Efficient MAC Protocols using Directional Antennas for Ad Hoc Networks

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Abstract

A Two-Channel MAC Protocol for Ad Hoc Networks using Directional Antennas

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Use of directional antennas in wireless networks has been widely studied in recent years. Since the Medium Access Control (MAC) protocol of the IEEE 802.11 standard is designed for the use of omnidirectional antennas, it cannot work properly when directional antennas are used. To maximize the system throughput in mobile ad hoc networks, one needs to design new MAC protocols that work well with directional antennas. In this thesis, we propose a new MAC protocol that can efficiently utilize the large throughput offered by directional antennas in ad hoc networks. The proposed MAC protocol makes use of two main modes; (i) omnidirectional mode where one antenna is used for the transmission of users' control frames, and (ii) directional mode where antenna arrays are used for the transmission of data frames. We define two separate channels, one for control information and the second for data transmission. We incorporate a simple routing protocol to deal with the out-of-range problem encountered in practical scenarios. The proposed protocol takes into consideration the effect of interference arising from the side-lobes of different stations. Using a practical ad hoc network model, we present performance comparisons with different MAC protocols (including the IEEE 802.11) where we show the large throughput gains achieved using the proposed MAC protocol.

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LIST OF ABBREVIATIONS

AP	access point
BSA	basic service area
BSS	basic service set
CFP	contention-free period
CMA	constant modulus algorithm
CP	contention period
CSMA/CA	carrier sense multiple access with collision avoidance
CSMA/CD	carrier sense multiple access with collision detection
CTS	clear to send
DCF	distributed coordination function
DCTS	directional clear to send
DIFS	distributed coordination function IFS
DMAC	directional medium access control
DNAV	directional network allocation vector
DOA	direction of arrival
DRTS	directional request to send
DSSS	direct sequence spread spectrum
EC-mode	extend omnidirectional transmission range communication mode
FDMA	frequency division multiple access
FHSS	frequency hopping spread spectrum

FIR	finite impulse response
GPS	global positioning system
IBSS	independent basic service set
IFS	interframe space
IR	infrared ray
LLC	logical link control
LSCMA	least squares constant modulus algorithm
MAC	medium access control
ML	maximum likelihood
MMAC	multi-hop RTS MAC
MMSE	minimum mean squared error
MPDU	MAC protocol data unit
NAV	network allocation vector
OCTS	omnidirectional clear to send
OC-mode	omnidirectional transmission range communication mode
ORTS	omnidirectional request to send
PDU	protocol data unit
PHY	physical layer
PIFS	point coordination function IFS
RTS	request to send
SIFS	short IFS
SINR	signal-to-interference-and-noise ratio

STA	wireless LAN station
SWAMP	smart antennas based wider-range access MAC protocol
ULA	spaced linear array
VCS	virtual carrier sense
WLAN	wireless LAN

Chapter 1

Introduction

1.1 Motivation

Mobile ad hoc networks consist of mobile nodes that randomly move in an autonomous area. Such networks are widely used in airport waiting rooms, hotel lobbies, and some military environments, where a wired network access is not available. Normally, these ad hoc nodes are assumed to have only omnidirectional antennas [1]. The distributed coordination function (DCF) of MAC layer protocols defined in IEEE 802.11 standard is usually used in ad hoc networks. Using carrier sense multiple access with collision avoidance (CSMA/CA), all nodes in an ad hoc network contend for a single channel. However, one problem of sharing a single channel is that when the number of nodes increases, the performance of the whole system will dramatically degrade due to the large number of collisions. To solve this problem, directional antennas are introduced to improve the spatial use of the channel [2]-[5]. By using directional antennas, the single channel is divided into several sub-channels. As a result, simultaneous transmissions on each channel can be achieved. This in turn, leads to increase in the total system throughput of the ad hoc network. In order to achieve this increase in throughput, one needs to adapt the existing MAC protocols to enable an efficient use of directional antennas.

