# THE ANALYSIS OF BURST PACKET LOSSES IN SATURATED AD HOC NETWORKS USING SEMI-MARKOV PROCESS

A Thesis by

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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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Neeraj Jaggi, Committee Member

Linda Kliment, Committee Member

# DEDICATION

To my beloved parents whose way of living their life has always given me inspiration to excel in my journey

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# ABSTRACT

Medium contention in saturated ad hoc networks increase collisions and retransmissions. As the number of transmissions for a frame exceeds the maximum retransmission count in the IEEE 802.11 networks, the medium access control (MAC) layer drops the frame and forwards this information to the network layer. The network layer further drops all the subsequent packets destined to the same next hop node or passing through this node, assuming the node to be unavailable. Furthermore, the transmitting node propagates this information through the route error (RERR) packet. The receiving nodes drop their packets destined to the same next hop node or if the packets are destined to through it to another destination. The network, therefore, experiences burst losses at several nodes, leading to the degradation of quality of real-time applications such as Voice over IP (VoIP). Hence, burst losses must be reduced and this requires a well-defined model to explain the burst losses and their causes. This thesis uses semi-Markov process based model for illustrating the burst losses in the IEEE 802.11 saturated ad hoc networks. It describes the frame transmission process in the IEEE 802.11 networks and the effect of medium contention on burst losses. Furthermore, the proposed model also computes the average frame service-time in the saturated ad hoc networks. Simulations are conducted to analyze the effect of variations in the network size (number of nodes in the network) and the data transmission rate on the burst losses. The results show that an increase in the network size and data transmission rate, independently increases burst losses. Simulation results present a range of acceptable network sizes to minimize the burst losses in the saturated IEEE 802.11 networks.

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LIST OF ABBREVIATIONS

| | |
|---|---|
| AC | Access Category |
| ACK | Acknowledgement |
| AIFS | Arbitrary Distributed Inter-frame Space |
| AMST | Average MAC Service Time |
| AODV | Ad hoc On Demand Distance Vector |
| AP | Access Point |
| ATA | Average Transmission Attempts |
| BEB | Binary Exponential Backoff |
| CA | Collision Avoidance |
| CBR | Constant Bit Rate |
| CSMA | Carrier Sense Multiple Access |
| CTS | Clear to Send |
| CW | Contention Window |
| DCF | Distributed Coordination Function |
| DIFS | Distributed Inter-Frame Space |
| DSR | Dynamic Source Routing |
| DSSS | Direct Sequence Spread Spectrum |
| DTMC | Discrete-Time Markov Chain |
| EDCF | Enhanced Distributed Coordination Function |
| IFQ | Interface Queue |
| LAN | Local Area Network |

| | |
|---|---|
| MAC | Medium Access Control |
| MNO | Maximum Number of Occurrence |
| MOS | Mean Opinion Score |
| NAV | Network Allocation Vector |
| NS | Network Simulator |
| PCF | Point Coordinated Function |
| PITA | Percentage Increase in Transmission Attempts |
| QoS | Quality of Service |
| RERR | Route Error |
| RET | Retransmission |
| RREP | Route Reply |
| RREQ | Route Request |
| RTS | Request To Send |
| SIFS | Short Inter-Frame Space |
| TOT | Telephone Organization of Thailand |
| TPD | Total Packet Drop |
| TTL | Time to Live |
| TXOP | Transmission Opportunity |
| UDP | User Datagram Protocol |
| UMTS | Universal Mobile Telecommunication System |
| VoIP | Voice over Internet Protocol |

LIST OF SYMBOLS

| | |
|---|---|
| $CW_{max}$ | Contention Window Maximum |
| $CW_{min}$ | Contention Window Minimum |
| $E[BO_j]$ | Expected Time of Backoff-interval |
| $E[CT_j]$ | Expected Time Spent in Cross-transmission |
| $E[ST_i]$ | Expected Frame Service-time for $i^{th}$ Transmission Attempts |
| $E[ST_{m+1}]$ | Expected Frame Service-time for m+1 Transmission Attempts |
| $E[T_{coll}]$ | Expected Time Spent in Collision |
| $E[T_{succ}]$ | Expected Time for Successful Transmission |
| m | Back off stage |
| $\mu$ | Propagation Delay |
| $N$ | Number of Nodes |
| $p$ | Conditional Collision Probability |
| $X_c$ | Sum of Constant Value in Collision |
| $X_s$ | Sum of Constant Value in Successful Transmission |
| $Y$ | Variable Dependent on Contention of the Medium |
| $\Gamma$ | Node's Packet Transmission Probability |

# CHAPTER 1

# INTRODUCTION

The use and popularity of real time applications has increased tremendously in recent years. Such applications include voice traffic, video, and live streaming. Implementation of these applications has greatly improved in the infrastructure network. However, maintaining the same level of quality as the infrastructure network in the ad-hoc network is still both challenging and desired. Real-time traffic is very sensitive to delay and packet-loss. Most of this traffic uses User Datagram Protocol (UDP) as a transport protocol, which provides unreliable data transfer in the network.

Real-time traffic in ad hoc network suffers additional loss due to the mobility of the node and the contention in the medium. An increase in the network size or the node's data transmission rate increases the medium contention in saturated ad hoc networks which results in collisions and retransmissions [1] – [3]. When the frame transmission count exceeds the threshold value in the IEEE 802.11 networks, the medium access control (MAC) layer drops the frame and forwards this information to the network layer [2]. The network layer assumes that the next hop node to reach the destination node is unavailable and drops all the subsequent packets that are destined to the next hop node or that need to pass through the next hop node. Furthermore, the network layer also propagates a route error (RERR) packet to its neighbors regarding the node unavailability. Thus, all the neighbors forwarding their packets through the currently unavailable node, or destined to this node, will drop their packets [4]. These events result in burst losses at several network nodes, finally degrading the quality of real-time applications such as Voice over Internet Protocol (VoIP) in saturated ad hoc networks.

It is very important to regulate these losses and this requirement necessitates a well-structured model that describes the frame transmission process in saturated ad hoc networks and the cause of burst losses. By identifying and fine-tuning the parameters responsible for medium contention and burst losses in saturated ad hoc networks, the burst losses can be regulated to support the essential real-time applications in the network.

The IEEE 802.11 wireless Local Area Networks (LANs) were previously analyzed using two-dimensional Markov chains [2] in terms of saturation throughput, node's packet transmission probability, and conditional collision probability. This model was further employed for computing the average frame-service time [5]. However, the two-dimensional Markov chain was not extensively used for describing burst losses in saturated ad hoc networks. Similarly, a simulation-based analysis of packet losses in ad hoc networks was conducted in [3]. This thesis uses semi-Markov process based model to explain the effect of contention (caused by an increase in the network size and data transmission rate) on the burst losses in saturated ad hoc networks. The semi-Markov process based model is employed for performance analysis of the IEEE 802.11 wireless LANs. Furthermore, it evaluates the minimum frame service-time in saturated ad hoc networks. The simulation performed on the Network Simulator (NS-2) show that an increase in network size increases the burst losses. Similarly, for a specific network size, an increase in data transmission rate increases the burst losses.

The remainder of this thesis is organized as follows: Chapter 2 presents the literature review completed for the research. Chapter 3 utilizes the analytical model to explain the effect of contention on the burst losses in saturated ad hoc networks. Chapter 4 presents the simulation, results, and the analysis of the results and Chapter 5 presents the conclusion.

# CHAPTER 2

# LITERATURE SURVEY

Ad hoc networks are self-manageable collections of wireless nodes that communicate with each other without any need of dedicated infrastructure or central authority. Nodes in these networks interact with each other on their own, create paths, manage routes, send data, and observe the environment for maintenance or updates. There are many protocols that are designed for different layers to achieve stability in the network. Implementation of these different protocols, (transport, routing, MAC etc.) depends on specific requirements. IEEE 802.11 Distributed Coordination Function (DCF) is the most commonly used channel access protocol at the MAC layer. Routing protocols are characterized into two main branches: reactive and proactive. As the name suggests, reactive protocols are mostly event-driven and operate on a per need basis. Proactive protocols are table-driven and maintain the route information of the network regardless of demand.

