

**MULTIMEDIA PACKET FORWARDING in 802.11  
NETWORKS WITH ACCESS POINT DIVERSITY**

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NETWORKS WITH ACCESS POINT DIVERSITY

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*To my family*

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## Abstract

The characteristics of the wireless communication channels are usually time-varying, where the quality of the received signal changes rapidly over time within a few packet transmissions. The time-variability of the wireless channel is due to noise, fading, interference and mobility. An important artifact of these effects is the bursty packet losses observed at the link layer, and multimedia applications such as voice over IP and video-on-demand require resiliency against packet losses.

Channel diversity is a well-known technique to alleviate the effects of time-variable wireless channels. In this thesis, we aim to apply link-layer diversity for resilient multimedia transmissions in IEEE 802.11 wireless networks. IEEE 802.11 wireless local area networks (WLANs) have widespread deployment in enterprises, public areas and homes.

In many cases, a mobile user has the option of connecting to one of several 802.11 access point (APs). Unlike the standard operation where a mobile user is connected to a single AP for the duration of a session, we consider the case where a user is connected to all available APs in the vicinity. This type of operation requires a new multi-access control (MAC) protocol, where the user has to decide to which AP the packet is forwarded. The selection of the AP for each packet is performed in our proposed MAC protocol based on the most recent channel observation and the collected long-term statistics of packet loss and burstiness, where the aim is to maximize the probability of successful transmission of each packet. Our analysis of the long-term channel statistics such as burst length and packet error rate also show that these parameters depend on the user load, channel coherence time and the number of users in the network. The proposed MAC protocol is analyzed using Qualnet network simulator, and it is shown that our proposed protocol can improve the efficiency of the system approximately by up to %25 over the standard 802.11 protocol where each user is associated with the AP that has the maximum received signal strength.

## Özet

Kablosuz haberleşme kanalları zamanla değişen bir özelliğe sahiptir ve bu değişim kanal üzerinde bir çok paketin iletimi boyunca devam eder. Kablosuz kanalların bu özelliği, alıcıda oluşan gürültü, çoklu sönmüleme, diğer kullanıcılardan kaynaklanan girişim ve kullanıcıların hareketliliğine bağlıdır. Bu özellikler veri katmanında patlamalı bir şekilde gerçekleşen paket kayıplarına neden olmaktadır ve kablosuz uygulamalar özellikle çoklu-ortam uygulamaları bu hatalara karşı bir esneklik içermektedir.

Kanal çeşitleme yöntemi zamanla değişen kanalların etkilerini önlemede sıkça kullanılan bir tekniktir. Bu tezde IEEE 802.11 ağlarında esnek çoklu ortam uygulamaları için veri katmanında çeşitleme yöntemini uygulamayı amaçladık.

Çoğu durumda bir kullanıcı birden fazla erişim noktasına aynı anda bağlı olmaktadır. Genel standardta kullanıcı bütün iletişim boyunca sadece bir erişim noktasına bağlı kalır. Genel standardtan farklı olarak, bu tezde kullanıcıların etrafında bulunan birden çok erişim noktasına bağlı kaldığı durumu göze aldık. Bu tür bir iletişim, hangi erişim noktasına iletim yapılacağını belirten yeni bir MAC protokolünün uygulanmasını gerektirmektedir. Erişim noktasının seçimi, en son gözlenen kanal durumuna ve kanalın patlama uzunluğu ve paket hata oranı gibi uzun dönem istatistiklerine bakılarak verilir. Buradaki amaç ise başarılı iletim olasılığını maksimum yapmaktır. Yaptığımız analizler, uzun dönem kanal özelliklerinin, kullanıcının trafik miktarına, kanalın uyum zamanına ve sistemdeki kullanıcı sayısına göre değiştiğini göstermektedir. Önerilen MAC protokolü Qualnet simülasyon programı ile analiz edildi ve önerilen MAC protokolünün standard uygulamaya göre sistem verimliliğini %25 oranına kadar geliştirebildiği gösterilmiştir.

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# Chapter 1

## Introduction

### 1.1 Motivation

Wireless channels have time-varying characteristics where signal quality changes rapidly in time. There are mainly five causes that make the wireless transmission complex: noise at the receiver, caused by electronics and thermal energy; attenuation caused by distance between receiver and transmitter and obstacles; interference from other transmitter; multipath [1] caused by atmospheric ducting, ionospheric reflection and refraction from such as buildings and user mobility which causes rapid changes in channels. These properties cause packet loss and delay. Many studies show that these packet errors occur in bursts in which two or more consecutive packets are dropped [2].

In many cases, a mobile user has the option of connecting to one of several 802.11 access points (APs), which forward packets between users in different wireless local area networks (WLANs), each using an independent channel. Users may split their traffic among all available APs, based on the channel burstiness, transmission rates and required application resilience. Such traffic splitting in the Internet among different Internet Service Providers (ISPs) is called multihoming and it was first proposed in [3].

Loss resilience is an important requirement in the design of indoor WLANs infrastructure. To cover a large area multiple APs are deployed and in common implementations stations (STAs) scan the wireless channel to find and associate with the AP which has the highest signal strength. In current WLANs, a station sends and receives data only via the AP which it is associated with. The stations switch to another AP if only the signal strength decays and several degradations occur. The transmission rate (often called the PHY rate) is also selected based on the signal strength. The PHY selects a higher rate if the signal strength is good and reduces the rate if the signal strength decays.

Among wireless applications multimedia applications such as video transmission and voice over IP applications have become very popular. These applications require resiliency against errors that occur in bursts. In many cases this resiliency is provided by source coding. Source adds redundancy in coding process to provide the necessary resiliency.

In this thesis we propose an efficient MAC protocol to distribute the traffic among APs in order to minimize redundancy and delay. In our protocol, the source keeps on transmitting packets until at least  $K$  packets are received by the destination then the original data can be successfully recovered. Due to channel errors and collisions the source needs to transmit more than  $K$  packets,  $N \geq K$ . We focus on the multihoming case where users split their traffic among available APs. In our protocol, each packet selection is based on the probability of successful packet transmission on each channel.

## 1.2 Objectives

One objective in this thesis is to find an efficient packet forwarding MAC protocol for multihomed users in WLANs. In this MAC protocol for each packet, the channel which has the highest probability of success is selected according to the long-term channel statistics and the outcome of the most recent transmission. Our splitting policy is made dynamically during all packet transmissions. We call our MAC protocol Multihomed Medium Access Control (MH-MAC).

Without any encoding/decoding process in this thesis, we assume that a source transmits  $N$  packets to a destination until the destination receives  $K$  packets successfully. If the destination receives  $K$  packets, the transmission is successful. Our performance parameters are efficiency, which compares the amount of redundancy and delay that is the difference of time between the time that the first packet is received and the time that the last packet is received. We compare MH-MAC with fixed path scheme which is the standard case where users associate with only one AP during all transmissions and there is no a splitting policy in terms of these performance parameters.

### 1.3 Contributions and Thesis Organization

Our main contributions in this thesis can be summarized as follows:

- The effects of the coherence time of the wireless channel, network traffic and the number of users on burst length and error rate are exploited.
- A new MAC protocol for multihomed users is developed.
- A realistic multi-user scenario is considered and two main problems that occur in multi-user case namely collision and delay due to other user transmission are defined and solutions to these problems are found.

The organization of the thesis is as follows. In Chapter 2, a general introduction and background of the problem are given with related works. In Chapter 3, the channel model and packet loss models are introduced. The effect of three parameters, load, number of users and usage of RTS/CTS on burst length and packet error rate are investigated. By exploiting them, a more efficient protocol can be designed. In Chapter 4, we introduce a new MAC protocol for our problem. An analytical model is defined for this MAC protocol. Two main problems in multi-user case are explained and we propose solutions to these problems. In Chapter 5, we present simulation results for various scenarios and parameters to show the performance

of our MAC protocol. Chapter 6 explains the simulation environment and our simulation network simulator, Qualnet. Chapter 7 concludes the thesis.



# Chapter 2

## Background

### 2.1 IEEE 802.11 Architecture

An 802.11 wireless local area network (WLAN) is based on a cellular architecture where the system is divided into cells namely Basic Service Set (BSS) where each cell is controlled by a base station called access point (AP) as seen in Figure 2.1. Multiple BSSs form a Extended Service Set (ESS) and ESS is connected to Distributed System that connects APs to different networks.

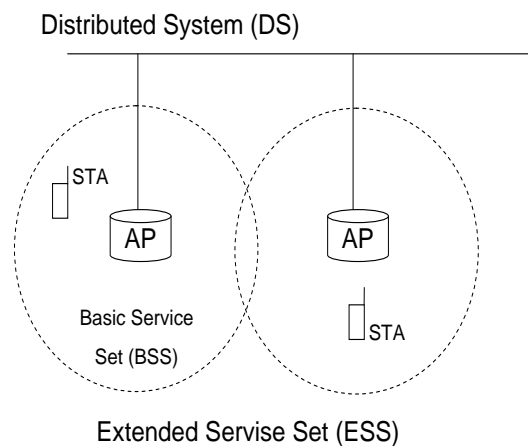


Figure 2.1: 802.11 Architecture

802.x standards define the Medium Access Control (MAC) and Physical Layer (PHY). MAC performs the fragmentation, retransmission, acknowledges and how a station accesses to channel. MAC layer defines two different access method: Distributed Coordination Function (DCF) and Point Coordination Function (PCF).

### **2.1.1 Distributed Coordination Function**

Distributed Coordination Function is based on basically Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). CSMA/CA works as follows: A station that wants to transmit first senses the channel, if the medium is busy it defers its transmission. If the medium is free for a specified time called Distributed Inter Frame Space (DIFS) then the station is allowed to transmit. If there is no collision and channel errors, the receiver sends an acknowledgement frame (ACK) that indicates the transmission is successful. If there is collision or unsuccessful transmission due to the channel errors then the station retransmits the packet until it gets an ACK or drops after a number of retransmissions.

In order to reduce the probability of collision between two stations that can not hear each other, the standard defines virtual carrier sense mechanism. In this mechanism, a station willing to transmit a packet, first transmits a short control packet called Request to Send (RTS), which includes source and destination address and the duration of the transaction. If the medium is free the destination station responds with a response control packet Clear to Send (CTS) which includes the same duration information. Then, all stations received RTS or CTS set their virtual carrier sense indicator called Network Allocation Vector (NAV) for the given duration then the stations use this information when sensing the medium.

### **2.1.2 Backoff Algorithm**

Backoff is a well known method to resolve the contention between stations willing to access to medium. The method requires for each station to choose a random number between 0 and a given number and wait until this number of slots are available before accessing to the medium. Figure 2.2 shows the access mechanism.

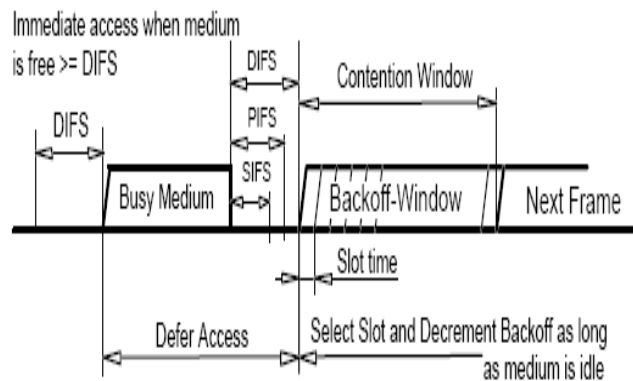


Figure 2.2: Access Method

The short interframe space (SIFS) is the shortest interval time followed by the slot time. The priority interframe space (PIFS) is equal to SIFS plus one slot time. The DIFS is equal to the SIFS plus two slots times.

### 2.1.3 Management Controls

Seven MAC frame types are dedicated to management control in WLANs.

- **Beacon Frame:** It is used to identify an AP. The access point periodically sends a beacon frame to announce its presence and relay information, such as time stamp and service set identifier (SSID). Stations continually scan all 802.11 radio channels and listen to beacons as the basis for choosing which access point is best to associate with.
- **Authentication Frame:** An authentication frame is used to authenticate with an AP before the station proceeds to associate with the wireless network.
- **Deauthentication Frame:** A station sends a deauthentication frame to the AP which it is currently authenticated with if it wishes to terminate secure communications.
- **Association request Frame:** : A station sends an association request frame to associate with an AP. The frame includes information regarding the

association, such as association ID and supported data rates.

- **Reassociation request Frame:** : If a station disassociates with the current AP and finds a new AP to associates with, it sends reassociation request frame to the new AP.
- **Reassociation response Frame:** : An access point sends a reassociation response frame containing an acceptance or rejection notice to the station requesting reassociation. The frame includes information regarding the association, such as association ID and supported data rates.
- **Disassociation Frame:** : A station sends a disassociation frame to the AP that it currently associates with, if it wishes to terminate the association

## 2.2 Path Diversity

Path diversity is a transmission technique that sends data packets over two or more paths (channels) in a packet-based network. The paths may contain one or multiple sources. Several research studies have been devoted to improving the reliability and performance of networks especially the Internet by using disjoint communication channels between source and destination. In the literature, there are two main proposals to increase the performance of the Internet. First approach is multihoming and the other one is overlay networks. Overlay networks refer to a virtual network that includes overlay nodes as in Figure 2.3. Overlay nodes cooperate with each other to send a packet to any destination. These nodes form an overlay network. It is possible to find a different path from the current path if the current path fails [3],[4],[5].

Application layer and physical layer approaches of path diversity are compared by Laneman et al. [7]. In [8],[9] Wang et al. and in [10] Zimmermann et al. show the benefits of path diversity in wireless ad hoc networks. Another work for wireless ad hoc networks is done by Jain and Das [11]. In [11], the authors develop an anycast mechanism at the link layer to exploit the path diversity by choosing the best next hop. An information theoretical point of path diversity are studied in [12],[13],[14].

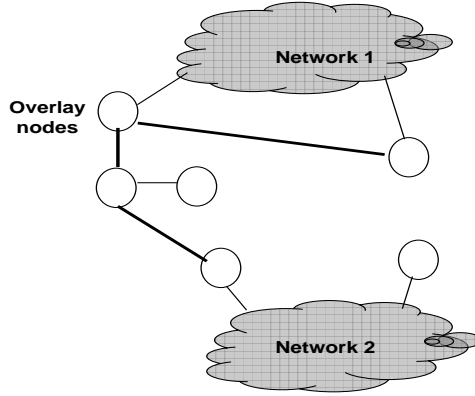


Figure 2.3: Overlay Networks

Path diversity has been used with coding techniques to combat the packet losses. Especially, the works done in [15],[16],[17],[18] for video application use and explain path diversity with coding.