In this thesis, we propose an efficient MAC protocol that is basically independent from upper layers. In that, the operation of the proposed MAC does not need to rely on upper layers for beamforming information and can be used in randomly-move ad hoc networks. Based on a performance comparison, it is shown that the proposed MAC protocol can achieve a much better performance than the standard IEEE 802.11 protocol and other directional MAC (DMAC) protocols [3] introduced in the literature.

To improve the throughput performance of mobile ad hoc networks, usually, researchers use two approaches. The first approach makes use of the frequency division multiple access (FDMA) technique as in [6], [7], [9], [10] and references therein. Given the most popular band for the IEEE 802.11 standard being the 2.4GHz and 5.2GHz bands, the IEEE 802.11b standard suggests that the frequency distance between two center frequencies of two adjacent channels should be at least 25MHz far away. The standard can in this case provide some orthogonal frequency channels in the band. For instance, Kobayashi in [7] proposed a multiple channel MAC protocol based on omnidirectional antennas. In this protocol, ad hoc stations are able to select a channel among N available channels by checking the status of these channels. Similar to [7], Jain et al. [6] divide the available bandwidth into N+1 non-overlapping frequency channels. One is a control channel, on which request to send (RTS), clear to send (CTS) and ACK are transmitted. In the paper, the authors in [6] use receiver based channel selection (RBCS) to select a data channel. Even though the above-mentioned works were aimed at improving the system throughput through multiple frequency channels, no use of directional antennas was discussed.

The second approach, taken to improve the overall system throughput, is through the use of directional antennas (also called smart antennas) at each ad hoc station [2]-[5]. Using directional antennas, Ko et al. [2] proposed two MAC schemes: one uses directional RTS (DRTS); the other uses both DRTS and omnidirectional RTS (ORTS). Also, Choudhury et al. [5] proposed two MAC schemes: a Directional MAC (DMAC) and multi-hop RTS MAC (MMAC). Based on the performance results presented in [5], the authors showed that both the DMAC and MMAC perform better than the IEEE 802.11 (although the performance is dependent on some conditions in the network). To extend the range of smart antennas in ad hoc networks, Takata et al. [3] and [4] proposed what is called smart antennas based wider-range access MAC protocol (SWAMP). This protocol consists of two main modes: omnidirectional transmission range communication mode (OC-mode) and extended omnidirectional transmission range communication mode (EC-mode). Also in their work, the authors performed a performance comparison and shown that the simple EC-mode performs as good as the MMAC.

The common problem in [2]-[5] is the hidden terminal problem, resulting from the transmission of DRTS and directional CTS (DCTS). Simply, the use of DRTS and DCTS will lead to situations where some stations (i.e., the ones that did not receive DRTS and DCTS) may not notice the ongoing transmission even if these stations are in the range of the active ones. Besides that, the shapes of transmitted beams in [2]-[5] are only determined by the position of the destination station, without taking the interfering signals into consideration.

In this thesis, we design a MAC protocol that combines the FDMA technique with directional antennas to take advantage of the large promised throughput offered by directional antennas. The proposed MAC protocol is tailored to resolve the hidden terminal problem in mobile ad hoc networks. The beamforming process, at each station, is based on the positions of the destination stations and the interfering ones.

1.2 Outline of Thesis

The thesis is organized as follows. In Chapter 2, an introduction of the IEEE 802.11 MAC sublayer protocol, which includes the distributed coordination function (DCF), point coordination function (PCF), carrier sense multiple access with collision avoidance (CSMA/CA), and backoff procedure will be given.

In Chapter 3, we introduce some basic concepts of adaptive antenna array that includes uniformly spaced linear array (ULA), beampattern, gain, adaptive array, and adaptive algorithms.

In Chapter 4, we present the proposed MAC protocol along with the simulation model.

In Chapter 5, we provide the simulation results for the proposed MAC protocol, the original 802.11b DCF protocol and the existing MAC protocols. A detailed comparison on the system throughput under different conditions is also given.

In Chapter 6, we give a brief summary and future works.