This chapter is divided into four sections. Sections 2.1, 2.2, and 2.3 discuss the IEEE 802.11 MAC protocols, ad hoc on demand routing protocols, and a few varieties of VOIP codecs used in IP networks. Section 2.4 presents the related work.

## 2.1 MAC Protocols

### 2.1.1 IEEE 802.11 Distributed Coordination Function (DCF)

IEEE 802.11 Distributed Coordination Function (DCF) is a mechanism to access the channel in the wireless medium. It uses the Carrier Sense Multiple Access/Collision Avoidance (CSMA/CA) technique to access the medium. It also uses the Binary Exponential Backoff (BEB) mechanism, which helps to avoid collision between the contending nodes [2]. BEB provides an extra layer of reliability while sending and receiving data. If a node has to send data, it first listens to the medium for a certain amount of time to see if the medium is busy, if it is found busy, the node waits until the medium is free and then reinitiates the communication. In case of a collision, all involved nodes wait for a random back off interval time before sending the next frame to the destination MAC, which decreases the probability of the next collision.

In 802.11, the DCF allows the transmission of the packet through the medium in two ways: the basic access mechanism and access with Request to Send/Clear to Send (RTS/CTS) frames. The first method helps a node to send a frame immediately after it listens to the medium for a Distributed Inter-frame Space (DIFS) time and the latter method involves an exchange of additional RTS/CTS frames before sending data to the destination MAC address. This method helps to avoid the hidden node problem, which occurs when any node that is out of range from both the source and the destination wants to send frames at the same time. RTS/CTS frames contain information for the length of the packet. This helps other neighboring nodes to update their Network Allocation Vector (NAV) for the time the channel will remain busy. Finally, the data transfer is complete with Acknowledgement (ACK) from the destination, indicating a successful transmission. Figure 1 shows the access mechanism with RTS/CTS [6].

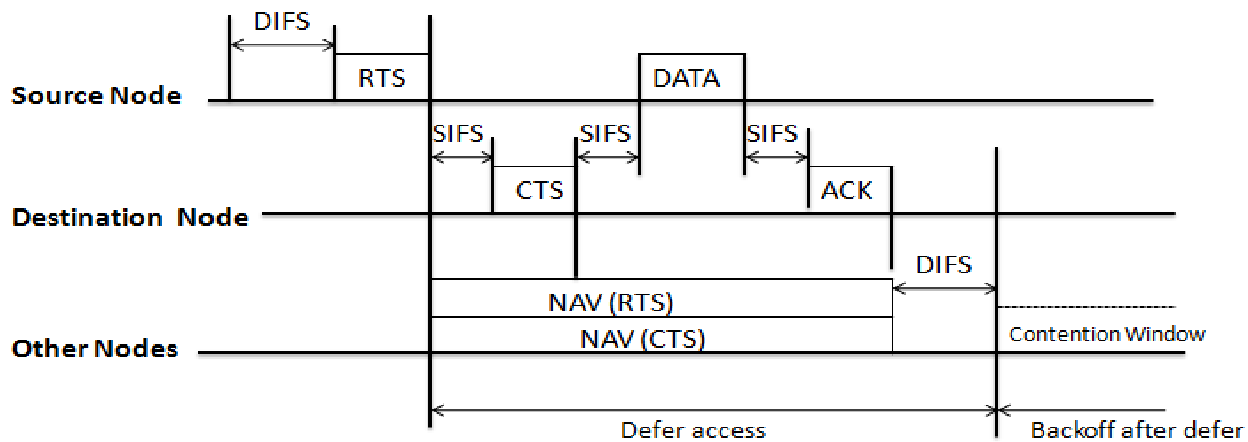


Figure 1. IEEE 802.11 DCF with RTS/CTS

With the increase in network size (the number of nodes), the probability of collisions also increases. DCF addresses this issue with a backoff counter. Nodes involved in a collision invoke a random timer generated from the range (0, CW-1), where CW is the contention window. The value of CW depends on the number of retransmission attempts. On the first attempt, CW is CWmin, i.e. minimum, and is increased twice each time a collision is observed until the value of CW reaches CWmax, i.e. maximum. After a successful transmission, the value of CW is set to CWmin. As the number of transmissions for a frame exceeds the maximum transmission count, the medium access control (MAC) layer drops the frame and forwards this information to the network layer. The frame is then discarded, and MAC services the next frame in the queue. Further, network layer further drops all the subsequent packets destined for the same next hop node or passes through this node, assuming it unavailable.

### 2.1.2 Enhanced Distributed Coordination Function (EDCF)

The legacy IEEE 802.11 DCF is not able to differentiate between different types of data traffic. It treats each stream of data with same DIFS and CW, regardless of the application and sensitivity to the delay. The Enhanced Distributed Coordinate Function (EDCF) addresses the QoS issue in wireless ad hoc networks. It works on top of the legacy DCF and defines four different channel access categories (AC), each with a different defer time, Arbitrary Distributed Inter-frame Space (AIFS), and CWmin/CWmax values [7]. Using different CW (min/max) and AIFS values for different streams of data traffic, EDCF provides priority from traffic over the other. EDCF has defined four different access categories, AC0 through AC3, 3 being the highest priority [7]. Depending on the application, each traffic stream is associated with each access category and maintained by the access point (AP). EDCF uses transmission opportunity (TXOP) for higher priority AC for transmission of several packets at one transmission instance. Table 1 shows the value of different parameters in EDCF.

TABLE 1

PARAMETERS FOR DIFFERENT ACCESS CATEGORIES IN EDCF

| Access Category | Priority | CWmin | CWmax | AIFS |
|---|---|---|---|---|
| Background | 0 | 31 | 1023 | 7 |
| Best Effort | 1 | 31 | 1023 | 3 |
| Video | 2 | 31 | 63 | 1 |
| Voice | 3 | 7 | 15 | 1 |

## 2.2 Ad Hoc On-Demand Routing Protocols

### 2.2.1 Ad Hoc on-Demand Distance Vector (AODV)

Ad Hoc on-Demand Distance Vector (AODV) is an on-demand routing protocol used in mobile ad-hoc networks. The route to a destination is established on a per need basis. AODV maintains only the next hop information to reach a particular destination, and uses broadcast RREQ packets to find a route to the destination. It maintains a dynamic routing table at each node to the destination. It can detect the status or health of the neighbor with regular HELLO packets. If a neighbor does not responds to two consecutive HELLO packets, it is considered dead and traffic via that node is re-routed to a different node for the next transmission of the packet [8]. AODV maintains the sequence number of both the source and destination to check the freshness of a given route and to look out for unnecessary propagation of RREQ or RREP packets. It also helps to avoid loops in the network. AODV is scalable to a large number of nodes and requires less overhead and memory consumption as compared to a proactive routing protocol.

When a node has to send data to a destination node, it first looks into its routing table, to see if it has a route to reach that node. If it does not, it initiates a Route Request (RREQ) packet. This is a broadcast packet that every node in the range receives. The RREQ packet contains [8],

<*source_address, source_sequence_number, broadcast_id, destination_address, destination_sequence_number, hop_count*>

On receiving a RREQ packet, the neighboring node checks in its routing table to see if it has a route to reach the destination. If it has, it sends back a Route Reply (RREP) packet to the source. If it does not, then it re-broadcasts the packet to its neighboring nodes. This process repeats until the RREQ reaches the destination and it sends the RREP back to the source. An intermediate node, on receiving the RREQ checks for *source_address, source_sequence_number* to make sure that it has not seen the same packet from another neighboring node [8]. If it has, then it suppresses that packet for duplication. Figure 2 shows the management of the duplicate RREQ in AODV. Thus, with each RREQ packet forwarded to a destination, a reverse path to the source is maintained at an intermediate node.