The works in [20] by Golubchik et al. and in [21] by Abdouni et al. are most related to our work. They follow a similar approach for coding data and channel model. However they do not have a transmission strategy with as we propose in this thesis. In addition, their performance metrics are different. Another related work to ours is proposed in [22]. Tsirigos and Haas consider multiple channels in ad hoc wireless scenarios and their strategy is to maximize throughput by using diversity coding. They try to maximize the packet success probability but their transmission strategy is not based on packet by packet splitting policy that we use in our work. The work by Vergetis et al [23] is another work related to ours. They consider a probabilistic transmission splitting policy. Their main objective is to develop path diversity policy that is easy to implement. They argue that if all channels are equal the benefits of path diversity can be provided. However we do not consider that all channels are equal in this work. We show that to get the benefits of path diversity the packets should be transmitted over those channels where the unsuccessful transmission probability is the least.

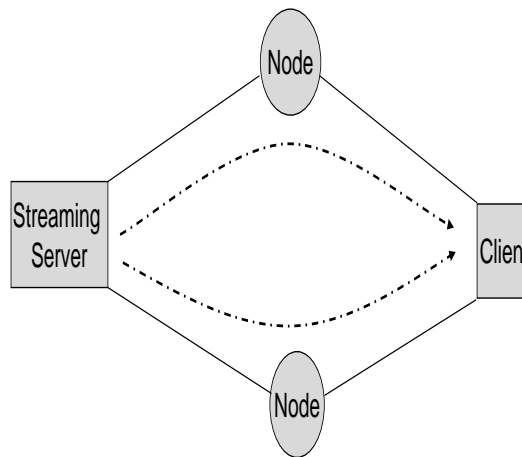


Figure 2.4: Path Diversity

### 2.2.1 The Benefits of Path Diversity

There are many reasons described below to use multipath diversity :

- ***Reduce burst length and improve the quality of the delivered data*** ; Wireless channel has bursty characteristics. A large number of consecutive packet losses , burstiness, not only causes significant degradation in signal quality but also obstructs correcting packet losses with error correction codes. Multipath diversity can decrease burst length of channel and improve the quality of transmission.
- ***Increase throughput*** ; Sending data over multiple paths increases amount of aggregated bandwidth and thus increases the throughput.
- ***Adapt network variation***; Transmission may take too much time especially for multimedia application. During the transmission network conditions may change. However not all paths experience the same changes such as losses or congestion. Thus with path diversity, one can easily adapt channel changes.
- ***Decrease delay***; Sending data over multiple paths decreases delay which is important to real-time application by decreasing burst length of channel.

- **Security**; An attacker can attack any node in the network. Sending data by using multipath diversity increases security because of decreasing the probability of attacking to a node over the multiple paths [24], [25].

It is important to resolve the following issues;

- During transmission some paths may contain bottleneck links. It should be determined that how many paths are needed for efficient transmission.
- Another issue is that one should decide on what kind of an error correcting codes should be used and one should decide effect of the redundant information on the transmission.
- The important concept is that how many packets should be transmitted over a path. This is important for quality of service and delay consideration.

### 2.2.2 Multihoming

Traditionally most networks have only one connection to the Internet [26]. To achieve required redundancy and to mitigate the channel effects it is possible to connect to the Internet over more than one connection. Multihoming refers to a kind of path diversity where a network has multiple connections to Internet. The internet protocol versions Ipv6 and Ipv4 can use multihome but multihoming is not yet a standard [27]. Multihoming future came back in 1980s but it was not used for the Internet widely. However, today multi-homed networks take place of a single network [28],[29]. While Internet traffic engineering and techniques increase multi-homed network is deployed widely [30],[31],[32].

There are many works on multi-homed networks in literature. Many of them are interested on path selection which is a critical part for multihomed users because it is difficult to decide which path is available or better than other path at a transmission time. In [34], the authors propose an active measurement for path selection. They try to modify stream control transmission protocol (SCTP) to support real time applications. In [34], the authors present an algorithm for delay-sensitive packets over multiple burst loss channels. They drive the packet loss ratio when Forward

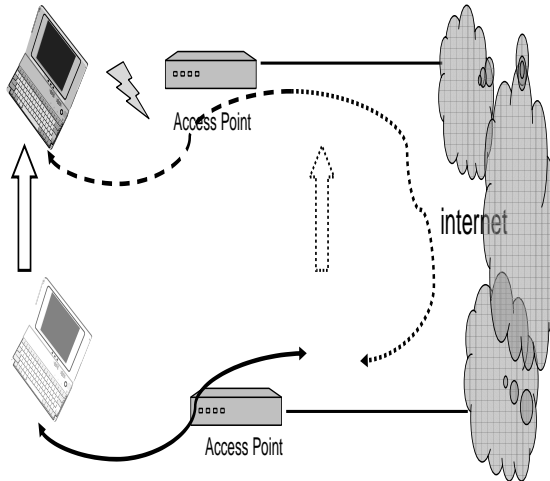


Figure 2.5: Multihoming

Error Correction (FEC) is used and they develop a dynamic programming to strip traffic optimally for FEC.

Other path selection algorithms for multihomed networks are proposed in [35],[36], [37]. A comparison study for multihomed mobile hotspot is done in [36]. The authors in [35] compare two different distribution schemes. While first scheme assumes no a priori knowledge of network change the second scheme knows the priori knowledge of the changes. In [38], they try to find the best available paths for a real-time application. They develop an improved transport layer protocol namely stream control transmission protocol (SCTP) called WiSE (Wireless SCTP Extension). WiSE is interested in congestion and channel losses. The main idea is to develop a new congestion control and multihoming feature to optimize resource utilizations.

## 2.3 Error Resilience

Many wireless standards based on IEEE 802.11 standards perform some physical layer adaptation which employs mostly coding techniques for error detection. Also these standards use re-transmission function to recover the packet loss. However for some real-time application re-transmission based recovery is not viable option due to delay sensitivity of real-time application. In the literature, there has been significant



previous work on error control, recovery, and concealment techniques that discuss different approaches to increase the quality of real-time applications such as forward error correction (FEC) [39] or error concealment algorithm [40].

### 2.3.1 Forward Error Correction

Forward error correction (FEC) is a method for error control and recovery for data transmission where the receiver has the capability to detect and correct fewer than a predetermined number of bits or symbols corrupted by transmission errors [41]. FEC is accomplished by adding redundancy to the transmitted data using a predetermined algorithm. Adding redundancy with FEC coding reduces data bandwidth of the channel since a portion of the channel bandwidth is used for parity bits of the FEC code. FEC techniques, however, are indispensable for channels with high error rates, e.g. wireless channels. The two main categories of FEC are block coding and convolutional coding. Block codes operate on fixed-size blocks of bits or symbols of predetermined size, while convolutional codes consider bit or symbol streams. Block and convolutional codes are frequently combined in concatenated coding schemes in which the convolutional code does most of the work and the block code detects any errors made by the convolutional decoder.

### 2.3.2 Diversity Coding

Diversity coding is an error control based approach. It is introduced for selfhealing and fault-tolerance in digital communication networks [42]. The scheme achieves nearly instantaneous recovery from channel errors. Channel errors are treated as an erasure channel problem. Recovery from channel errors is done at the receiver side. There is no need for a feedback channel if there is only a single destination. In the proposed scheme, the information bits are divided into  $N$  data channels in which the data is sent in parallel.  $M$  extra channels are added in order to recover from any  $M$  channel failures out of  $N + M$  channels.  $M$  extra channels are used for transmitting the parity bits which are constructed from information bits by linear transformations. This is called  $M$ -for- $N$  diversity coding. However we do not use any real diversity coding. In our transmitting process, the source

keeps on transmitting packets ( $N$  packets) until the destination receives  $K$  packets successfully. When the destination receives  $K$  packets then original data can be recovered.

## 2.4 Wireless Channel Model

The propagation characteristics of wireless channels vary according to the type of the communication system. Terrestrial and cellular wireless communications can be given as systems having different propagation characteristics. In terrestrial communication, propagation is point to point and only channel noise is added to the received signal. This type of communication systems are modeled with additive white gaussian noise (AWGN) channel model where zero mean noise having a Gaussian distribution is added to the signal, [43]. On the other hand, cellular systems have multiple propagation paths due to the obstructing materials in the environment. Multipath propagation causes fading in the received signal. Fading is divided as slow fading and fast fading. Slow fading is the result of large reflectors and diffracting objects along the transmission path, [43]. However, fast fading is the rapid variation of signal levels when the receiver moves in short distances, [41].

In a typical multipath environment, the envelope of the received signal is represented with Rayleigh distribution where received instantaneous signal to noise ratio (SNR),  $\gamma$ , is distributed exponentially with probability density function, [43];

$$p(\gamma) = \frac{1}{\gamma_0} \exp\left(-\frac{\gamma}{\gamma_0}\right) \quad (2.1)$$

where  $\gamma_0$  is the average SNR.

### 2.4.1 Gilbert-Elliott Model

Rayleigh fading channel can be represented by using Markov models. A classical approach is two-state Gilbert-Elliott model in [44], [45]. In this model, the channel can be at one of two states at a time. One state shows erroneous case (bad state), while the other shows free of error case (good state). The probability of being in a state is called steady state probability, and moving from one state to the other

is called transition probabilities. The objective of the Gilbert-Elliott model is to calculate steady state and transition probabilities depending on the movement speed of the mobile receiver and calculate bit-error rates occurring at each state. In this thesis we use Gilbert model as our channel model.

### 2.4.2 Finite State Markov Channel Model

Gilbert-Elliott model is a useful model to an extent, because it does not consider interim conditions of the channel. That means the channel does not behave always as good or bad. Therefore, finite state Markov channel (FSMC) model is used for more exact representation of the channel. FSMC is considered in the works, [46], [47] where the general objective is the same as the Gilbert-Elliott channel model. However when number of states increases complexity of Markov chain also increases.

The FSMC models for fading typically model amplitude variations only. A detailed FSMC model for Rayleigh fading was developed in [47]. In this model the time varying SNR  $\gamma$  lies in the range  $0 \leq \gamma \leq \infty$ . The FSMC model discretizes this fading range into regions so that the  $j$ th region  $R_j$  is defined as  $R_j = \{\gamma : A_j \leq \gamma < A_{j+1}\}$ . This model assumes that  $\gamma$  stays within the same region over time interval  $T$  and can only transit to the same region or adjacent regions at time  $T + 1$ . Thus, given that the channel is in state  $R_j$  at time  $T$ , at the next time the channel can only transition to  $R_{j-1}$ ,  $R_j$ ,  $R_{j+1}$  and the transition probabilities between regions are derived in [47] as

$$p_{j,j+1} = \frac{L_{j+1}T}{\pi_j}, p_{j,j-1} = \frac{L_jT}{\pi_j}, p_{j,j} = 1 - p_{j,j+1} - p_{j,j-1} \quad (2.2)$$

Where  $L_j$  is the level crossing rate at  $A_j$  and  $\pi_j$  is the steady-state distribution.

# Chapter 3

## Burst Length and Error Rate of Loss Process

### 3.1 Long-Term Channel Characteristics

Links between communicating nodes are wireless channels. Wireless channels are frequently exposed to bursts of errors due to noise, fading/shadowing effects, interference, etc. The channels are usually not memoryless, which means errors on the channel are correlated. So the assumption that packet losses are independent is not appropriate. Packet losses occur in bursts and many of these bursty errors are on the order of ten packets.

In any transmission opportunity, a successful packet transmission has a probability. This probability depends on transmission rate, channel conditions such as fading and noise, collision and mobility. These factors are modeled as a finite state Markov chains in [47],[48] and transitions between states of different packet loss rate occur according to transition probabilities. The studies done in [46],[47],[48] show that packet losses occur in burst and to model this kind of error requires multi-state Markov model. Willing et al. in [49] show an experimental results over 2.4 GHz 802.11b radios and their results show that packet errors are bursty. Koksal et al. in [51] present a similar work that shows packet error characteristics and packet error process have a long-term dependency.

We consider a typical network scenario where there are more than one nodes

transmitting and we showed unlike others works that long-term channel characteristics mainly depend on number of nodes transmitting, usage of RTS/CTS and node load (traffic). To show the changes of the characteristics with the number of active nodes, traffic and usage of RTS/CTS, we run simulation. In this simulation, there are 10 nodes transmitting to an AP, the maximum distance between the AP and a station is 100 meters and the minimum distance is 50 meters, coherence time is 20 ms, retransmission function is disabled, all nodes use CBR application with 1500 bytes fixed packet size, channel are Rayleigh fading channel. We run the simulation for three different seed values. Error burst length and packet error rate are averaged over 10000 packets and only the results of one node are shown. Our simulations are performed in Qualnet network modeler [57]. Simulation environment is shown in Figure 3.1.

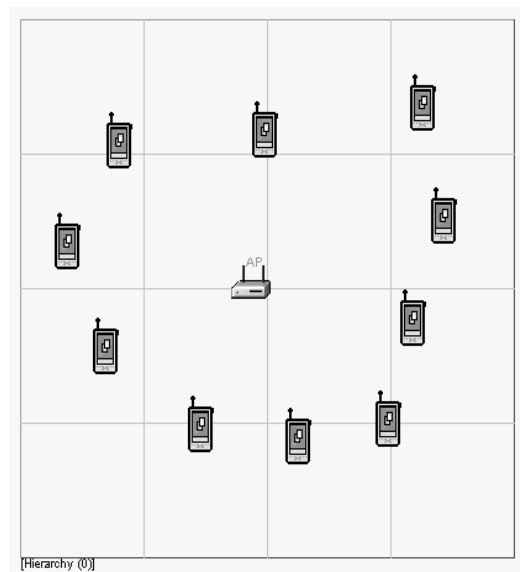


Figure 3.1: Simulation Environment

## 3.2 Number of Nodes and Burst Length

In a typical WLAN there are many users transmitting. If the number of users increases channel access delay in the network increases. Channel access delay is

the delay that is the total time for a station contending for a channel to wait before the contention is successful. Burst length depends on this delay in the network. For example, let  $STA_i$  represents the lost of packet  $i$  sent by  $STA$ . Then,  $P(STA_{i+k}|STA_i)$  for  $k \geq 0$ , represents the auto-conditional loss probability that the  $(i + k)^{th}$  packet is lost, given that  $i^{th}$  packet is lost in the same packet stream. When the delay is taken into account, this auto-conditional loss probability will not be realistic because channel correlation will decay due to the delay and the memoryless property of the channel will fail. The delay is not only due to number of users but also control packets, carrier sensing and hand-off, etc. The change of burst length with the number of users is shown in Figure 3.2. As expected as the number of users increase delay will increase. At this time the channel is not correlated any more and burst length diminishes.

