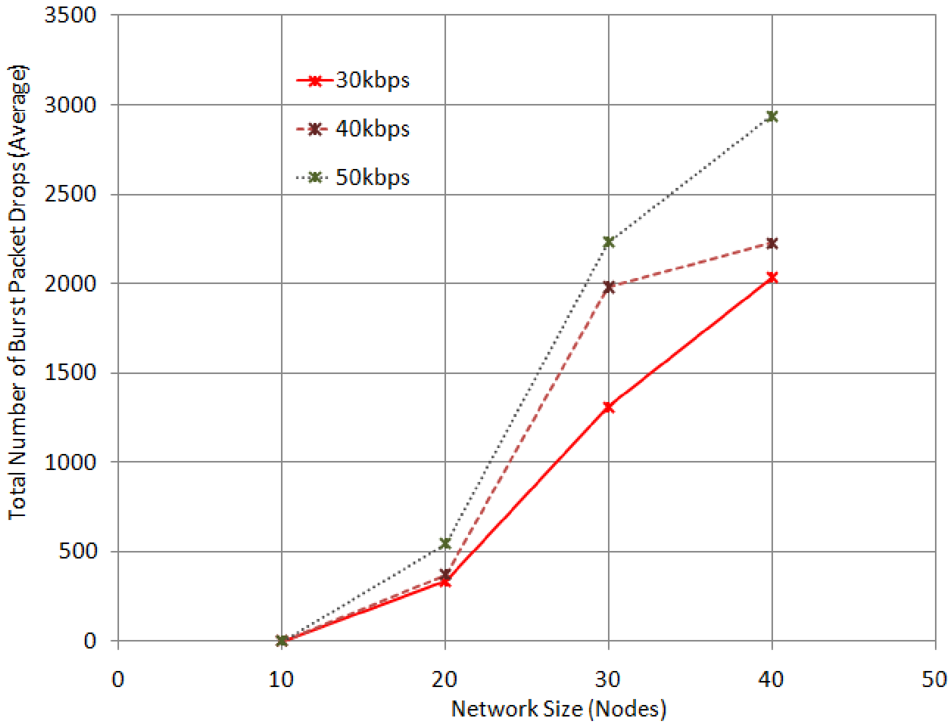
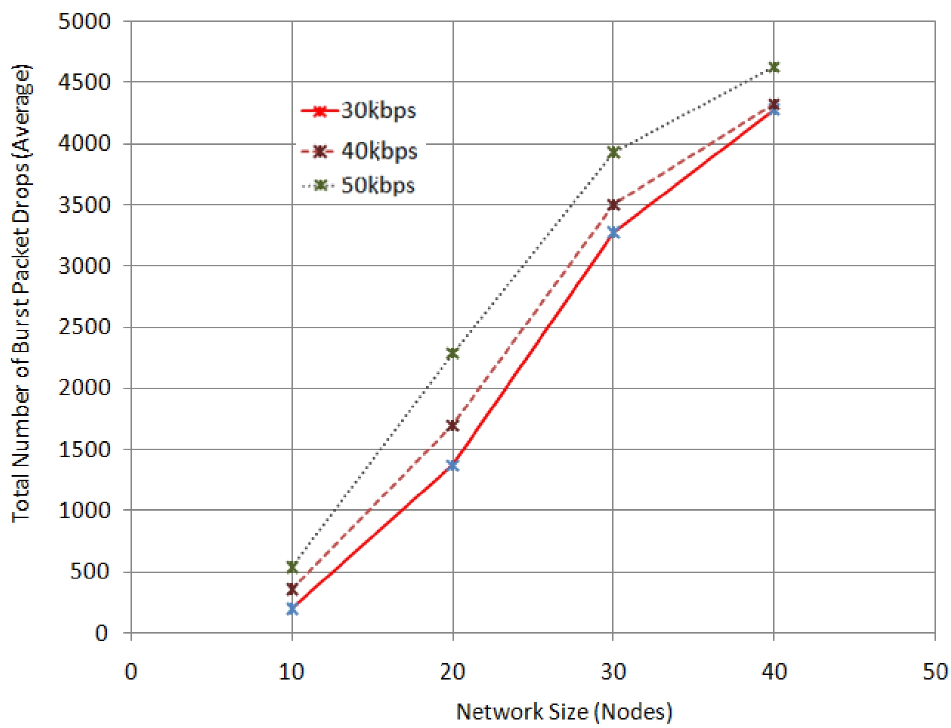


Figure4. Network Size Vs Burst Packet Drop (Average) in Stationary Network



. Figure5. Network Size Vs Burst Packet Drop (Average) in Mobile Network

Burst Losses at the Neighbors: Once the transmitting node experiences RET and drops the packets destined for the next-hop, or which pass through the same next-hop node, it forwards a route error (RERR) packet to all its neighbors indicating a route failure. The nodes which receive this message and which have the RERR forwarding node as the next-hop node on their routing table will also drop all the packets which were required to pass through this node. Hence, the number of packets dropped because of one RET can range from zero through $D_{nei}=n_{nei}\ x\ l$, where n_{nei} is the average number of neighbors of the source node of RERR packet which have this node as their next-hop node on the routes to their destination, and *'l'* is the average number of packets in the node's queue. Table VI presents a sample of packet drops (average) at the neighbors of source nodes experiencing RET which broadcasts the RERR packet. Column II and III present the total number of packet drops in stationary and mobile networks, respectively.

TABLE VI

TOTAL NUMBER OF BURST PACKET LOSSES (AVERAGE) AT THE NODES RECEIVING RERR (DATA RATE OF 50KBPS)

NS	STATIONARY NETWORK	MOBILE NETWORK
10	0	4
20	188	1911
30	747	2301
40	1374	2677

Hence, the medium contention and route failure in saturated DCF networks have severe impact on burst losses. The burst losses in turn have adverse effects on real-time traffic, such as voice transmissions in ad hoc networks. Because the real-time (UDP) traffic can neither be acknowledged nor retransmitted, the burst losses in real-time traffic can result in a performance degradation of the application.

The simulation results show that burst losses increase with an increase in MAC contention (network size) in stationary networks. Route failures primarily caused burst losses in mobile networks, both on source nodes as well as on the neighboring nodes. Simulation results for stationary networks showed that the burst losses were zero for a network size of 10 nodes, and these losses initiated for a network size of 15 nodes as shown in Table VII. As the network size increased, the burst losses also increased. Therefore, the burst losses can be effectively regulated by reducing the network size to less than 15, which can be considered the best network design with the nodes in the transmission range of each other.

TABLE VII

TOTAL NUMBER OF PACKET LOSSES (AVERAGE) IN STATIONARY NETWORK FOR A DATA RATE OF 30KBPS

NS	STATIONARY NETWORK
10-14	0
15	100
20	333

CHAPTER 5

CONCLUSION

Increase in network size (NS) causes an increase in MAC contention resulting in an increase in the average number of transmission attempts required for successful delivery of a packet in a saturated wireless ad hoc network. After reaching the MAC retransmission threshold, the source node drops the current packet and the subsequent packets waiting in buffer to be forwarded through same next hop node resulting in burst losses. Route failure caused by mobility also causes burst losses on the source node as well as the neighboring nodes. In this thesis, the effect of MAC contention and route failures on burst losses was studied for saturated and mobile ad hoc networks. Simulation results show that the MAC contention increases the burst losses in stationary networks and the route failures increase the burst losses in mobile networks; whereas an increase in data rate affects both stationary and mobile networks.

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