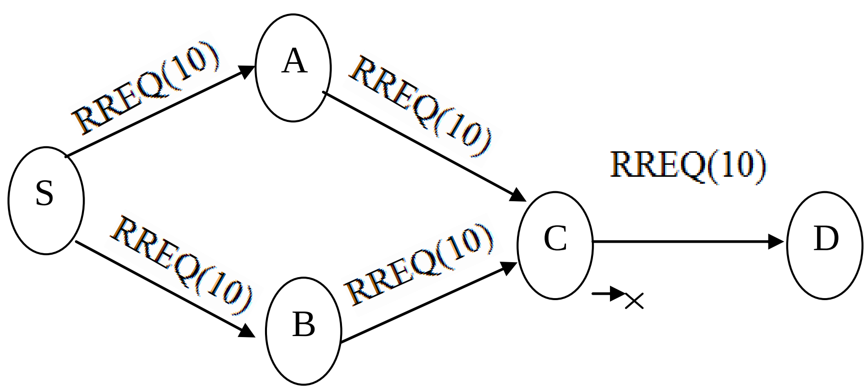


Figure 2. Management of Duplicate RREQ in AODV

When a destination receives the RREQ packet, it sends the RREP via the reverse path established during the RREQ propagation process. Each intermediate node on receiving the RREP from the destination updates its routing table by comparing its *destination_sequence_number* for the freshness of the route. RREP is an unicast packet destined for the source. If a node receives multiple RREP from different paths it compares *destination_sequence_number,* and if the number is higher than previously seen or lower hop count with the same sequence number it updates its routing entry or it drops the RREP packet from further propagating in the network.

### 2.2.2 Dynamic Source Routing Protocol (DSR)

Dynamic Source Routing Protocol (DSR) is another on-demand routing protocol used in mobile ad-hoc networks. It restricts the bandwidth consumption of the medium, as compared to other table-driven proactive routing protocol, because it does not require regular route maintenance updates. DSR uses cache information to optimize the route look up process. Route look up is done with Route Request (RREQ) packets sent to all neighboring nodes within the range of the transmission. Once the neighbor receives the RREQ, it looks into its routing table, and if the route is available it replies to the source node with the Route Reply (RREP) packet. If the route is not available it re-broadcasts the packets to its entire neighbor and the process is repeated by intermediate nodes until the packet reaches the destination. With each broadcast of the packet, from source to destination, all intermediate nodes add their information (node ID) in the packet as the path followed [9]. When that packet reaches the destination, it has a complete reverse path to the source, and it replies with RREP.

An intermediate node can also get route information even if it is not participating in route discovery. This is done when a node is in promiscuous mode, where it stores the route based on overhearing from neighboring nodes. Routes stored at a source node for a longer period are purged after the time to live (TTL) expires, and prevents the node from using stale nodes in the mobile network. DSR maintains a route cache at every node for exchanging efficient route information with its neighbors. Route information is added, modified or deleted based on the route request, route reply, and route error packets. A particular node at a given time may have multiple routes to the same destination. DSR uses the hop count as a metric to choose the best path among available options.  DSR also uses packet salvaging, which helps in the efficient recovery of the route in case of a broken path.

When a route to the destination breaks for some reason (node movement, dead node etc.), a route error packet is generated by the previous hop to the destination. From Figure 3, node B generates the REER packet after the link breaks between B and D. All intermediate nodes including the source node flush the cache for that destination and look for an alternative path. The node switches to an alternative path if available, or it generates a new RREQ for the destination.

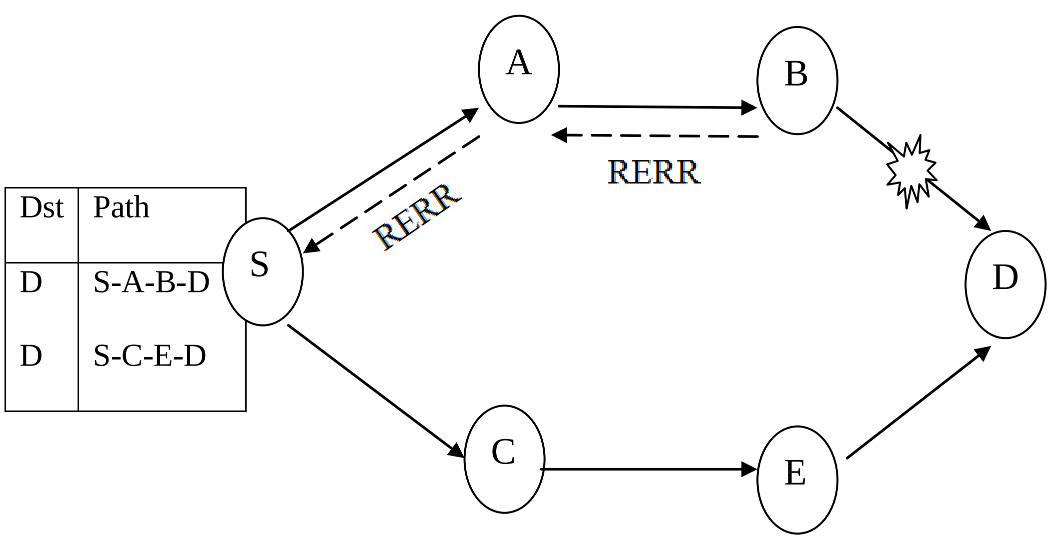


Figure 3. Route maintenance in DSR

## 2.3 VOIP Codes (Coder – Decoder)

Voice streams are analog signals which are digitized using different encoding algorithms and transmitted over the medium to the destination and decoded back to original form. There are several voice codecs available today and using any particular codec depends on the requirement of the user. The complexity of the codec along with the provided quality, throughput, delay etc. varies from one algorithm to another [10]. Some of the popular codecs are G.729, G.711, G.723, and GSM [10]. Different parameters are considered before choosing a particular codec. Table 2 briefly summarizes the different parameters of the codecs [10]. Bandwidth calculation is an important factor while designing network capacity. The voice payload is the number of bytes of data that are in each packet. Packets per second are the number of packets that required per second for the codecs bit rate. Mean Opinion Score (MOS) gives the measure of the quality of the codec and its value ranges from 1 through 5, where 1 is considered bad and 5 is considered excellent.

TABLE 2

VOICE CODEC PARAMETERS

| Codec & Bit Rate (Kbps) | Voice Payload | Packets per Second | MOS | Bandwidth Ethernet (Kbps) |
|---|---|---|---|---|
| G.729 (8Kbps) | 20 | 50 | 3.92 | 31.2 |
| G.723 (6.3Kbps) | 24 | 33.3 | 3.9 | 21.9 |
| G.711 (64Kbps) | 80 | 50 | 4.1 | 87.2 |

## 2.4 Related Work

Barcelo et al. [11] presented a joint VoIP/IEEE 802.11 model for ad hoc networks. They made several assumptions for the model to calculate the average number of collisions and average packet delay. Some of the assumptions included ideal channel conditions, packet drops only due to collisions, an uncongested channel, and unsaturated sources. Their results show that not only configuration parameters (codec, data rate, and packetization interval), but the distribution of the stations' VoIP packet generation times in a packetization affects the delay.

Bolot et al. [12] analyzed the issue of distributing audio data over the internet. They show that internet does not provide a guarantee of the resources such as bandwidth and performance assurance. To resolve this issue they have proposed two approaches. The first approach is to extend the current protocols and switch the scheduling discipline to provide performance assurance and the second approach is to control the rate at which packets are sent over the connections. They conclude the paper by saying that the number of consecutively lost audio packets is small when the load is less and they can be reconstructed using open loop error control mechanisms.

Velloso et al. [13] analyzed the voice transmission capacity on IEEE 802.11 ad hoc networks. They studied delay, jitter, loss rate and consecutive loss by varying the node density and mobility on the network. Their work shows that the mobility of the node has a significant effect on the degradation of the voice transmission. They showed that a decrease of 60% of the number of voice transmissions can be observed in the multihop environment and the density of the node should be considered for voice transmission.

Pragtong et al. [14] used the traffic data from the network of the Telephone Organization of Thailand (TOT) to present the characteristics and traffic modeling of VoIP conversation. They proposed that the new state “long burst” which shows that the background noise at the caller’s end is added to the model based on the continuous-time Markov process. Their simulation showed that background noise significantly increases the data rate mean, but reduces variance of the data rate of VoIP traffic.