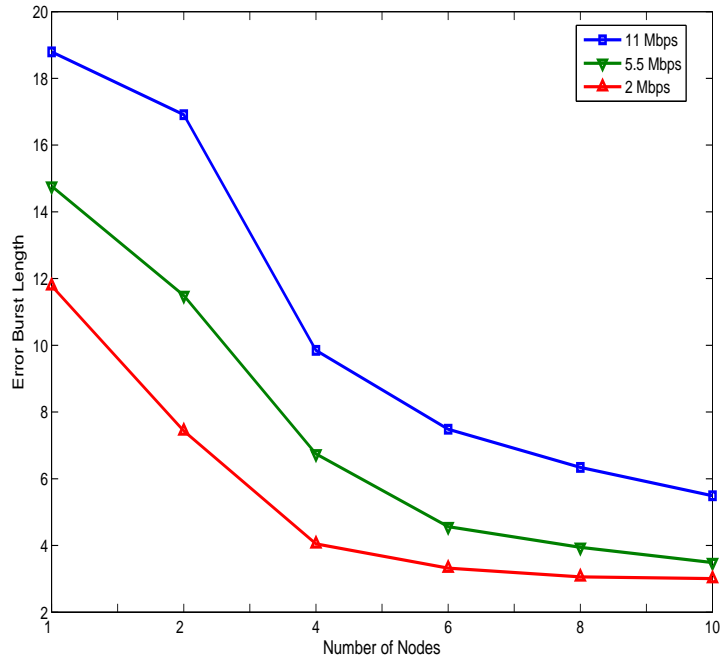


Figure 3.2: Error Burst Length via Number of Nodes

RTS/CTS function is enabled in this simulation. Nodes transmit 10000 packets at a rate of 250 packets per second. Mean burst length is the number of packets lost in a burst of two or more consecutive packets divided by the number of occurs

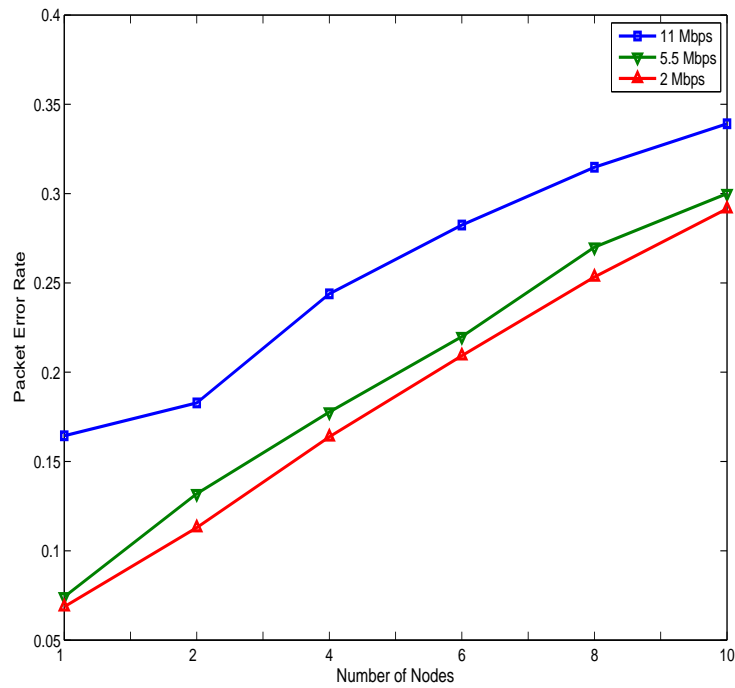


Figure 3.3: Packet Error Rate via Number of Nodes

of bursty case. Mean packet error rate is the number of packets lost divided by the total number of packets sent.

Figure 3.3 shows the packet error rate of the channel for the node. The longest burst length occurs when 11 Mbps PHY rate is used because higher data rate causes more packet losses. The smallest burst length is at 2 Mbps. While burst length decreases packet error rate increases. This is due to mainly the RTS and less probably data collision. Figure 3.4 shows the delay that time difference between the first packet access time to the channel and the last packet access time to the channel. As the number of nodes increases the channel access delay also increases. The least delay occurs with 11 Mbps as expected.

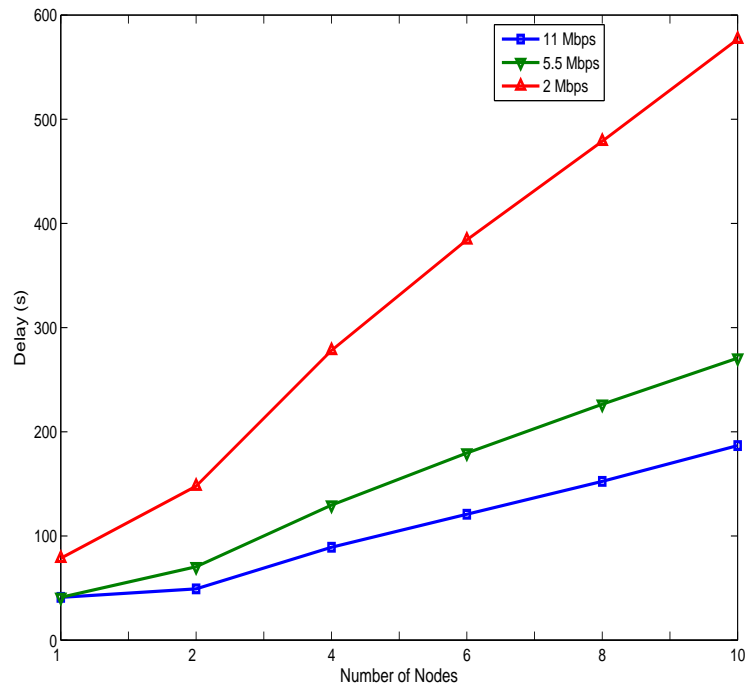


Figure 3.4: Channel Access Delay via Number of Nodes

### 3.3 User Load and Burst Length

The other effect that changes burst length is node load. Node load depends on the packet size and packet inter-arrival time. If inter-arrival time is low, the load is high. For example let packet size be 1000 bytes and inter-arrival time is 1 millisecond. Then, the load is 8 Mbps. If inter-arrival time increases the number of packet sent a particular interval decreases. When this case is considered for burst length calculation for high inter-arrival time the number of packet in a bursty period decreases and burst length also will decrease. Node load is changed only by inter-arrival time. To show the effect of only inter-arrival time on burst length, there is only one transmitting node. Figure 3.5 shows burst length.

While burst length increases with decreasing inter-arrival time, packet error rate shows a slight decreasing as in Figure 3.6. Changes in burst length depend on channel coherence time. If the coherence time is longer than inter-arrival time, many consecutive packet will be effected by same coherence time and this will increase burst length. Otherwise, the consecutive packets will not effected in the same way



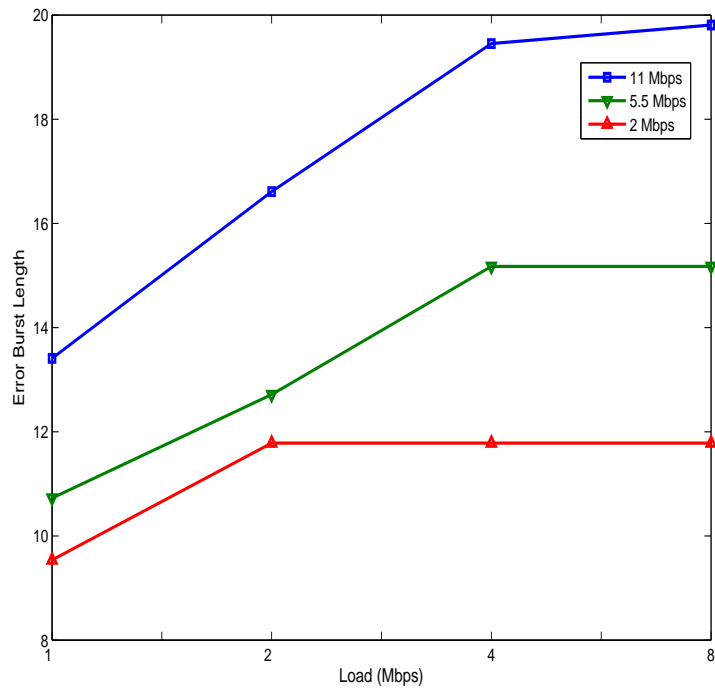


Figure 3.5: Error Burst Length via Node Load

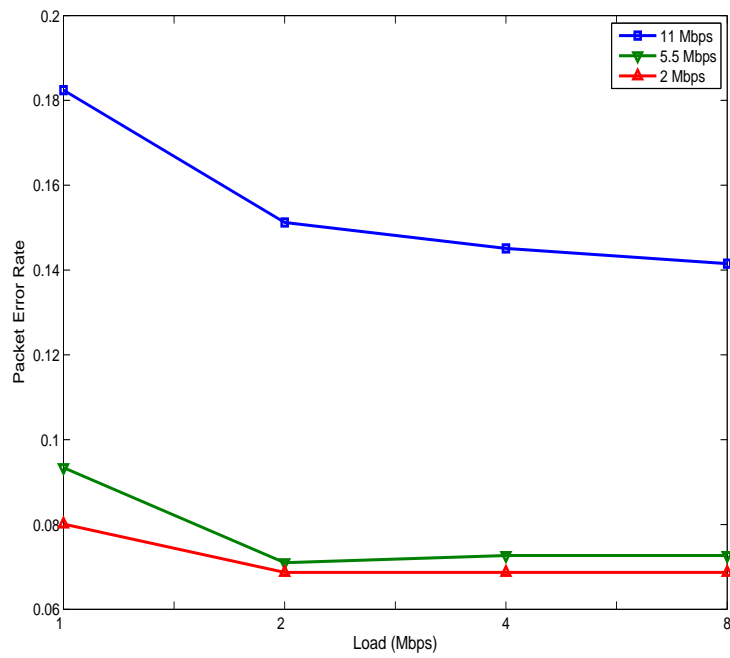


Figure 3.6: Packet Error Rate via Node Load

by the channel and burst length will decrease.

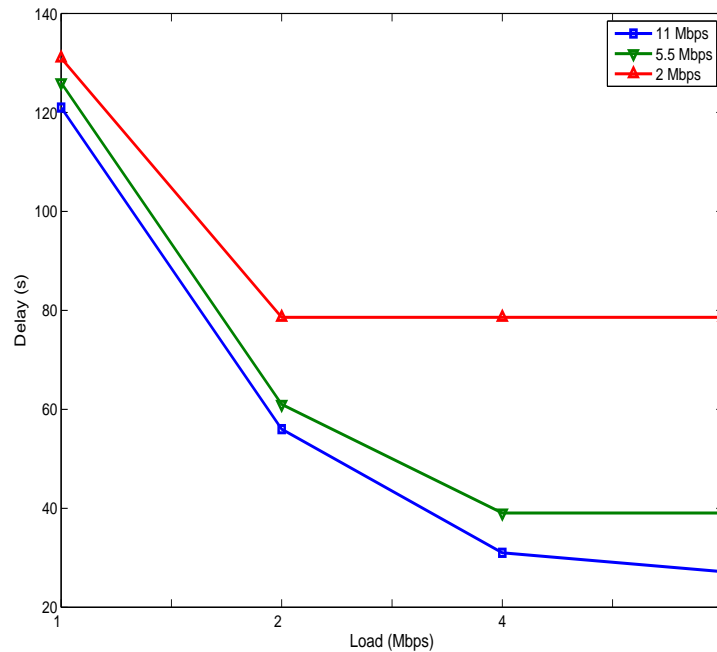


Figure 3.7: Channel Access Delay via Node Load

Figure 3.7 shows the access delay of the node. As the inter-arrival time decreases, load increases, delay to access channel reduces. The highest access delay occurs with the least data rate, 2Mbps.

### 3.4 Burst Length without RTS

RTS control packets effect burst length and packet error rate due to delay and collision. When RTS is used there is a delay due to control packets transmission. As shown in Figure 3.8 burst length decreases with RTS due to this delay. Without RTS much less delay occurs and burst length decreases more slowly because delay is less without RTS than with RTS. In this simulation for both cases, with and without RTS/CTS, the data rate is 11 Mbps.

However without RTS more collisions occur and this increases packet error rate as shown in Figure 3.9.

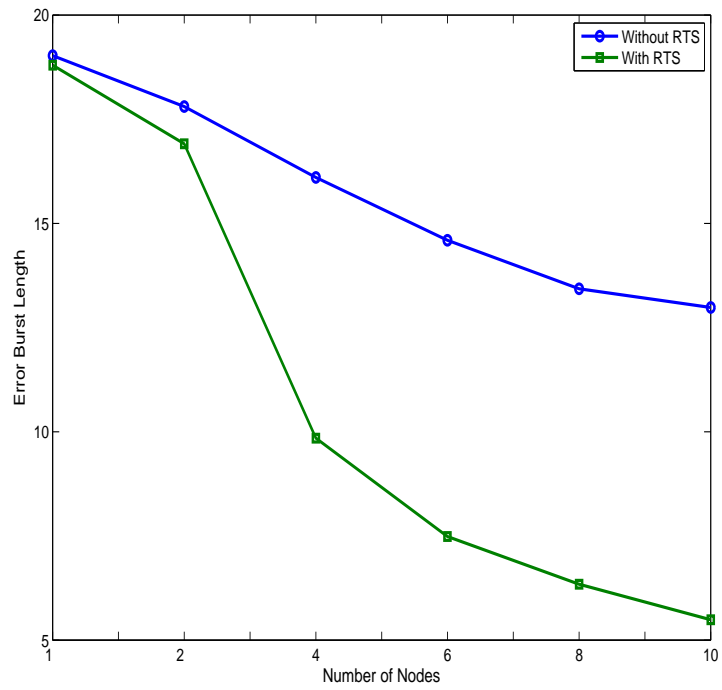


Figure 3.8: Error Burst Length with RTS

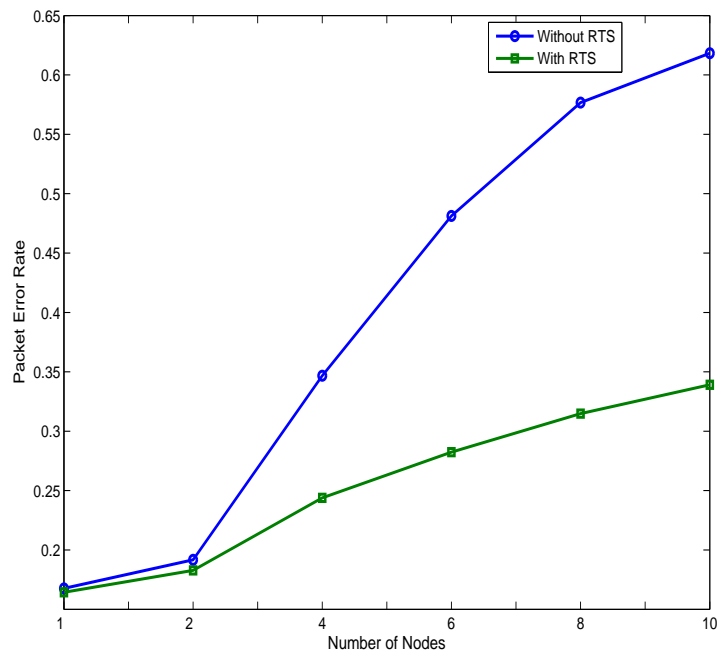


Figure 3.9: Packet Error Rate with RTS

The channel access delay is less with without RTS than with RTS. This is due to an extra delay for RTS/CTS packets transmissions. This delay is shown in Figure 3.10.

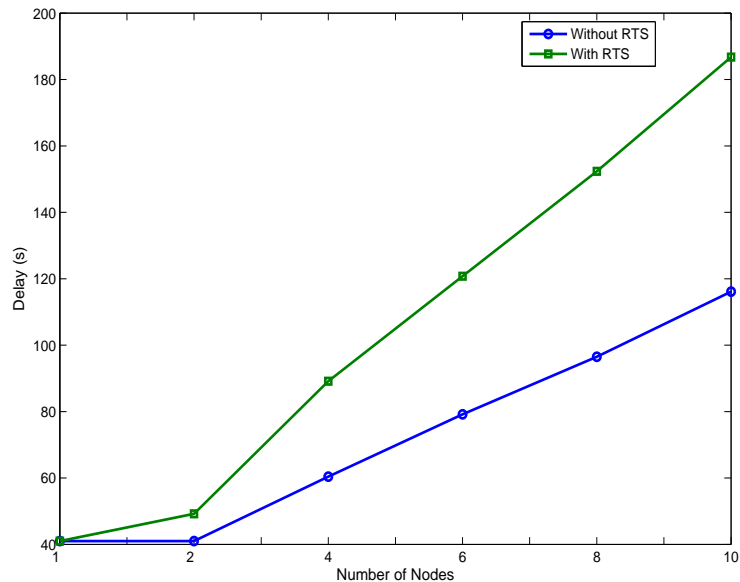


Figure 3.10: Channel Access Delay with RTS

# Chapter 4

## Multihoming with a New MAC Protocol

### 4.1 System Model

#### 4.1.1 Source Model

We consider a system where (STAs) need to transmit  $K$  packets successfully to a destination with CBR application or multimedia application such as VBR or video transmission. When the destination receives  $K$  packets, the source transmits  $N$  packets by splitting among available APs. If  $K$  packets are received successfully by the destination the original information can be recovered. The efficiency of splitting policy is given as  $R = K/N$ .

#### 4.1.2 Packet Loss and Channel Model

When a real time application and recovery of corrupted packets are dealt together retransmission is not a viable option in some real-time application. A delayed packet is not preferred for multimedia applications and therefore we assume that MAC layer retransmission functionality is disabled similar to the previous work e.g [2]. Our MH-MAC protocol replaces the original retransmission functionality for multihomed networks.

In [2], the authors proposed Markov based model at the packet level and byte

level. Although [50] outlines the inadequacy of the two states Markov chain for GSM based networks, [48] and [2] showed that two-state Markov model is adequate for the bursty 802.11b packet loss process. Khayam et al. in [2] performed a real-world experiments with single AP and three STAs and they found the transition probability of two-state Markov chain with Gilbert model and UDP transport protocol. Also they showed that the channels are bursty as shown in table 4.1. UDP protocol is more appropriate for multimedia application since TCP transport protocol causes unnecessary end-to-end retransmission.

Table 4.1: Burst Length and Error Rate of packet level Markov model for 802.11b protocol stack

Bit rate (Mbps)	$\alpha$	$\beta$	Error rate $p$	Burst Length $b$
2	0.3529	0.999	0.0002	1.5454
5.5	0.6836	0.820	0.3614	3.1606
11	0.8966	0.376	0.8576	9.6712

We use the Gilbert model in Figure 4.1 which is a well known model that captures the bursty nature of channels and is simple enough for a computational analysis. In good (bad) state of the Gilbert model probability of correct transmission is  $P_G = 1$  ( $P_B = 0$ ). The packet losses occur in bursts. Transition probabilities between good state and bad state depend on mean burst length and mean packet error rate of channels. The transition probability matrix is defined as follows:

$$P = \begin{bmatrix} \beta & 1 - \beta \\ 1 - \alpha & \alpha \end{bmatrix}$$

The steady state probability of being in state G is:

$$\pi_G = \frac{1 - \alpha}{1 - \alpha + 1 - \beta} \quad (4.1)$$

Transition probabilities  $\alpha$  and  $\beta$  are related to mean loss rate and mean burst length as follows:

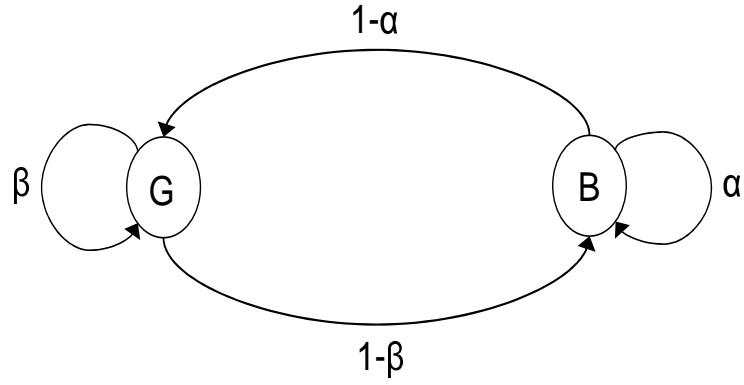


Figure 4.1: Gilbert Model

$$\beta = 1 - \left(\frac{p}{b(1-p)}\right) \quad \alpha = 1 - \frac{1}{b} \quad (4.2)$$

Where  $b$  and  $p$  are mean burst length and mean error rate of packet losses respectively.

## 4.2 Problem Definition

The source should transmit  $K$  packets to the destination successfully. Until  $K$  packets are received, the source needs to send more than  $K$  packets ( $N \geq K$ ) where  $N$  is total number of packets sent by the source until  $K$  packets are received. Assume that there are  $I$  channels available from a specific source to the destination and each packet out of  $N$  packets is transmitted over among  $I$  channel that has the highest success probability.

$$\max[Pr(\text{success transmission for packet } k \text{ among } I \text{ channel})] \quad (4.3)$$

Our objective is to find the best channel in terms of success probability before each packet transmission. By maximizing success probability we also try to increase the efficiency.

In [23], the authors have considered a similar objective. However there is an important difference between our objective and the objective in [23]. The authors in [23] use  $(N, K)$  coding technique to ensure a probability of successful message delivery greater than or equal to  $P_{min}$ . They try to deliver at least  $K$  out of  $N$  packets. This means that the idea in [23] has a probabilistic manner and it does not guarantee that all  $K$  packets are successfully received. On the other hand, according to our policy successful transmission of all  $K$  packets is guaranteed because our policy continues to transmit packet until  $K$  packets are received.

Also the authors in [23] offer a deterministic traffic splitting policy which is not computationally efficient. According to their policy each packet is sent by a predefined schedule. If there are  $C$  channels in the system, there are  $C^N$  policies where  $N$  is the length of the frame to be sent. However in our algorithm there is no a predefined policy. In our policy, each channel is selected before a packet transmission dynamically. Additionally, according to [23] channel diversity yields substantial benefits mostly when channel are approximately equivalent. However in our policy channel can be quite different from each other.

#### 4.2.1 MH-MAC Theory

In this section we propose a new MAC protocol for multihomed users in a bursty wireless channel. Our MAC protocol is based on a dynamic policy where the best channel is selected for each packet based on the outcome of the previous transmission. Multihomed MAC policy can also be considered as an enhancement of the MAC layer retransmission functionality in 802.11 when stations multihome.

The basic idea behind Multihomed MAC (MH-MAC) is as follows: At first station does not know the instantaneous channel condition but it has long-term channel characteristics which can be derived from Gilbert model for each channel  $i$ . Additionally, the station keeps track of the time and outcome of the last transmission over channel  $i$ , for  $i = 1, \dots, I$ . Let  $t_i$  be the last transmission opportunity when the channel  $i$  was used. Also let  $s_{t_i} = \{G, B\}$  be the state or the outcome of the last transmission observed at time  $t_i$ . Then, the station calculates the probability that



the current transmission over channel  $i$  at time  $t$  is successful given the knowledge of the outcome of the last transmission as follows:

$$Pr(\text{success at time } t \text{ on channel } i | s_{t_i}) = p_{s_{t_i}G}^{(t-t_i)} \quad (4.4)$$

Where  $p_{sG}^{(n)}$  is the  $n$  step transition probability from state  $s$  to state  $G$  and it can be calculated from two-state Markov process:

$$\begin{aligned} P^n &= \begin{bmatrix} p_{GG}^{(n)} & p_{GB}^{(n)} \\ p_{BG}^{(n)} & p_{BB}^{(n)} \end{bmatrix} \\ &= \frac{1}{1-\alpha+1-\beta} \begin{bmatrix} 1-\alpha & 1-\beta \\ 1-\alpha & 1-\beta \end{bmatrix} \\ &\quad + \frac{(\alpha+\beta-1)^n}{1-\alpha+1-\beta} \begin{bmatrix} 1-\beta & -(1-\beta) \\ -(1-\alpha) & 1-\alpha \end{bmatrix} \end{aligned}$$

MH-MAC chooses channel  $i^*$  at time  $t$  for which (4.4) is the maximum and updates  $t_i^* = t$ ,  $i^* = i$  and  $s_{t_i}$  according to the outcome of the transmission. The iteration concludes when all  $K$  packets are successfully transmitted.

### 4.2.2 MH-MAC Step by Step

In this section a description of our protocol with an example is given step by step .

#### ***Step 1 : Initialization***

All nodes aim to transmit their  $K$  packets to a destination as in Figure 4.2. In

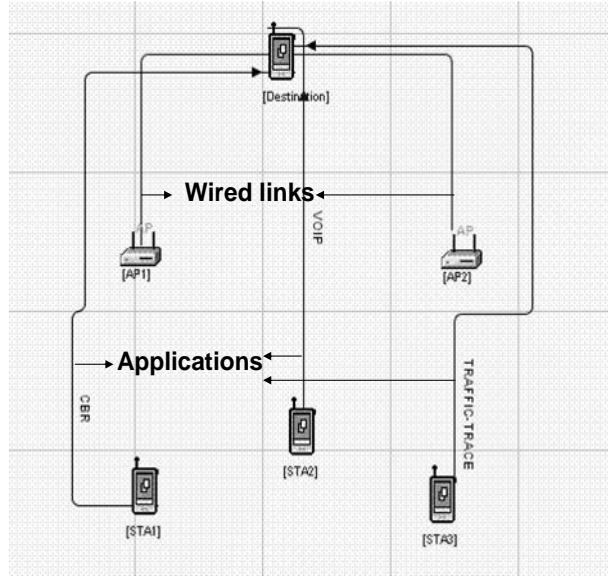


Figure 4.2: An Example Scenario

this example figure, there are three STAs and two APs. STA1, STA2 and STA3 will implement CBR, VoIP and video applications respectively.

Nodes transmit their packets to *destination* over *AP1* and *AP2* that are connected to the *destination* by wired as in Figure 4.2. STAs connect to APs by wireless. We assume that wired links capacities and queues buffer of the nodes are large enough to prevent any losses due to wired links and buffer overflows. This means that all errors occur in wireless channels.

At the beginning of communication, the station should find the other stations or AP in order to communicate the AP or the station in an infrastructure BSS. The process of finding an AP is scanning. There are two kinds of scanning; active and passive scanning. In passive scanning, APs send beacon frames in a particular interval, 200 ms in our work, that are frames that have control information and are transmitted in each channels and help a wireless station to identify nearby wireless AP. Beacon frames also contain AP address. Nodes know the AP address and the channel index that AP uses by using this information. The stations use the knowledge of AP address and the channel index of the APs when switching among the APs.

### *Step 2 : Training Phase*

After all stations receive beacon from the APs then, only one node starts its training phase while other nodes remain silent. In the training phase, the nodes send 5000 training packets to each channel to measure the burst length and error rate of the loss process of each channel according to the application that the node implements. The value of 5000 packets is enough for a good estimation for burst length and packet error rate calculation. We observed that if the number of packets used in training is less than 5000 then estimation of mean burst length and packet error rate will not be well enough. If we use more than 5000 packets for training, training process will start to be costly. Actually the number of training packets depends on the channel coherence time. As the coherence time increases the more training packets should be used to get more information about the channel.