Jiang and Schulzrinne [15] discuss the modeling and analysis of delay and losses of real-time applications over the internet. They show that Quality of Service (QoS) of the applications must be measured to provide feedback on rate and error control. They explain that the extended Gilbert model is a suitable loss model and an inter-loss distance metric as suitable for capturing burstiness between the loss runs. They show that the losses in the real time traffic are mostly bursty in nature and the delays have a strong temporal dependency.

Cao and Gregory [16] evaluated the performance of VoIP services for different codecs. They used a hybrid network including Universal Mobile Telecommunication System (UMTS) network segment to study the end-to-end delay on VoIP transmission. Their simulation showed the correlation between the delays, and implemented codecs and the number of voice frames in each VoIP packet. Their results showed G.729A had the best result with respect to delays by comparing it to other codecs used in simulation.

Kopsel and Wolisz [17] compared the basic DCF and Point Coordination Function (PCF) for the transmission of audio data in a real-time scenario. They propose a simple priority mechanism used on mobile and access points to improve the QoS in terms of bandwidth. They also suggest another implicit scheme for improving the channel capacity by avoiding unsuccessful PCF polling attempts.

Hole and Tobagi [18] evaluated the delay constraints, channel conditions and voice call quality requirements for the IEEE 802.11b network. They evaluated the G.711 and G.729

encoding schemes with a range of packet sizes. From the simulation they show that the capacity of 802.11b is highly sensitive to delay budgets allocated to packetization and wireless network delay. They explain the tradeoff between throughput and delay constraints and show that voice quality requirement per scenario may help in determining the codec scheme to use. Their simulation results show G.729 to allow greater capacity with a slight compromise on voice quality.

Kadiyala et al. [19] presented their study on dependency of burst losses on packet inter-arrival time in VoIP. They used the semi-Markov process to explain the mode. The simulation was done using Cisco's routers instead of an ad hoc device or with the help of legacy NS-2. They showed that the burst losses increased for lower packet inter-arrival times, thus explaining the dependency of burst losses on packet inter-arrival time.

From the above discussions it is clear that MAC protocols, such as the IEEE 802.11 DCF protocol, were previously analyzed using the two-dimensional Markov chain [2] and the semi-Markov process [1]. These models compute the saturation throughput, the node's packet transmission probability, and conditional collision probability and help in the performance analysis of the MAC protocols in ad hoc networks. Several analytical models, including those based on the two dimensional Markov chain proposed in [2], were developed to evaluate the frame service-time [3]. The burst losses in wireless ad hoc networks were analyzed based on simulation results in [5]. However, an analytical model describing the burst losses in saturated ad hoc networks, which is an important requirement, was never developed. This thesis fulfills this requirement by using semi-Markov process based model for illustrating the burst losses in saturated ad hoc networks. Furthermore, the expected time spent in each transmission attempt of a packet is thoroughly analyzed.

# CHAPTER 3

# ANALYTICAL DISCUSSIONS OF BURST LOSSES USING SEMI-MARKOV PROCESS BASED MODEL

This chapter presents the analytical discussions of burst losses in saturated ad hoc networks. This model is based on the semi-Markov process based model discussed in the literature [1].

## 3.1 Analysis of Burst Losses

The semi-Markov process based analytical model [1] describes the operation of the Binary Exponential Mechanism (BEB) of the IEEE 802.11 Distributed Coordination Function (DCF) protocol. The states of the semi-Markov process and their state holding times represent the backoff stages of the BEB mechanism and their backoff intervals, respectively [1]. Figure 4 presents the Discrete-Time Markov chain (DTMC) of the model [1]. When a node needs to transmit a packet, it enters state 0 and transmits the frame after its backoff interval. If the packet transmission was successful, the node returns to the state 0 for the next frame transmission. Otherwise, if the frame collides, where $N$ is the number of nodes, $\Gamma$ is node's packet transmission probability, and conditional collision probability $p$ [1], [2] is given as,

$$p = 1 - (1 - \Gamma)^{(N-1)}$$

the node enters state 1 and retransmits the frame. This process continues until the node successfully transmits the frame or the frame transmission attempt count exceeds the transmission threshold after consecutive collisions. If the frame transmission was unsuccessful, then the node drops the frame and informs the network layer. The network layer assumes that the next hop neighbor is unavailable, and it drops all subsequent packets which are destined to the same neighbor, or which are required to pass through this neighbor to reach the destination. This results in burst losses at the transmitting node. Furthermore, the transmitting node forwards a route error packet (RERR) packet to its neighbors conveying message about the link failure. Thus, all the neighbors which use this network link for the packet transmission process drop all their subsequent packets destined to the unavailable node or which pass through this node to reach their destination. Hence, the burst losses increase in the network. As the burst losses increase, the quality of real-time application, such as VoIP, decreases. With an increase in the network size, the medium contention and hence the conditional collision probability $p$ increases [1], [2]. Therefore, a node enters higher back off stage (from 1 through m) more often due to collisions. Figure 4 shows semi-Markov process based model. Similarly, the node's data transmission rate increases the medium contention. Therefore, the medium contention has to be regulated to minimize the burst losses in saturated ad hoc networks by selecting the acceptable range of network sizes in the saturated ad hoc networks and data transmission rates; such losses can be reduced. By fine-tuning the combination of network size and data transmission rates in the saturated ad hoc networks, such losses can be reduced to the desired value.

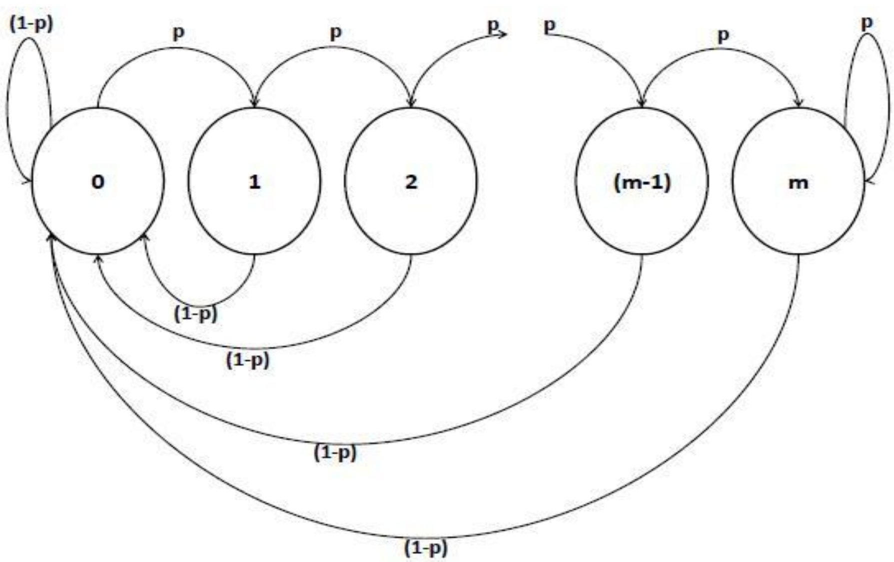


Figure 4. (m+1)-state Markov chain with state transitions probability

## 3.2 Computation of Frame Service-time in Saturated Ad Hoc Network

In this subsection, the frame service-time in saturated ad hoc networks is computed using the semi-Markov process based model discussed in the previous subsection.

Suppose that the distributed inter-frame space is represented by DIFS, the time required for RTS frame transmission is represented by RTS, the time required for the CTS frame transmission is given by CTS, DATA is the time required for the data transmission, ACK is the time required to receive the ACK from destination MAC, SIFS is the short inter-frame space, and μ is the propagation delay. $E[BO_j]$ represents the expected value of the backoff-interval for backoff stage $j$ given by

$$E[BO_j] = 2^j CW_{min}/2$$

$E[CT_j]$ represents the expected value of time spent in cross-transmissions for the backoff stage $j$.

The expected time taken by a node to successfully transmit a MAC frame in the backoff stage $j$ is [4]

$$E[T_{succ}] = DIFS + RTS + CTS + DATA + ACK + 3SIFS + 4\mu + E[BO_j] + E[CT_j] \quad (1)$$

Thus, (1) represents the sum of constant and variable delays of the frame transmission process, which can be rewritten as [4]

$$E[T_{succ}] = X_s + Y \quad (2)$$

where $X_s = DIFS + RTS + CTS + DATA + ACK + 3SIFS + 4\mu$, and,

$Y = E[BO_j] + E[CT_j]$

Similarly, the expected time spent by a node in an unsuccessful transmission (collision) is $E[T_{coll}]$ is given by [4]

$$E[T_{coll}] = X_c + Y \quad (3)$$

where $X_c = DIFS + RTS + \mu$ is constant.

If a node requires (m+1) transmission attempts for successful delivery of a frame, then the expected frame service-time $E[ST_{m+1}]$ is [4]

$$E[ST_{m+1}] = mX_c + X_s + \sum_{j=0}^{m}(E[BO_j] + E[CT_j]) \tag{4}$$

The expected time $E[ST_i]$ for the successful delivery of a frame in $i$ transmission attempts is [4]

$$E[ST_i] = (i\text{-}1X_c + X_s + \sum_{j=0}^{m}(E[BO_j] + E[CT_j]) \tag{5}$$

Because it is difficult to analytically model cross-transmission and since cross-transmissions increase with the network size, the time spent in cross-transmissions is studied with simulations, and neglected in the analytical model. The delays $E[BO_j] + E[CT_j]$ are dependent on the MAC contention in the saturated networks. With an increase in network size and data transmission rate, the MAC contention increases and hence the time spent in cross transmissions and the number of attempts required for successful delivery of the frame also increases.

Thus, the DTMC of semi-Markov process [1] is only used to explain problem of burst losses (in chapter 3) in IEEE 802.11 DCF networks. This model is not a contribution of this thesis and is not evaluated in this work. The actual contribution of this thesis is the quantitative analysis of the effect of network size and packet transmission rates on burst losses in saturated ad hoc networks that is presented in chapter 4. In addition, this problem is addressed with simulations on NS-2. In summary, chapter 3 and chapter 4 are independent, and chapter 4 presents the entire contributions of this thesis.

# CHAPTER 4

# SIMULATIONS AND RESULTS

This section presents the simulations carried out to analyze the impact of MAC contention and data transmission rate on burst losses in saturated stationary ad hoc networks.

## 4.1 Network Topologies

Network Simulator (NS-2) is used for performing the required simulation. The effect of contention is studied by varying the number of nodes in the network (defined as network size), and observing the total packet losses at the associated transmitting nodes in the network. A set of three different scenarios are generated by varying the positions of the nodes. The impact of packet transmission rates on burst losses is analyzed by varying the data transmission rates of the CBR traffic by employing different VoIP codecs, namely, G.729 (8Kbps) and G.723 (6.3Kbps and 5.3Kbps). Network sizes are varied from 10 through 50 with a step size of 10. The data transmission rates used in the simulation are 8Kbps, 6.3Kpbs and 5.3Kbps.

Table 3 presents the values used for various parameters associated with IEEE 802.11 DCF protocol in the simulations. The routing protocol used in the network is AODV.

TABLE 3

DEFAULT PARAMETERS USED IN SIMULATION

| Parameter | Value |
|---|---|
| Physical Layer Standard | DSSS |
| DIFS | 50µs |
| SIFS | 10µs |
| Slot-time | 20µs |
| RTS | 44bytes |
| CTS | 38bytes |
| DATA | 20bytes (G.729-8Kbps)<br>20bytes (G.723-6.3Kbps)<br>20bytes (G.723-5.3Kbps) |
| ACK | 38bytes |
| Data Rate | 8Kbps (G.729)<br>6.3Kbps (G.723)<br>5.3Kbps (G.723) |
| Routing Protocol | AODV |
| IFQ | 200 |
| Node Transmission Range | 250m |
| Simulation Time | 100s |

## 4.2 Simulation Result

Tables 4 through 12 show the relationship between the network size and the total packet drops in burst losses caused by MAC contention for 3 different scenarios. The columns of these tables represent the network size (NS), total packet drops due to burst losses (TPD), packet drop percentage (PDP), and the maximum number of RETs occurrences (MNO). The total packet drop (TPD) is the number of packet drops that occur when a node exceeds the maximum retransmission attempt (RET). The percentage of packet drops (PDP) is the ratio of packets dropped due to the RET errors to the total number of data packets generated by the application layer. MNO is the maximum count of RETs encountered by a node in a particular network scenario of a particular network size. For example, in an average calculation for a 20 nodes scenario, if the total number of data packets generated at the application layer is 43,200, the total drops due to RET error is 2,512, so this means that 5.81% of the total data packets are dropped due to RET errors.

### 4.2.1 Scenario 1

The CBR traffic representing the standard VoIP traffic generated by the codecs G.729 with 8Kbps and G.723 with data transmission rates of 6.3Kbps and 5.3Kbps are generated from the source nodes to the destination nodes. For a network size of 10 nodes, the nodes are configured to represent 9 sources transmitting the packets to destination node 10. Similarly, for network sizes of 20, 30, 40, and 50 nodes, there are two destination nodes receiving traffic from 18, 28, 38, and 48 source nodes, respectively. Half of the source nodes transmit their packets to one particular destination and the remaining source nodes transmit their packet to another destination node. The contention in the network increases with an increase in the network size. Table 4, 5, and 6 shows the relationship between variations in the total drops for the variation in the network size for the packet transmission rates of 8 Kbps, 6.3Kbps, and 5.3Kbps.

TABLE 4

VARIATIONS IN TOTAL DROPS FOR VARIATIONS IN NETWORK SIZE FOR PACKET TRANSMISSION RATE OF (G.729) 8Kbps

| NS | TPD | MNO | PDP |
|---|---|---|---|
| 10 | 0 | 0 | 0.00% |
| 20 | 2537 | 2 | 5.87% |
| 30 | 10652 | 4 | 15.85% |
| 40 | 10808 | 5 | 9.38% |
| 50 | 10471 | 4 | 9.08% |

TABLE 5

VARIATIONS IN TOTAL DROPS FOR VARIATIONS IN NETWORK SIZE FOR PACKET TRANSMISSION RATE OF (G.723) 6.3Kbps

| NS | TPD | MNO | PDP |
|---|---|---|---|
| 10 | 0 | 0 | 0.00% |
| 20 | 968 | 3 | 3.38% |
| 30 | 9719 | 4 | 19.83% |
| 40 | 9978 | 5 | 16.51% |
| 50 | 9875 | 7 | 12.93% |

TABLE 6

VARIATIONS IN TOTAL DROPS FOR VARIATIONS IN NETWORK SIZE FOR PACKET TRANSMISSION RATE OF (G.723) 5.3Kbps

| NS | TPD | MNO | PDP |
|---|---|---|---|
| 10 | 0 | 0 | 0.00% |
| 20 | 892 | 3 | 3.14% |
| 30 | 8052 | 6 | 18.25% |
| 40 | 7497 | 7 | 16.55% |
| 50 | 6829 | 4 | 9.03% |

### 4.2.2 Scenario 2

Tables 7, 8, and 9 show the relationship between variation in the total drops for the variation in the network size for packet transmission rate of 8 Kbps, 6.3Kbps, and 5.3Kbps for the second scenario. Scenarios are differentiated from each other by changing the node positions.

TABLE 7

VARIATIONS IN TOTAL DROPS FOR VARIATIONS IN NETWORK SIZE FOR PACKET TRANSMISSION RATE OF (G.729) 8Kbps

| NS | TPD | MNO | PDP |
|---|---|---|---|
| 10 | 0 | 0 | 0.00% |
| 20 | 2537 | 2 | 5.87% |
| 30 | 10199 | 4 | 15.77% |
| 40 | 10562 | 4 | 9.16% |
| 50 | 10132 | 5 | 8.71% |

TABLE 8

VARIATIONS IN TOTAL DROPS FOR VARIATIONS IN NETWORK SIZE FOR PACKET TRANSMISSION RATE OF (G.723) 6.3Kbps

| NS | TPD | MNO | PDP |
|---|---|---|---|
| 10 | 0 | 0 | 0.00% |
| 20 | 892 | 3 | 3.11% |
| 30 | 8032 | 4 | 16.39% |
| 40 | 8798 | 4 | 14.56% |
| 50 | 8562 | 5 | 11.22% |

TABLE 9

VARIATIONS IN TOTAL DROPS FOR VARIATIONS IN NETWORK SIZE FOR PACKET TRANSMISSION RATE OF (G.723) 5.3Kbps

| NS | TPD | MNO | PDP |
|---|---|---|---|
| 10 | 0 | 0 | 0.00% |
| 20 | 951 | 3 | 3.32% |
| 30 | 8456 | 4 | 17.25% |
| 40 | 8744 | 5 | 14.47% |
| 50 | 8546 | 6 | 11.20% |

### 4.2.3 Scenario 3

Tables 10, 11, and 12 show the relationship between variation in the total drops for the variation in the network size for packet transmission rate of 8 Kbps, 6.3Kbps, and 5.3Kbps for the third scenario. Scenarios are differentiated from each other by changing the node positions.