If a node is willing to implement CBR application then it transmits 5000 fixed size CBR packets that the node will send during the actual transmission to each channel or if the node implements video transmission, the node sends 5000 video packets for measuring the long-term channel characteristics because the characteristics of burst length and packet error rate depend on what kind of packets and applications that the nodes use. Training phase duration for a node depends on mainly data rate, user traffic, number of training packets, packet size and usage of RTS/CTS. If a node generates 10000 packets at a rate of 250 packets per second with a size of 1500 bytes then it transmits at data rate of 11 Mbps with RTS/CTS control packets, the duration takes approximately 40 seconds. Characteristics of packets of CBR application and video application are quite different. A node can not send different kinds of packets from actual information packets in the training phase. That is why we send same kind of packets as original packets for training.

To measure the long-term channel statistics 802.11k amendment gives a practical opportunity [53]. WLAN Radio Measurements enable STAs to observe and gather data on radio link performance and on the radio environment. A STA may choose to make measurements locally, request a measurement from another STA, or may be requested by another STA to make one or more measurements and return the results. Radio Measurement data is made available to STA management and upper

protocol layers where it may be used for a range of applications. The request and report measurements are as follows:

- **Beacon**; Enables a STA to request from another STA a list of APs it can receive on a specified channel or channels.
- **Frame**; Returns a picture of all the channel traffic and a count of all the frames received at the measuring STA.
- **Channel Load**; Returns the channel utilization measurement.
- **STA statistics**; Returns groups of values for STA counters and for BSS Average Access Delay. RTS, ACK success count, retry and failed count etc.
- **Link Measurement**; Provides measurements of the RF characteristics of a STA to STA link. This measurement indicates the instantaneous quality of a link.

If training phase finishes for an AP (channel), nodes switch to the other AP (to other channel) by using the AP address that the node gets from beacon frames at the beginning of the communication.

### ***Step 3: Beginning of MH-MAC***

After each node ends up the training phase for each channel, the nodes know the mean burst length and mean error rate of the loss process of the channels. Now, without loss of generality let us assume that the state of channels before first packet transmission is bad state (B). A node transmits its first packet over the channel that has the highest  $p_{BG}^{(1)}$  probability. After the node receives the outcome of the first transmission it follows the step 4 below.

### ***Step 4: Process of MH-MAC***

After the outcome of the first packet transmission is observed, now the node should determine the channel that has the highest probability of successful transmission for second packet transmission. The second packet transmission is a *new*

*transmission opportunity* to transmit. If the first packet was transmitted over channel 1 (to AP1) then  $t_1 = 1$ , ( $p_{B_1G}^{(1)} \geq p_{B_2G}^{(1)}$ ), otherwise  $t_1 = 0$ . If the node transmitted its first packet over channel 2 (to AP2) then  $t_2 = 1$ , otherwise  $t_2 = 0$ . *New transmission opportunity* ( $t$ ) is a slot in which a single packet is transmitted and this slot is determined in the next section. Let us assume that the first transmission occurred over channel 1 and the outcome of the transmission is good (success,  $G$ ). This means that  $t_1 = 1$  and  $t_2 = 0$ . The node now calculates the transition probabilities,  $p_{GG}^{(1)}$  and  $p_{BG}^{(2)}$  for channel 1 and channel 2 respectively.  $n$ -step size of channel 1 is 1 because last transmission was done over that channel ( $t - t_1 = 1$ ,  $t - t_2 = 2$ ) and *new transmission opportunity*,  $t = 2$ , at the second transmission time and we assumed that the outcome of the first packet is  $G$ , and the state of second channel is  $B$ . Every node calculates the transition probability in the transition matrix by using the mean burst length and mean packet error rate measured in the training phase for every packet transmission. However burst length and packet error rate depend on the number of users and user traffic in the network as explained in Section 3. Due to this fact, user needs to update the burst length and the packet error rate of each channel during the all transmissions. This updating process is important. Updating the mean burst length and mean packet error rate provides more realistic and accurate results of transition probability in Markov chain because long-term channel statistics change with time due to the network conditions. However updating brings costs. These costs are computational complexity and delay. If the updating occurs frequently, for example every 10 packet, this provides better results but also brings more cost. With these information we update the mean burst length and mean packet error rate after every 100 packets transmission.

When a new user attempts to associate with an AP, the user should know the channel statistics. The new user goes to *Step 2* and during its training phase the other users remain silence.

### ***Step 5: Switching AP***

Switching to a new AP that is different from the current AP only occurs when the transition probability from state  $s$  to state  $G$ ,  $p_{sG}^{(n)}$ , is higher for the new AP

than for the current AP. This means that the new channel is better. For example, let us assume that in *Step 3*  $p_{BG}^{(2)}$  is higher than  $p_{GG}^{(1)}$ , then the node decides to switch to AP2 (to channel 2). In this process firstly the node deassociates/deauthenticates with the current AP (AP1) by sending deauthentication and disassociation frames to the current AP and associates/authenticates with the new AP (AP2) by sending reassociation and reauthentication frames to the new AP. Then the second packet is transmitted over channel 2. If  $p_{GG}^{(1)}$  is higher than  $p_{BG}^{(2)}$  the node stays in the same channel and with the same AP (AP1). While switching to a new AP, there will be delay due to exchange of association and authentication frames. This delay reduces the correlation of channel. The problem is explained in the next section. We assume a 25 ms fixed delay when a node changes AP [52]. In [52] the authors show that stations can connect to multiple accessible wireless networks by using a single wireless card and the switching delay between APs is on the order of 25 ms.

802.11r amendment provides a fast and secure hand-off for switching APs [54]. This amendment specifies fast Basic Service Set (BSS) transitions between access points. Under the amendment procedure minimal connectivity loss of under 50 ms and application disruption can be achieved. In current standard, the Authentication Server and the mobile station would have agreed upon one set of Pairwise Master Keys (PMK) to provide authentication. This PMK is then distributed to the AP. Both the STA and AP would use this PMK to mutually authenticate each other. The aim of the amendment is to ensure that most of the authentication processes are performed before the station actually begins switching. Based on the current draft, the recommended method to generate the PMK (IEEE 802.1X) will be done once the station joins the network. The PMK is sent to all the APs that are authenticated in the subnet. Hence, when a station roams across APs, the PMK is assumed to be present.

### ***Step 6: Final Process***

Finally, all nodes that use MH-MAC in the network follow the same procedures above. If a node is successful to transmit all  $K$  packets to the *destination* the transmission is successfully done otherwise, the nodes goes to *Step 4* and the process

keeps on until the *destination* receives  $K$  packets from the node.

### 4.2.3 Pseudocode for MH-MAC

In this section, to understand the MH-MAC algorithm a pseudocode is given below.

```

for  $i = 1$  to  $I$  do
  while  $k \leq 5000$  do
    Send training packet  $k$  to  $AP_i$ 
    if Outcome of packet  $k=B$  then
      array[k]=1
    else
      array[k]=0
    end if
     $k \leftarrow k + 1$ 
  end while
  Calculate mean burst length and mean packet error rate
  while  $k \leq 5000$  do
    if array[k]=1 and array[k+1]=1 then
      A bursty case occurred
       $NumberofOccurofBurstCase ++$ 
      for  $i = k$  to 5000 do
        if array[i]=0 then
          Stop. Burst Case Finished
           $lengthofBurstyCase = lengthofBurstyCase + (i - k + 1)$ 
        end if
      end for
       $k \leftarrow i$ 
    end if
    if array[k-1]=0 and array[k]=1 and array[k+1]=0 then
      No bursty Error
       $SingleError = SingleError + 1$ 

```

```

    end if
end while
Mean Burst Length = lengthofBurstyCase/NumberofOccurofBurstCase
Mean Packet Error Rate =
(SingleError + lengthofBurstyCase)/5000
if Training finishes for channel i then
     $i \leftarrow i + 1$ 
end if
end for
print Training Phase finished

while number of received packets  $\geq K$  do
    New Transmission opportunity  $\leftarrow$  New Transmission opportunity + 1
    print Update Mean Burst Length and Mean Packet Error Rate for every
    AP after every 100 packets transmissions
    if New Transmission opportunity = 100 then
        Updating Starts
        Go to Training phase to calculate Burst Length and Packet Error rate
    end if
    if New Transmission opportunity ( $t$ ) = 1 then
        for  $i = 1$  to  $I$  do
            Calculates  $P_{BG}^1$  for each  $AP_i$  and find max [  $P_{B_iG}^1$  ]
        end for
        Transmit first packet to  $AP_{i_{max}}$  that has the max [  $P_{B_iG}^1$  ]
    end if
    for  $i = 1$  to  $I$  do
        Calculate  $P_{s_iG}^{t-t_i}$ 
        if  $P_{s_iG}^n = \max[P_{s_iG}^n]$  then
             $NextChannel \leftarrow i$ 
             $t_i \leftarrow$  New Transmission opportunity
        end if
    end for
end for

```



```

Transmit packet to  $AP_i$ 
if outcome=B then
     $s_{t_i} = B$ 
else
     $s_{t_i} = G$ 
    number of received packet  $\leftarrow$  number of received packet + 1
end if
end while

```

### 4.3 MH-MAC for Multi-User Case

The theory behind MH-MAC protocol should be modified when we consider a network in which many users transmit any kind of application such as multimedia or CBR. In this case basically delay and collision problems occur. In this section, we try to understand the effects of these problems on our protocol and then we find solutions for these problems.

#### 4.3.1 Definition of Delay Problem

Basically delay problem occurs in multi-user scenario due to three main reasons.

##### *1. Delay due to Switching*

Switching delay occurs because of changing access point. Recall Figure 4.2 and it is shown to understand switching delay problem clearly. In this figure there are two APs and three station (STAs). This kind of delay occurs when a STA decides that the probability of being in good state for next channel is greater than the probability of being in good state for the current channel. For example, at time  $t = 1$  STA1 transmitted a packet to AP1 based on  $p_{sG}^{(n)}$  and after STA observed the outcome of the transmission it observed that probability of being in good state for AP2 is greater than probability of being in good state for AP1 at next packet

transmission time,  $t = 2$ . Then STA1 decides to switch to AP2 to transmit the next packet at time  $t = 2$ . After switching delay, channel 2 will not be the channel observed right after time  $t = 1$  if we assume that coherence time of the channel is less than switching delay and the second channel will change. As indicated in [55] the coherence time changes with mobile speed and at mobile speeds of 1 m/s the coherence time is approximately 122.88 ms for a center frequency of 2.4 GHz. This means that as mobile speed increases the coherence time will become smaller and switching delay ( 25 ms) will be larger than the coherence time and decision probabilities will not be realistic anymore.

It is obvious that if a station switches among channels frequently this brings more delay and it causes the station makes false decision in terms of transition probabilities. In theory of MH-MAC, we assume that next packet is transmitted after the outcome of first packet is observed. However with delay the next packet will not transmitted right after the first packet transmission finishes. The next packet transmission time will expire and increase with delay. In calculation of the transition probabilities,  $n$ -step size should be modified because there will not just only one slot between the next packet transmission time ( $t$ ) and the last packet transmission time ( $t_i$ ) in  $n = t - t_i$ . We calculate delay in terms of slots when giving solution to the delay problem.

### ***2. Delay due to control packets***

The second reason that causes delay is the transmission of control packets. As known, when control packets namely RTS, CTS and ACK are used, delay will occur in order to transmit the control packets. This kind of delay also fails probability decision as above. However, this delay can be neglected if the channels are slow fading or if the channel coherence times do not change during a single packet transmission. When stations are not mobile the coherence time is high and this means that the delay due to the transmission of control packets does not effect probability decision.

### ***3. Delay due to other transmission***

The IEEE 802.11 MAC layer is responsible for a structured channel access

scheme and is implemented using a Distributed Coordination Function based on the (CSMA/CA) protocol. The CSMA/CA based MAC protocol of IEEE 802.11 is designed to reduce the collisions due to multiple source transmitting simultaneously on a shared channel. In a network employing the CSMA/CA MAC protocol, each node with a packet to transmit first senses the channel to ascertain whether it is in use. If the channel is sensed to be idle for an interval greater than the (DIFS), the node proceeds with its transmission. If the channel is sensed as busy, the node defers transmission till the end of the ongoing transmission. This delay can be estimated according to [56]. In [56], the authors propose a analytical model to estimate the channel access delay.

### 4.3.2 Solution to Delay Problem

We solved the delay problem as follows: If consecutive two packets are transmitted within a coherence time, these two packet are affected almost equally by the channel. If a delay mentioned previous section occurs then the next packet will not be within the same coherence time like the first packet and the next packet will be affected distinctly. We calculated channel coherence time in terms of packet transmission time and a packet transmission time (PT) is defined as one slot time. Figure 4.3 shows the coherence time and packet time with slot.

In this example, consider that coherence time (CT) is 5 slots. Our MAC protocol keeps track of the transmission time of each packet and the delay that is the difference between the next packet transmission time and the previous packet transmission time. Formulary: Assume that a station transmitted a packet to the channel at time  $t1$ . Then the station will transmit the next packet to the channel at time  $t2$ . Delay (D) is:

$$D = t2 - t1 \tag{4.5}$$

In our motivation example, delay plus PT shown in Figure 4.3 is 7 slots. Delay may be any kind of delay explained. This means that next packet will be transmitted 7 slots later after the outcome of the previous packet appears. If the delay is greater

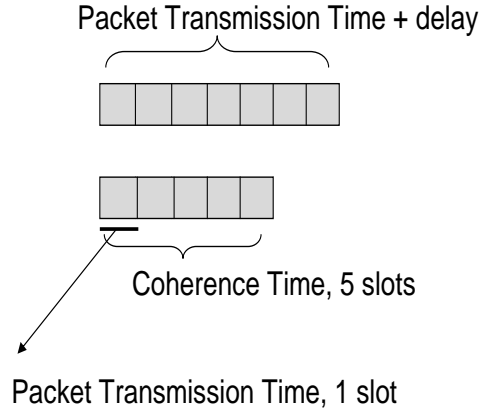


Figure 4.3: Coherence Time and Packet Transmission Time

than the coherence time the next packet step size  $n$  is calculated according to this delay. For example, coherence time is 5 slots and the delay is 7 slots, in this case the step size  $n$  is equal to  $\lfloor \frac{7}{5} \rfloor = 1$ . ( $\lfloor(x)\rfloor$  function rounds  $x$  to the nearest integer less than  $x$ ). This means that the next packet will not be within the same coherence time as first packet and it will be affected in a different way from the previous packet by the channel. A general formula for step size  $n$  is:

$$n = \lfloor \frac{D}{CT} \rfloor \quad (4.6)$$

### 4.3.3 Collision Problem

The second problem occurring in multi-user case is collision. Consider the example shown in Figure 4.2. If two or more STAs decide to send their packets to the same AP at the same time, there will be a collision. Assume that STA1 and STA2 decide to send their packets to AP1 at the same transmission time. After transmitting the packets, the packets will drop due to collision. Both of STAs will observe that the channel is in bad state at the transmission time after the outcome of collided packets is observed and this may be a improper observation. They may stay in same AP or change their AP (channel) according to their transition probabilities,  $p_{sG}^{(n)}$ . Let us assume that both stations transmit their packets to AP1 at different times and there is no collision and both transmissions are successful. When the stations transmit their next packets the stations calculates the transition probability from state  $G$

to state  $G$ ,  $p_{GG}^{(n)}$  because the previous transmission is successful for both stations. If there is a collision the stations will transmit their next packets according to the transition probability from state  $B$  to state  $G$ ,  $p_{BG}^{(n)}$  because the previous transmission is unsuccessful due to the collision.