TABLE 10

VARIATIONS IN TOTAL DROPS FOR VARIATIONS IN NETWORK SIZE FOR PACKET TRANSMISSION RATE OF (G.729) 8Kbps

| NS | TPD | MNO | PDP |
|---|---|---|---|
| 10 | 0 | 0 | 0.00% |
| 20 | 2463 | 2 | 5.70% |
| 30 | 10267 | 4 | 15.27% |
| 40 | 10463 | 5 | 9.08% |
| 50 | 10218 | 4 | 8.79% |

TABLE 11

VARIATIONS IN TOTAL DROPS FOR VARIATIONS IN NETWORK SIZE FOR PACKET TRANSMISSION RATE OF (G.723) 6.3Kbps

| NS | TPD | MNO | PDP |
|---|---|---|---|
| 10 | 0 | 0 | 0.00% |
| 20 | 951 | 3 | 3.32% |
| 30 | 8456 | 4 | 17.25% |
| 40 | 8744 | 5 | 14.47% |
| 50 | 8546 | 6 | 11.20% |

TABLE 12

VARIATIONS IN TOTAL DROPS FOR VARIATIONS IN NETWORK SIZE FOR PACKET TRANSMISSION RATE OF (G.723) 5.3Kbps

| NS | TPD | MNO | PDP |
|---|---|---|---|
| 10 | 0 | 0 | 0.00% |
| 20 | 853 | 3 | 3.00% |
| 30 | 7320 | 4 | 16.59% |
| 40 | 7520 | 5 | 16.60% |
| 50 | 7129 | 5 | 9.42% |

Tables 13, 14, and 15 show the relationship between the network sizes and the average total packet drops in burst losses caused by MAC contention for 3 different scenarios.

TABLE 13

VARIATIONS IN TOTAL AVERAGE DROPS FOR VARIATIONS IN NETWORK SIZE FOR PACKET TRANSMISSION RATE OF (G.729) 8Kbps

| NS | TPD | MNO | PDP |
|---|---|---|---|
| 10 | 0 | 0 | 0.00% |
| 20 | 2512 | 2 | 5.81% |
| 30 | 10373 | 4 | 15.63% |
| 40 | 10611 | 4.6 | 9.21% |
| 50 | 10274 | 4.3 | 8.86% |

TABLE 14

VARIATIONS IN TOTAL AVERAGE DROPS FOR VARIATIONS IN NETWORK SIZE FOR PACKET TRANSMISSION RATE OF (G.723) 6.3Kbps

| NS | TPD | MNO | PDP |
|---|---|---|---|
| 10 | 0 | 0 | 0.00% |
| 20 | 937 | 3 | 3.27% |
| 30 | 8736 | 4 | 17.82% |
| 40 | 9173 | 4 | 15.18% |
| 50 | 8994 | 6 | 11.78% |

TABLE 15

VARIATIONS IN TOTAL AVERAGE DROPS FOR VARIATIONS IN NETWORK SIZE FOR PACKET TRANSMISSION RATE OF (G.723) 5.3Kbps

| NS | TPD | MNO | PDP |
|---|---|---|---|
| 10 | 0 | 0 | 0.00% |
| 20 | 871 | 3 | 3.06% |
| 30 | 7364 | 5 | 16.69% |
| 40 | 7566 | 6 | 16.70% |
| 50 | 7133 | 4 | 9.43% |

When a node exceeds the maximum transmission attempt to send a packet to the destination MAC, the node drops the frame at the MAC layer. It then informs the network layer that the next hop is no longer available and the network layer drops all subsequent packets destined for that MAC address.

The sample from the trace file that shows the behavior is below. From the trace, it is seen that at 41. 155586639, node 5 drops the frame with RET error. MAC layer informs the network layer and the network layer drops subsequent data packets, 1288, 1289, 1290, 1291, and so on.

D 41.155586639 _5_ MAC  RET 0 RTS 44 [58e 6 5 0]

D 41.155586639 _5_ RTR  CBK 5153 cbr 40 [13a 6 5 800] ------- [5:0 4:1 30 6] [1288] 0 0

D 41.155586639 _5_ RTR  CBK 5157 cbr 40 [0 6 5 800] ------- [5:0 4:1 30 6] [1289] 0 0

D 41.155586639 _5_ RTR  CBK 5161 cbr 40 [0 6 5 800] ------- [5:0 4:1 30 6] [1290] 0 0

D 41.155586639 _5_ RTR  CBK 5165 cbr 40 [0 6 5 800] ------- [5:0 4:1 30 6] [1291] 0 0

Tables 16, 17, and 18 show the variations in the average retransmission count for the variations in the network size for various transmission rates. The columns of these tables represent the network size (NS), average number of transmission attempts (ATA), and percentage increase in transmission attempts (PITA). To PITA, the number of attempts required for 10 nodes is taken as a base to compute the other nodes.

TABLE 16

VARIATIONS IN AVERAGE RETRANSMISSIONS COUNT FOR VARIATIONS IN NETWORK SIZE FOR DATA TRANSMISSION RATE OF (G.729) 8Kbps

| NS | ATA | PITA |
|---|---|---|
| 10 | 1.25 | 0.00% |
| 20 | 1.48 | 15.54% |
| 30 | 2.23 | 43.95% |
| 40 | 3.29 | 62.01% |
| 50 | 4.58 | 72.71% |

TABLE 17

VARIATIONS IN AVERAGE RETRANSMISSIONS COUNT FOR VARIATIONS IN NETWORK SIZE FOR DATA TRANSMISSION RATE OF (G.723) 6.3Kbps

| NS | ATA | PITA |
|---|---|---|
| 10 | 1.24 | 0.00% |
| 20 | 1.38 | 10.14% |
| 30 | 1.85 | 32.97% |
| 40 | 2.83 | 56.18% |
| 50 | 3.95 | 68.61% |

TABLE 18

VARIATIONS IN AVERAGE RETRANSMISSIONS COUNT FOR VARIATIONS IN NETWORK SIZE FOR DATA TRANSMISSION RATE OF (G.723) 5.3Kbps

| NS | ATA | PITA |
|---|---|---|
| 10 | 1.24 | 0.00% |
| 20 | 1.37 | 9.49% |
| 30 | 1.68 | 26.19% |
| 40 | 2.67 | 53.56% |
| 50 | 3.58 | 65.36% |

Tables 19, 20, and 21 present the variations in frame service-time for the variations in the network size. The columns of this table represent the network size (NS) and the average MAC service-time (AMST). The average MAC service-time indicates the time gap between the arrival and departure of the frame at the MAC layer. Once a packet is received from the network layer, the MAC layer will send or forward the frames to the destination MAC once the medium is free. The packet must wait until the medium is free, which causes a delay in transmission. Table 22 presents the variations in total packet drop in burst losses for the network sizes of 10 through 20, with a step size of 1.