Shortly, collision makes STAs incorrect decision in terms of transition probabilities. Collision also effects the long-term channel characteristic such as burst length and packet error rate.

#### 4.3.4 Solution to Collision Problem

To handle with collision problem we employ two approaches.

**1-)** First approach is to update the channel statistics. We need to update channel statistics because network conditions change in terms of active nodes and user traffic. If the number of active nodes and user traffic increase or decrease long-term channel characteristics will be affected by this situation as shown in Chapter 3. By updating mean packet error rate and mean burst length of the channels, we ensure to stay in the correct steady state. Errors due to collision will be a part of long-term channel statistics.

**2-)** In second approach collision problem can be assumed as a delay problem explained in 4.3.1. Consider that STA1 and STA2 in Figure 4.2 transmit their packets and the stations can know if a collision occurred or not. This means that stations use a collision detection algorithm. With this information, the station caused collision keeps their state and their last transmission opportunity,  $t_i$ , same as before collided packet transmission. For example, STA1 transmitted  $k^{th}$  packet to AP1 at time  $t = 1$  and outcome of the transmission was good the station knows that a collision did not occur. Assume that when the transmission of  $(k + 1)^{th}$  packet at time  $t = 2$ , there is a collision due to the transmission of STA2 at the same time.  $(k+1)^{th}$  packet is collided and it will drop. Both stations know that there is collision. When the transmission of  $(k + 2)^{th}$  packet of STA1, STA1 keeps the outcome of last transmission as state  $G$  because the outcome of last transmission (transmission of packet  $k^{th}$ ) before collision is good and STA1 keeps the last transmission opportunity

as  $t = 1$ . Transition probability for  $(k + 2)^{th}$  packet is  $p_{GG}^{(t-1)}$  instead of  $p_{BG}^{(t-2)}$  where  $t$  is the transmission time of  $(k + 2)^{th}$  packet.

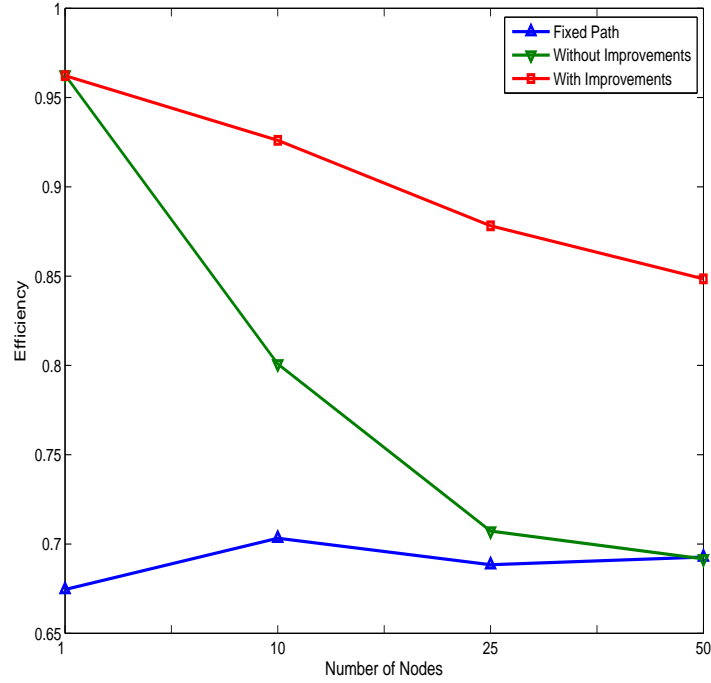


Figure 4.4: Efficiency with and without Improvements

As seen in Figure 4.4, after the improvements are performed a significant improvement appears in the efficiency of a node. Delay is the least with MH-MAC in which the improvements are performed. Delay without the improvements MH-MAC is higher than the delay with fixed path scheme as shown in Figure 4.5. This is due to the inaccurate transition decision in terms of transition probabilities lead by collision. In this simulation all nodes have 10000 ( $K$ ) CBR packets. RTS/CTS mechanism is enabled, retransmission is disabled and data rate is 11 Mbps.

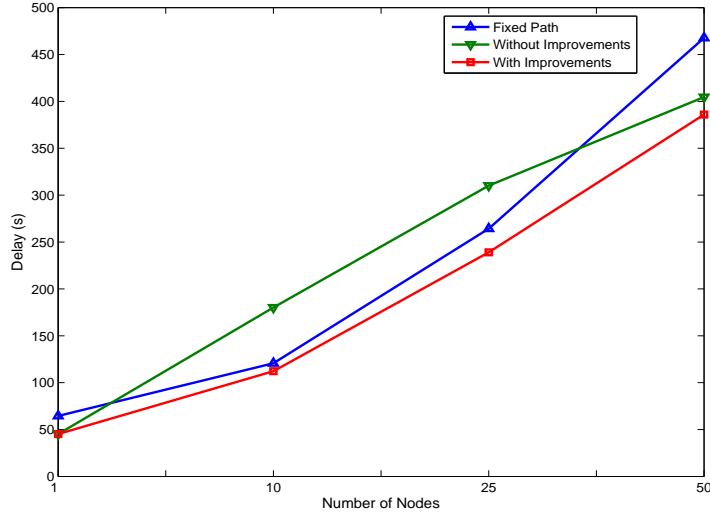


Figure 4.5: Delay with and without Improvements

## 4.4 Extension to Different Transmission Rates

We assume in previous section that all channels have the same transmission rate. In this section we consider the case that channels transmit the packets with different PHY rate. By selecting the channel which is most likely to have a successful transmission the efficiency can be improved. In addition, when the available channel differs in transmission rates, we can modify MH-MAC to select the *most likely to be successful and fastest channel* in each transmission slot. Formally, we select the channel,  $i$ , at time  $t$  as follows:

$$\max[p_{st_i}^{(t-t_i)} r_i] \quad (4.7)$$

### 4.4.1 Rate Adaptation in MH-MAC

802.11b and other standards support to use different transmission rate during a session. Transmission rates are selected based on the signal strength indicated. This algorithm is called as auto rate fallback. The algorithm selects a higher rate if the signal strength is good and cuts down the rate as the the signal strength decays. In this thesis we do not use any kind of auto rate fallback algorithm. Our rate adaptive algorithm bases on transition probabilities that determine the most

likely to be successful and fastest channel. To adapt the rate adaptation all stations need to know the mean burst length and mean packet error rate of channels for each data rate, 2 Mbps, 5.5 Mbps and 11 Mbps. Due to this fact, in the training phase, a station sends 5000 training packets at every data rate to measure the long-term channel characteristics at different rates. After training phase, the algorithm selects the best channel as follows:

Recall Figure 4.2. Let  $s_{t_1 2}, s_{t_1 5.5}$  and  $s_{t_1 11}$  be the channel states at 2 Mbps, 5.5 Mbps and 11 Mbps respectively and let  $s_{t_2 2}, s_{t_2 5.5}$  and  $s_{t_2 11}$  be the channel states for second channel. The algorithm finds the channel and rate according to below formula.

$$\max[p_{s_{t_i} r_i}^{(t-t_i)}] \quad (4.8)$$

Assume that STA1 knows burst length and error rate for channel 1 and channel 2 for each transmission rate. Then, assume that for the first packet transmission, at time  $t = 1$ , without loss of generality channel 1 has the highest success probability at 11 Mbps. This means that  $p_{s_{t_1 11} G}^{(1)}$  is maximum among the other probabilities. Then, STA1 associates to AP1 and change its transmission rate to 11 Mbps. After the packet transmission, assume that the outcome of the transmission is bad (unsuccessful), then at time  $t = 2$ , the algorithm will try to find the highest transition probability,  $p_{s_{t_i} r_i}^{(t-t_i)}$ . If the highest probability occurs with channel 2 at 5.5 Mbps ( $p_{s_{t_1 5.5} G}^{(t-t_1)}$  is maximum) then the station associates to AP2 and changes its transmission rate from 11 Mbps to 5.5 Mbps. This iteration concludes until all  $K$  packets are received successfully.



# Chapter 5

## Performance Analysis

### 5.1 Simulation Model

In this chapter, we present the simulation results to evaluate the performance of our new MAC protocol. The proposed MAC protocol is compared with fixed path scheme where a station associates with only one AP during all packets transmission. Our performance metrics are efficiency and delay described below. Important parameters of the simulation are as follows:

- $N$ : Number of packets transmitted from the source node.
- $K$ : Number of packets should be received by the destination node.
- $\eta = \frac{K}{N}$ : Efficiency.
- $T_D$ : Delay between the time at which the first packet received and the time at which the last packet received.
- $T_C$ : Coherence time of a channel.

Our MAC protocol aims to transmit minimum number of redundant packets so we compare our protocol in terms of efficiency. Each node counts the successfully received ACK frames for a data transmission. If the number of received ACK frames equals to  $K$  the node stops to transmit packets. Otherwise, it continues to send packets until  $K$  packets are received. The other performance metric is delay. Delay

is an important metric that shows quality of transmission especially for multimedia applications. Data applications can tolerance delay but a delayed packet is not preferred for multimedia application. All applications in our simulations, video, voIP, VBR and CBR applications, use UDP. TCP is the standard transport-layer protocol in the Internet. However, TCP is generally considered to be inappropriate for delay-sensitive applications such as multimedia.

### 5.1.1 Scenario 1

At first, we test MH-MAC in a scenario that contains two APs that operates on 11 Mbps transmission rate and 8 transmitting nodes. Each node needs to send 20000 packets to a destination over APs. This means that  $K$  value is same for each node and it equals to 20000. All nodes use Constant Bit Rate application (CBR). In CBR, inter-arrival time of incoming packets is fixed and it does not change during the packet transmission. In this simulation node traffic changes between 1 Mbps and 3 Mbps. Inter-arrival time is different for each node. Minimum distance between a station and an AP is 30 meters and maximum distance is 120 meters.

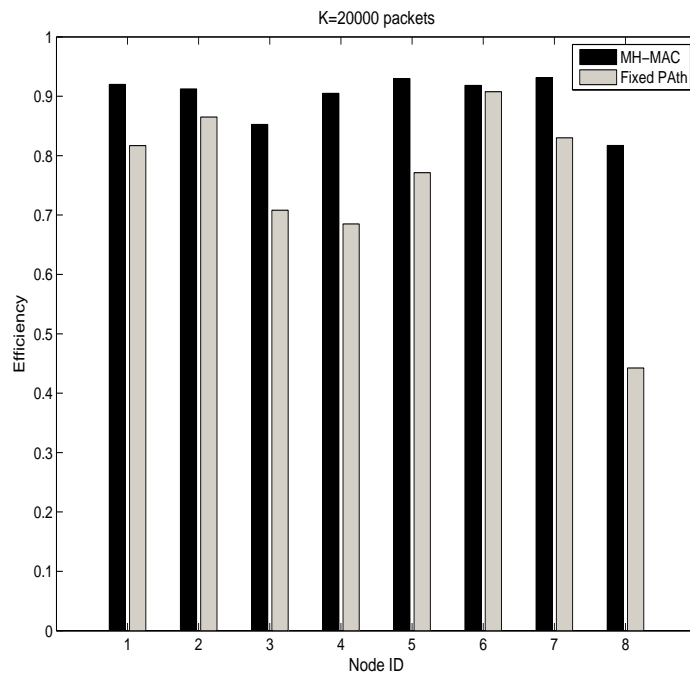


Figure 5.1: Comparison of MH-MAC with Fixed path via Efficiency

Figure 5.1 shows the results for a simple network topology. It can be observed from this result that MH-MAC outperforms fixed path scheme in terms of efficiency. Efficiency is higher with MH-MAC because MH-MAC transmits less redundant packets. On the other hand fixed path scheme needs more retransmissions to receive  $K$  packets. Delay with fixed path is higher than with MH-MAC as shown in Figure 5.2. This is also basically due to more retransmissions that occur in fixed path scheme.

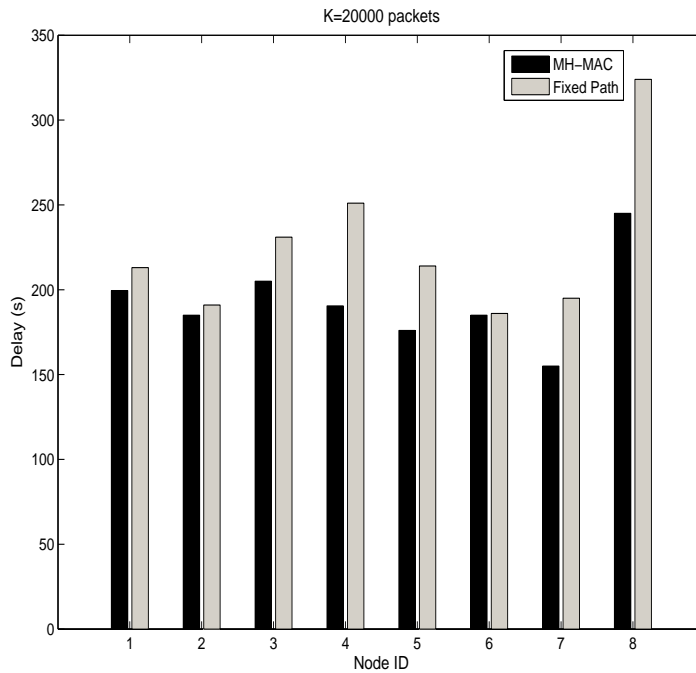


Figure 5.2: Comparison of MH-MAC with Fixed path via Delay

### 5.1.2 Scenario 2

In this case we expand our network. The scenario contains three APs and 50 transmitting nodes in a 300x300 meters area. Each AP uses an orthogonal channel. Three orthogonal channels of 802.11b are used. Frequency of these channels are 2.412 GHz, 2.437 GHz and 2.462 GHz. All APs operate on 11 Mbps PHY rate. Each node uses a CBR application with a different inter-arrival time and packet size and  $K$  value is different for each node and it differs between 2000 packets and

8000 packets randomly. Simulation results are averaged over three runs. Figure 5.3 shows an example scenario.