TABLE 19

VARIATIONS IN EXPECTED MAC SERVICE-TIME WITH VARIATIONS IN NETWORK SIZE (NS) FOR PACKET TRANSMISSION RATE OF (G.729) 8Kbps

| NS | AMST |
|---|---|
| 10 | 0.051246 |
| 20 | 0.128543 |
| 30 | 0.298536 |
| 40 | 0.654693 |
| 50 | 0.858824 |

TABLE 20

VARIATIONS IN EXPECTED MAC SERVICE-TIME WITH VARIATIONS IN NETWORK SIZE (NS) FOR PACKET TRANSMISSION RATE OF (G.723) 6.3Kbps

| NS | AMST |
|---|---|
| 10 | 0.026534 |
| 20 | 0.134256 |
| 30 | 0.276486 |
| 40 | 0.584896 |
| 50 | 0.768853 |

TABLE 21

VARIATIONS IN EXPECTED MAC SERVICE-TIME WITH VARIATIONS IN NETWORK SIZE (NS) FOR PACKET TRANSMISSION RATE OF (G.723) 5.3Kbps

| NS | AMST |
|---|---|
| 10 | 0.010859 |
| 20 | 0.123576 |
| 30 | 0.214839 |
| 40 | 0.532173 |
| 50 | 0.697346 |

TABLE 22

VARIATIONS IN TOTAL DROPS FOR VARIATIONS IN NETWORK SIZE FROM 10 TO 20 FOR PACKET TRANSMISSION RATE OF (G.729) 8Kbps, (G.723) 6.3Kbps, AND (G.723) 5.3Kbps

| NS | TDP (8Kbps) | TDP (6.3Kbps) | TDP (5.3Kbps) |
|---|---|---|---|
| 10 | 0 | 0 | 0 |
| 11 | 0 | 0 | 0 |
| 12 | 473 | 0 | 0 |
| 13 | 1202 | 0 | 0 |
| 14 | 2353 | 0 | 0 |
| 15 | 2374 | 106 | 53 |
| 16 | 2408 | 670 | 85 |
| 17 | 2458 | 726 | 656 |
| 18 | 2489 | 743 | 708 |
| 19 | 2515 | 954 | 863 |
| 20 | 2537 | 892 | 868 |

## 4.3 Analysis of Simulation Results

This section analyzes the simulation results presented in the above sub-section. Table 13, 14, and 15 present the effect of MAC contention caused by the increase in the network size on the burst losses. The total number of packet drops associated with all the burst losses in the network increased with an increase in the network size. This pattern was also observed in the packet drop percentage and the maximum count of burst losses. This pattern remains constant for the network sizes of 10, 20, 30, and 40 nodes. For a network size of 50, the burst losses decrease, resulting in overall packet drops. This is because of the internal IFQ drops at the nodes caused by the inability of the node to transmit the packets due to high MAC contention. This pattern was seen for all the data transmission rates employed in the network. For network size of 30, the percentage of packet drops due to RET error is highest, as compared to other network sizes. As the number of nodes in a network increases, contention also increases, so the number of retransmission attempts increases, and so do the corresponding drops. On increasing the network size from 30 to 40, there is a significant increase in the internal IFQ drop. Since the packets are dropping before arriving at the buffer itself, there is a decrease in the percentage of packet drops. This pattern is seen on increasing the network size from 40 to 50.

However, the frequency of occurrence of burst losses is very high for high transmissions rates, and low for lower transmission rates. This is due to increment in the contention with the increase in data rates. Figure 5 shows the variations in the total packet drops in burst losses for the variations in the network size for data transmission rates of 8Kbps, 6.3KBPS, and 5.3Kbps.

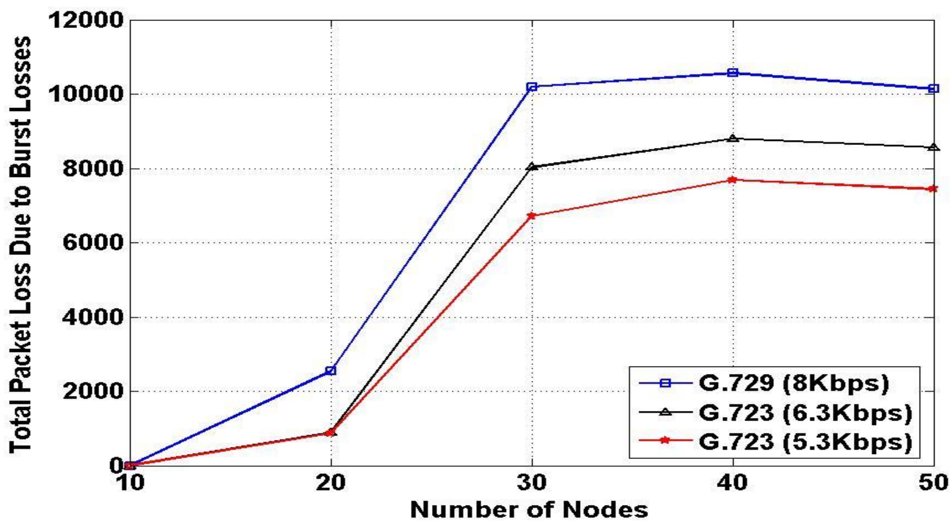


Figure 5. The variations in the total packet drops in burst losses, for the variations in the network size for data transmission rates of 8Kbps, 6.3KBPS, and 5.3Kbps

Tables 16, 17, and 18 show the effect of the increase in the network size on the MAC contention in terms of the average number of transmissions required for successful delivery of a frame to the MAC destination. The number of average transmissions required for the frame's successful delivery increases with an increase in the network size. Figure 6 shows the variations in the average transmission attempts count for the variations in the network size for the data transmission rates 8Kbps, 6.3KBPS, and 5.3Kbps. This pattern was observed to be followed for all transmission rates for an increase in network size. An important observation in this regard is that the average retransmissions required for the successful delivery of packets increases rapidly for the network adopting higher transmission rates over those with lower transmission rates.

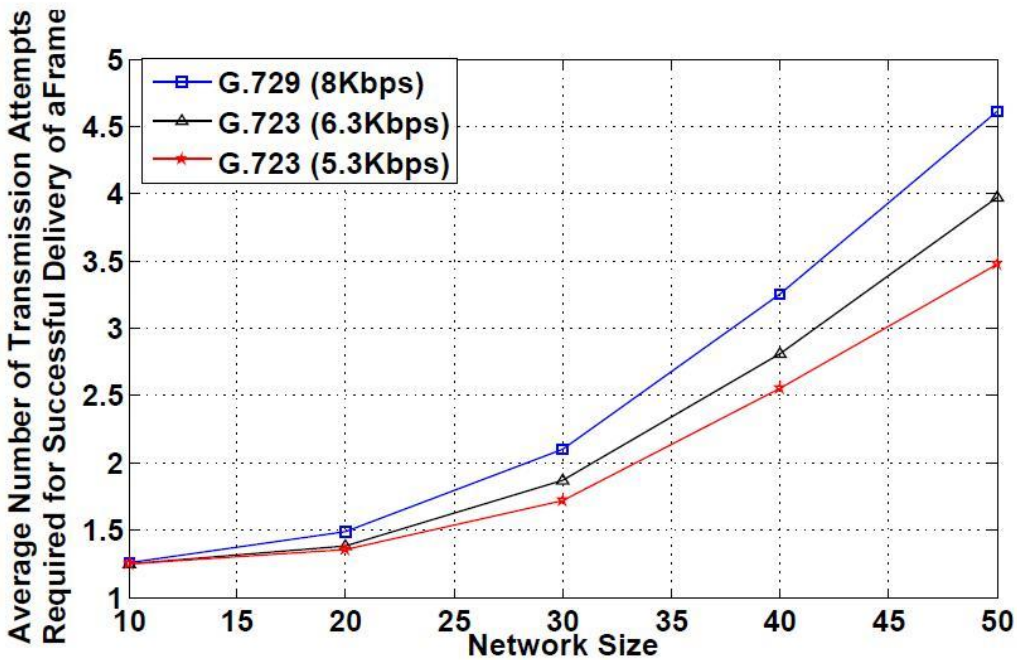


Figure 6. The variations in the average transmission attempts count, for the variations in the network size for data transmission rate 8Kbps, 6.3KBPS, and 5.3Kbps

The average transmission required for successful delivery of a frame in a network of size 30 nodes with the data transmission rates of 5.3Kbps, 6.3Kbps, and 8Kbps is 1.7, 1.8, and 2.1 respectively. Therefore, the average transmission required for successful delivery of a frame in the network size increases with an increase in the data transmission rate. Also, the values associated with the percentage increase of number of transmission attempts shown in Tables 7, 8, and 9 support the above analysis. As the data transmission rate increased each node attempted to deliver more packets within a particular second. Because all the nodes in the network follow the same pattern, the MAC contention increases, causing more collisions and back off. Hence, the average frame service-time also increases with an increase in the data transmission rates for the networks of the same sizes. This pattern can be seen in Tables 19, 20, and 21. Figure 7 shows the variations in the average MAC service-time, for the variations in the network size for the data transmission rate 8Kbps, 6.3Kbps, and 5.3Kbps.