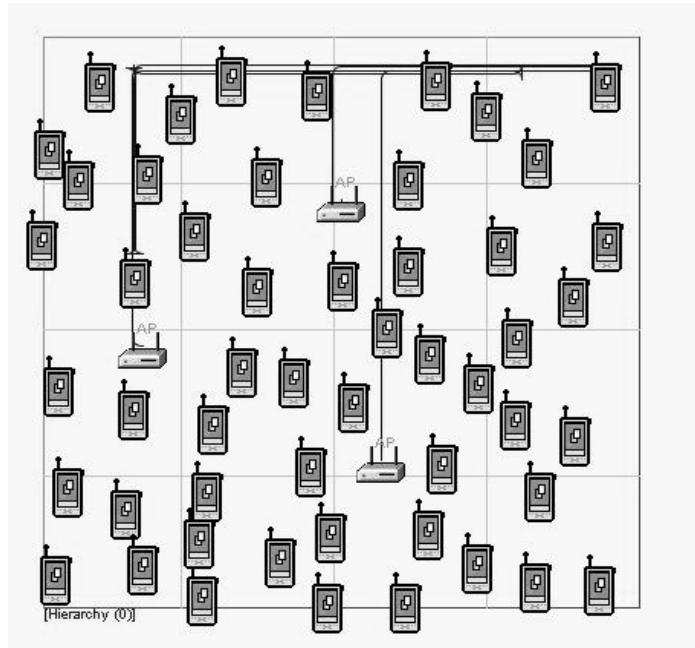


Figure 5.3: Scenario with 50 Nodes and Three APs

The result is shown in Figure 5.4 and Figure 5.5. In these figures, the results are shown for only ten of 50 nodes. Mean burst length and mean packet error rate of each channel for these ten nodes are shown in Table 5.1. In this table  $p_1$ ,  $p_2$ ,  $p_3$  refer to packet error rate of channel 1, channel 2 and channel 3 respectively and  $b_1$ ,  $b_2$ ,  $b_3$  refer to burst length of channel 1, channel 2 and channel 3 respectively. As seen in the figures, MH-MAC has better results for a large network. Efficiency improvement is different for each node. For some nodes with MH-MAC has much better performance. These are the nodes that is farthestmost nodes to the APs. If a node is close to an AP, it uses mainly that AP during all packet course and efficiency results for MH-MAC and fixed path gets similar for both proposals. Figure 5.5 shows the delay results for the same simulation. As similar to scenario 1, more retransmissions occur in fixed path because in a bursty period of packet loss a node that uses fixed path cannot avoid bursty error and this results in more unsuccessful packet transmission.

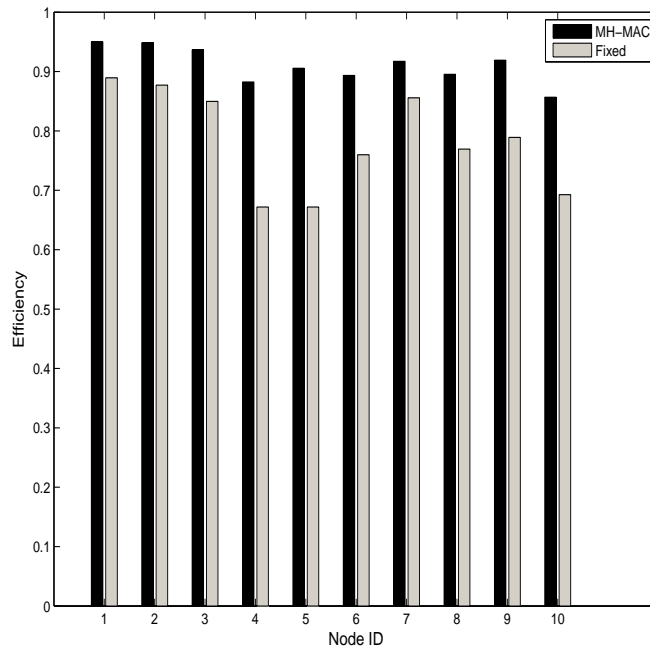


Figure 5.4: Efficiency with 50 Nodes

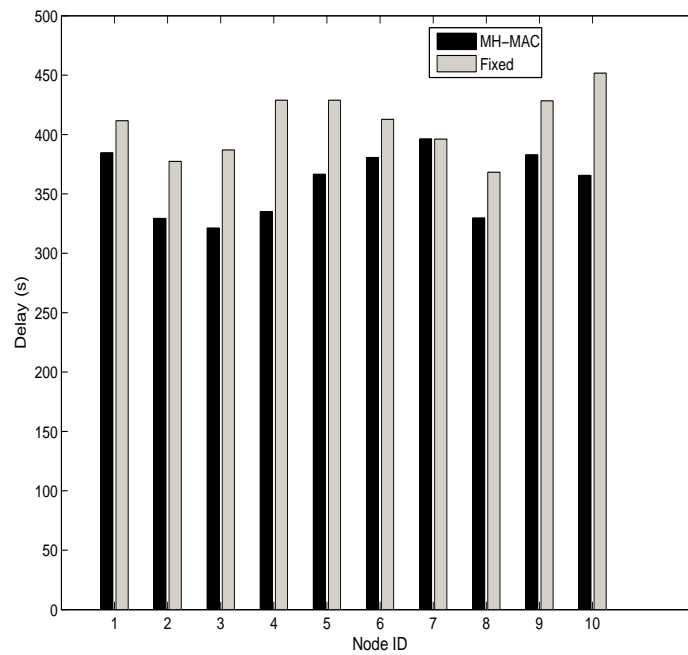


Figure 5.5: Delay with 50 Nodes

Table 5.1: Burst Length and Packet Error Rates of Channels

Node ID	$p_1$	$b_1$	$p_2$	$b_2$	$p_3$	$b_3$
1	0.25	27.32	0.02	21.37	0.37	42.08
2	0.11	25.00	0.08	27.11	0.34	39.14
3	0.32	28.85	0.10	30.58	0.04	29.70
4	0.67	47.72	0.47	36.31	0.30	38.44
5	0.59	41.54	0.14	23.85	0.45	37.83
6	0.57	47.51	0.15	35.63	0.62	45.33
7	0.58	43.20	0.44	45.60	0.13	26.05
8	0.50	44.87	0.20	25.68	0.12	37.70
9	0.30	37.51	0.63	38.42	0.19	24.00
10	0.60	45.09	0.25	38.71	0.22	23.21

## 5.2 Multimedia Scenario

In this scenario, we simulate our protocol for a general network. In this network, nodes implement CBR, variable bit rate (VBR) applications, video and voice over IP (VoIP) transmissions. Distribution of inter-arrival time of VBR have uniform and exponential distribution for this simulation with different mean. VoIP generates a voice traffic that can be modeled with a simple two-state Markov chain. The chain consists of an ON state of exponentially distributed duration with an average of 500 ms and an OFF state of 500 ms. This means that nodes transmit a voice packet with a 0.5 probability. In addition, the voice application uses PCM encoding which generates voice that has a packet size between 160 bytes and 200 bytes. Video traffic is a soccer game that is ten-minute MPG video. The video traffic is characterized with an average rate of about 0.64 Mbps and a peak rate of a little more than 4 Mbps. Network conditions are same as in Scenario 2.

Figure 5.6 and Figure 5.7 show the results of a node that uses the video traffic in the multimedia scenario. The results are averaged over time. Each sample in the Figures presents a session in which  $K$  packets are transmitted. The node selects a  $K$  value between 2000 and 8000 packets. After a session finishes, it selects a random  $K$  value again. Each sample has been run three times and the averaged results are shown.

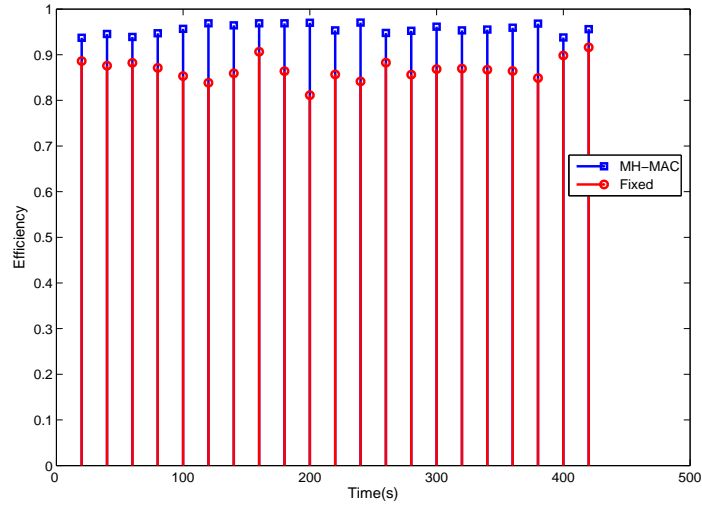


Figure 5.6: Efficiency Comparison of MH-MAC with Fixed path in Multimedia Scenario

Figure 5.6 shows that MH-MAC outperforms fixed path scheme for also multimedia application. Efficiency performance is significantly high for MH-MAC protocol. Figure 5.7 shows that delay performance is also better with our protocol. This is due to the fact that our protocol can switch between APs according to the transition probabilities determined in Section 4.

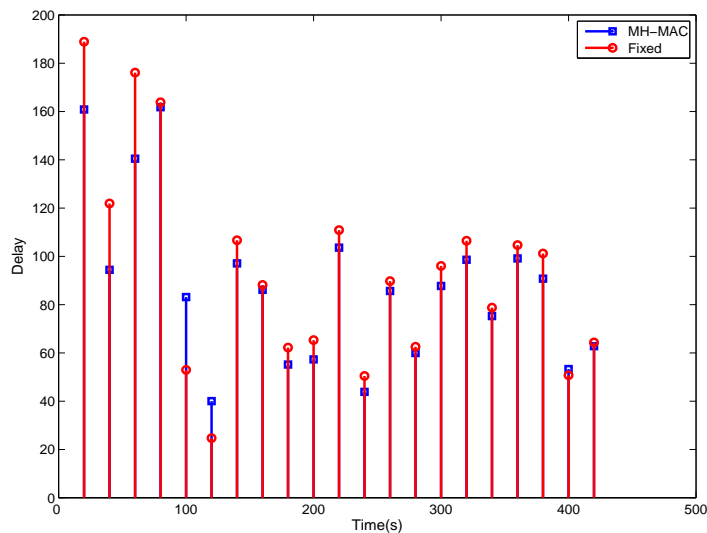


Figure 5.7: Delay Comparison of MH-MAC with Fixed path in Multimedia Scenario

Figure 5.8 shows general results. In the results network efficiency and network delay are considered. Network efficiency means that average efficiency determined by all nodes transmitting until a particular time (50 seconds). Network delay is the delay that is experienced by all nodes transmitting until a particular time (50 seconds).

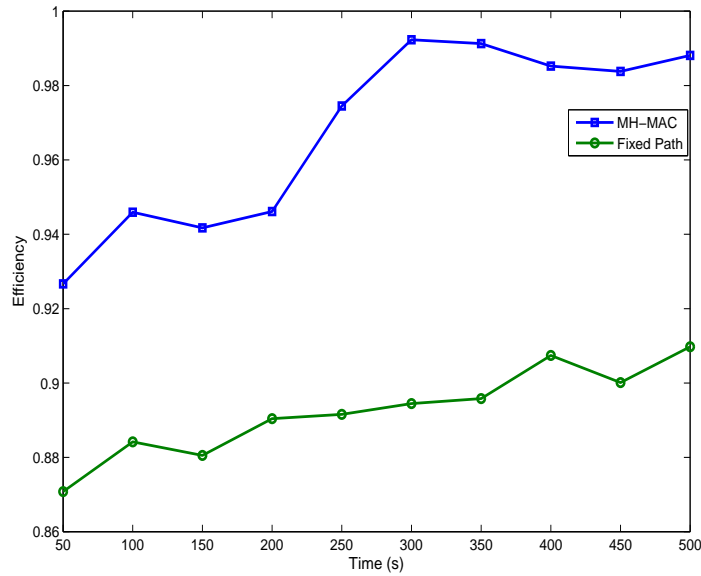


Figure 5.8: Network Efficiency

The network efficiency is averaged over every 50 second of simulation time. As seen in Figure 5.8, network efficiency starts to increase after nearly 250 seconds. This is due to the fact that after that time, all CBR applications finish and there are only the nodes that transmit multimedia applications in the network. This means that the number of nodes transmitting in the system starts to decrease as the time passes. This increases the channel correlation and our protocol starts to give better results. On the other hand network efficiency also increases with fixed path scheme. This is due to the fact that less collisions occur and packet error rate starts to decrease as the number of users is decreasing. Network delay with MH-MAC is less than with fixed path scheme as in Figure 5.9. At 250 seconds all CBR applications finish and delay decays for both proposals. However almost every session, our protocol outperforms fixed path in terms of network delay.



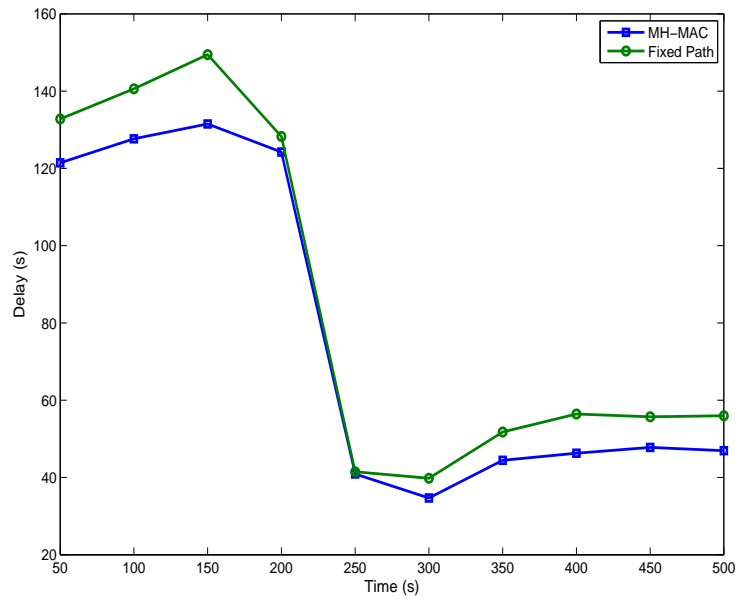


Figure 5.9: Network Delay

To see the results in detail, we run a different experiment. In this experiment only one node uses MH-MAC protocol and the other nodes use fixed path scheme. We compare the result with the scenario where the node uses fixed path and other nodes use MH-MAC protocol.

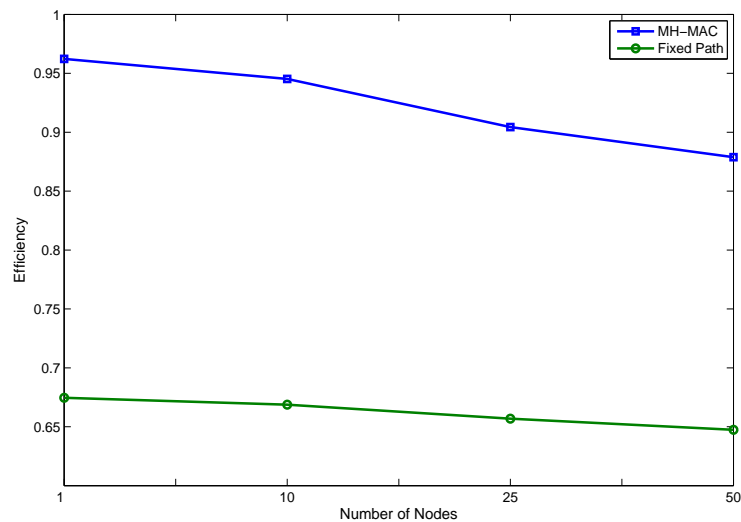


Figure 5.10: One Node Uses MH-MAC

Efficiency of an arbitrary node decreases with MH-MAC when the number of nodes increases. This is due to decreasing in channel correlation and collision. When the node uses fixed path, efficiency decreases also because more collision occurs as the number of nodes increases. However MH-MAC outperforms with fixed path scheme in terms of efficiency and delay as shown in Figure 5.10 and Figure 5.11 respectively.

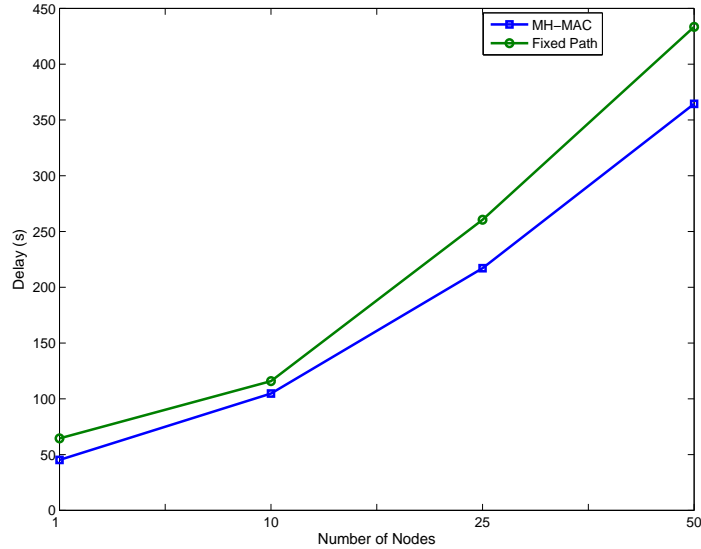


Figure 5.11: One Node Uses MH-MAC

### 5.3 Throughput of MH-MAC

In this experiments we compare both proposals in terms of aggregated throughput. The scenario for this simulation is same as scenario for multimedia.

As number of nodes increases total throughput for both proposal also increases as expected. Saturated throughput occurs approximately with 10 transmitting nodes. At this point the aggregated throughput with MH-MAC is 13.5 Mbps and it is nearly 12.22 Mbps with fixed path scheme. The main reason is that efficiency is higher with MH-MAC and the delay with MH-MAC is less than with fixed path proposal.

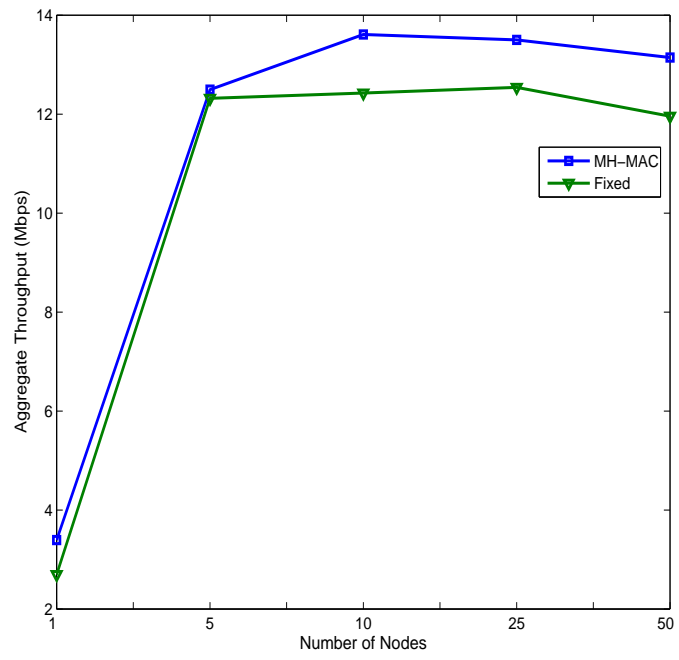


Figure 5.12: Throughput of MH-MAC

# Chapter 6

## Simulation Environment

### 6.1 Network Model in Qualnet

QUALNET [57] is a network simulation software, providing a comprehensive development environment for the specification, simulation and performance analysis of wired and wireless networks. In this chapter an overview of wireless simulation with Qualnet is given and the models created to simulate our work are explained.

QualNet Developer is ultra high-fidelity network evaluation software that predicts wireless, wired and mixed-platform network and networking device performance. This developer provide us to specify models in details, create any kind of network scenarios and execute the simulation and analyze the results.

QualNet is a discrete-event simulator. In discrete-event simulation, a system is modeled as it evolves over time by a representation in which the system state changes instantaneously when an event occurs, where an event is defined as an instantaneous occurrence that causes the system to change its state or to perform a specific action. Examples of events are: arrival of a packet, a periodic alarm informing a routing protocol to send out routing update to neighbors, etc. Examples of actions to take when an event occurs are: sending a packet to an adjacent layer, updating state variables, starting or restarting a timer, etc.

Each protocol operates at one of the layers of the stack. Protocols in QualNet essentially operate as a finite state machine. The occurrence of an event corresponds to a transition in the finite state machine. The interface between the layers is also

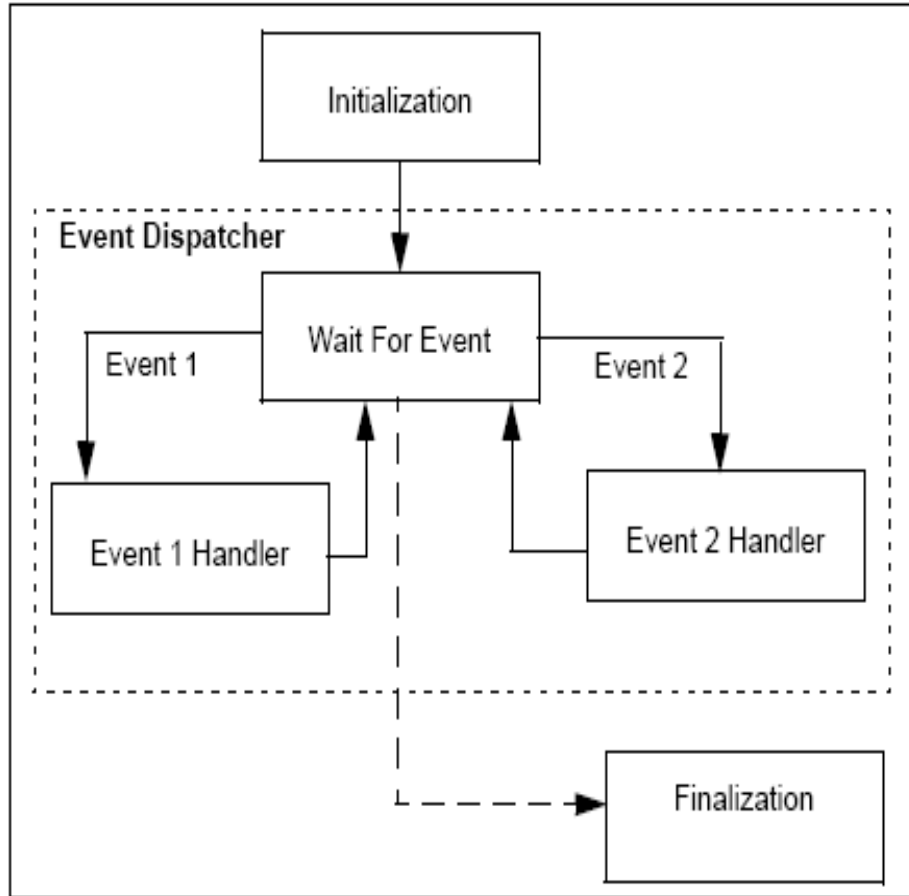


Figure 6.1: Qualnet Modeler

event based. Each protocol can either create events that make it change its own state (or perform some event handling), or create events that are processed by another protocol. To pass data to, or request a service from, an adjacent layer, a protocol creates an event for that layer.

Figure 6.1 shows the finite state machine representation of a protocol in QualNet. At the heart of a protocol model is an Event Dispatcher, which consists of a Wait For Event state and one or more Event Handler states (see Figure 6.1). In the Wait For Event state, the protocol waits for an event to occur. When an event for the protocol occurs, the protocol transitions to the Event Handler state corresponding to that event (e.g., when Event 1 occurs, the protocol transitions to the Event 1 Handler state). In this Event Handler state, the protocol performs the actions corresponding to the event, and then returns to the Wait For Event state. Actions

performed in the Event Handler state may include updating the protocol state, or scheduling other events, or both.

## 6.2 Communication Channel Model

The wireless communication channel in Qualnet is modeled by antenna gains, propagation and transmission delay, signal-to-noise ratio, noise and interference. The transmit power of station is 15 dBm (31 mW). To create a more realistic path loss model, we use wall effects as in [58]. Multi wall (MW) is expressed as free space added with MW effect,  $M_w$ . Formulary,

$$L(d) = L_{free} + M_w \quad (6.1)$$

Where  $L_{free} = 20\log_{10}(d) + l_0$  and  $l_0$  is the path loss at 1 meter distance,  $l_0 = 40.22dB$  for a center frequency of 2.45GHz. The authors run a experimental work and they showed that for medium wall (thickness (20,40] cm), attenuation due to medium wall is approximately 4.2 dB. We use this result for  $M_w$  in our simulation.

## 6.3 Changes in Qualnet Source Code

Qualnet is implemented in C/C++. The necessary part of multiple AP scenarios in our simulation are channel switch and AP deassociation/reassociation. These parts were not implemented in Qualnet. We changed the source code to implement these functions. Table 6.1 shows the function names used in our simulations.

Table 6.1: Functions used in our simulation

Function Name	Description
MacDot11ManagementMyReset	To Disassociate from current AP
MacDot11MyChangeToChannel	To change channel when associates with a different AP
MacDot11MyJoin	To associate a new AP
MacDot11MyJoinComplete	To set IP address of destination AP

# Chapter 7

## Conclusion and Future Works

In this thesis we proposed a new MAC protocol for multihomed user in WLANs. Our protocol selects the AP that has the highest probability of successful transmission before each packet transmission according to the outcome of the last transmission and long-term channel statistics such as burst length of error process and packet error rate. The proposed protocol switches between APs according to transitions probabilities. These probabilities are determined from the channel model that is modeled as a two-state Markov chain.

Unlike previous works, we showed that mean error burst length and mean packet error rate of packet loss process depends on number of user in the system, channel coherence time and the amount of node traffic. Additionally, in this thesis we defined two issues that occur in a multi-user scenario. The first issue is a decreasing channel correlation due to the delay and the second issue is collision. We presented solutions to these problems and compared the protocol with and without the solutions to understand the usefulness of the solutions. The results show that with the proposed solution our protocol gives better performance.

We use iterative transmission scheme with multihoming. In this iterative transmission scheme, destination needs to receive  $K$  packets successfully to recover the original data and due to channel errors and collisions the source should send  $N$  packets,  $N \geq K$ . It is a well known fact that sending multiple copies of the same packet over multiple channels increases packet delivery ratio. However, we do not send multiple copies of the same packet since it causes delay that is not tolerable



for multimedia application. Our first aim is to reduce the redundant data by minimizing the total number of packets sent by the source. In another word, we aim to maximize the efficiency,  $R = K/N$ . The other aim is to minimize the delay that the time until  $K$  packets are received successfully.

We compared our MAC protocol with fixed path scheme that is the standard proposal in which a node uses only one AP during all transmissions. Simulation results have showed that our protocol outperforms fixed path scheme in terms of efficiency ( $K/N$ ) and the delay. This is basically due to the fact that our protocol can switch between the channels and thus bursty effect of channels can be mitigated.

As a future work interference effect of channels may be investigated. Interference may cause more packet error rate and delay. To find possible solution to this problem is one of our future work. More state Markov chain can be used to model the packet loss process and improvement and disadvantages of more than two state Markov chain may be researched. One of the other future work is to test our MAC protocol for fast fading channel.

It is important to reorder packets at the destination for video and voice application. Investigating the reordering problem is our future work. Last future work is to simulate rate adaptation algorithm defined in this thesis.

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