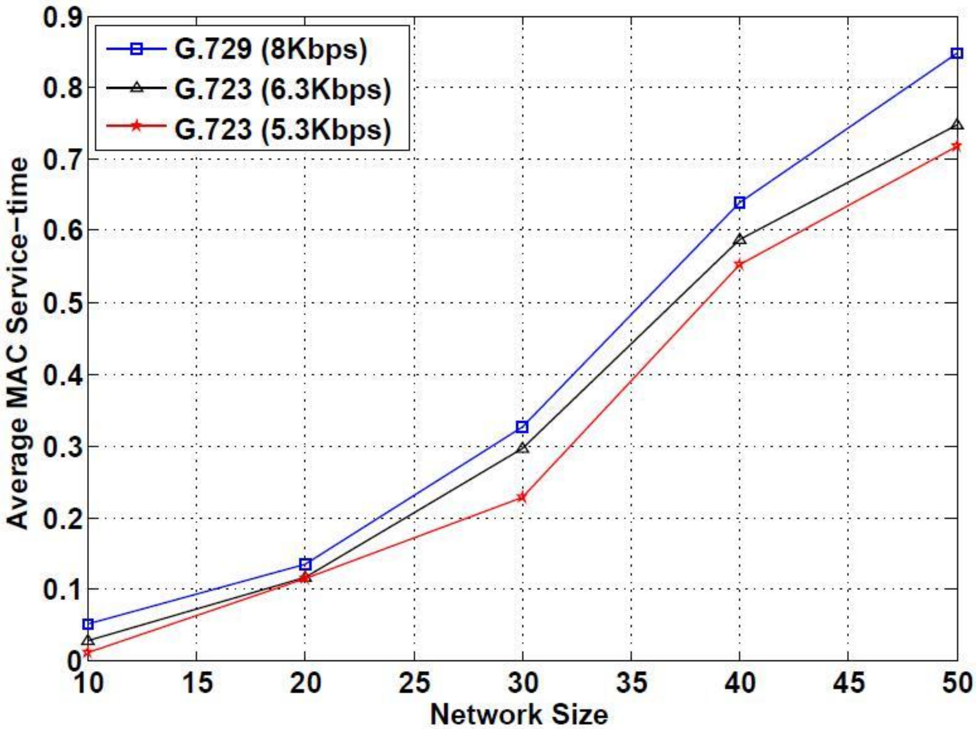


Figure 7. The variations in the expected MAC service-time, for the variations in the network size for data transmission rate (G.729) 8Kbps, (G.723) 6.3KBPS, and (G.723) 5.3Kbps

From the above analysis, it can be seen that the burst losses increase with an increase in the network size and the data transmission rates. Hence, the burst losses can be minimized by fine-tuning the network size and data transmission rate within the saturated network. The simulation results show that the burst losses can be restricted within the desired limit for a combination of network size and data transmission rate. Figure 8 presents the maximum network size that can be adopted for different data transmission rates to restrict the burst losses to zero. The combination of (11nodes, 8Kbps), (14nodes, 6.3Kbps), and (14nodes, 5.3Kbps) restricted the burst losses to zero.

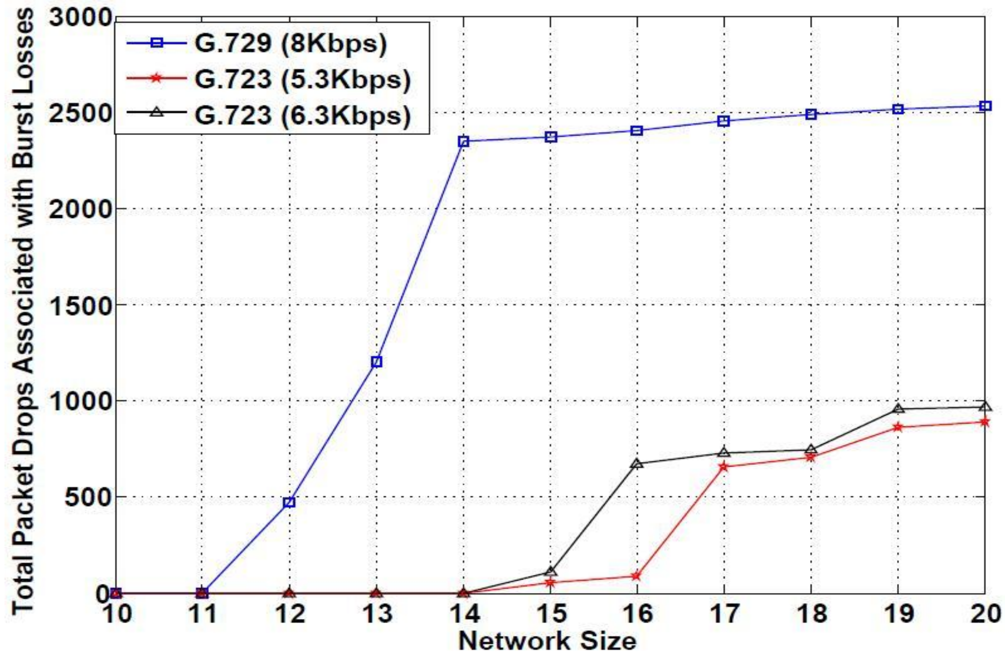


Figure 8. The variations in the total packet drops for the variations in the network size for the data transmission rates of 8Kbps, 6.3Kbps, and 5.3Kbps

Thus, the impact of MAC contention on burst losses increases with an increase in the network size and the data transmission rate. The total packets dropped due to the burst losses increases along with the number of transmission attempts required for successful transmission of a frame and average MAC service-time. These lost packets severely degrade the quality of the real-time traffic. For applications such as VoIP, the decoder on the receiver cannot interpolate data correctly due to burst losses.

# CHAPTER 5

# CONCLUSIONS

This thesis uses semi-Markov process based model for illustrating the burst losses in the IEEE 802.11 saturated ad hoc networks. It describes the frame transmission process in the IEEE 802.11 networks and the effect of medium contention on burst losses. With an increase in the network size and data transmission rate the MAC contention increases, thereby increasing the burst losses in the network. Therefore, the network size and data transmission rates directly affect the medium contention and the burst losses in the network. Simulation results show that the total packet drops associated with the burst losses increase and then decrease with an increase in network size. This decrease in the total packet drop accounts for the internal interface queue (IFQ) drops at the nodes as a result of the node's inability to store the new packets because of unsuccessful transmissions due to high MAC contention. The total packet drops always increase with an increase in the data transmission rate. The expected value of MAC service-time for a packet also increases with an increase in contention in the network. Hence, an increase in the network size and data transmission rate severely affects the burst losses, thereby lowering the quality of real-time applications such as the VoIP. Thus, burst losses in real-time applications must be reduced for better application throughput. Further investigation shows that the combination of network size and data transmission rate must be appropriately fine-tuned to limit the burst losses below the desired values in the saturated ad hoc networks. From the simulation results, the quantitative analysis showed that the combination of network size and data transmissions rate of (11nodes, 8Kbps), (14nodes, 6.3Kbps), and (14nodes, 5.3Kbps) restricted the burst losses to zero. The simulation results showed that the number of nodes were consistent for all 3 scenarios. Depending on the requirement and environment of wireless network deployment, users can use specific number of nodes with their respective codecs to reduce the burst losses. This thesis not only presents and analyzes the problem, but also provides solution to minimize the burst losses to desired values for above-mentioned codecs.

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