Chapter 2

IEEE 802.11 Wireless Local Area Networks

2.1 Introduction

This chapter reviews some basics of the IEEE 802.11 WLAN. In section 2.2, a comparison between WLAN and LAN is provided. Section 2.3 has the background and architecture of the IEEE 802.11 standard. In Section 2.4 and 2.5, the details of physical layer and MAC layer of the IEEE 802.11 standard are introduced.

2.2 WLANs Vs LANs

Wireless communication is a rapidly developed technology that almost everyone uses these days. Like their wired counterparts, wireless local area networks (WLAN) are being studied to provide local information service in a limited space. Compared to the wired local area networks (LAN), WLANs have their advantages and disadvantages.

Advantages:

(i). Mobility: Because of the enormous success of the Internet and the applications over it, mobile intelligent terminals, like portable computer and personal data assistant (PDA), play a more and more important role in our life. These mobile terminals can somehow be considered as extension of wired networks, and they provide better service to users than wired networks do. WLANs are fixed to their users; on the other hand, WLAN users can move with their terminals in a limited area without interrupting the ongoing communication.

(ii). Flexibility: The installation of WLANs is relatively easy, compared with LANs. Instead of propagating through wires, WLANs use free space as a media to transmit signals. This character ensures that all the space in the radio range can be covered by service access point (AP). On the contrary, in LANs, it is impossible that service can cover all the space. Besides that, WLANs are more convenient in upgrade. Increasing or decreasing the number of APs can solve the problem very easily instead of redesigning the laying of the unshielded twisted pairs (UTP). Finally, in WLANs, changing the topology of network is very simple; obviously, in LANs, it costs more.

(iii).Economic: As we know, in some cases, like place rent for a short period, road shows, and outdoor shows, having a whole new design and installation of LANs is not economic. Nevertheless, with WLANs, costs can be minimized.

Disadvantages:

(i). Interference: Interference in WLANs is caused by simultaneous transmissions by different mobile terminals that are sharing the same frequency band. In traditional 802.11 based Ad Hoc networks, only one transmitting station can transmit at a time. Although with RTS/CTS, the "hidden terminal" problem can be solved, collisions might still occur during the transmission. In the IEEE 802.11 based ad hoc network equipped with adaptive antenna array, interference are caused by other simultaneous transmissions. Some published work [1] show that for packetized voice, packet loss rates on the order of 10^{-2} are generally acceptable; for uncoded data, a bit-error-rate (BER) of 10^{-5} is regarded as acceptable.

(ii). Bandwidth: Ideally, the throughput of WLANs should approach that of LANs, but because of some physical limitations and limited available bandwidth, WLANs like 802.11b standard, only operate at data rates between 1 to 11Mb/s. That means the terminals under an AP are only sharing 11Mb/s bandwidth. The gap is large, compared with the 100Mb/s and 1GMb/s LANs.

(iii).Power Consumption: Not like devices in WLANs, which are powered by 110V or 220V provided in a building, almost all the mobile terminals are powered by batteries. As a result, power consumption is a very important issue when designing physical specifications and higher layer protocols. Here again, we have to make a tradeoff between the achieved BER performance and power consumption.

2.3 Background and Architecture of IEEE 802.11 Standard

2.3.1 The IEEE 802.11 Standard

The IEEE published the 802.11 WLAN specifications in 1997 [1], the first WLAN standard. In this standard, the physical (PHY) layer and the media access control (MAC) layer have been defined in details. The standard addresses the architecture and different functions to adapt the characteristics of WLAN standard.

Compared with the IEEE 802.3 ethernet standard, we can see that the 802.11 standard has many similarities. In general, the 802.11 standard defines:

1. Different physical layer interfaces and signaling techniques.
2. DCF function and PCF function.
3. The details of how mobile terminals move among different WLANs and when mobile terminals are in two or more overlapping 802.11 WLANs.
4. MAC layer media access control method and the interface to upper layers.
5. Data privacy and encryption of the 802.11 WLANs.

2.3.2 Architecture of the IEEE 802.11 Standard

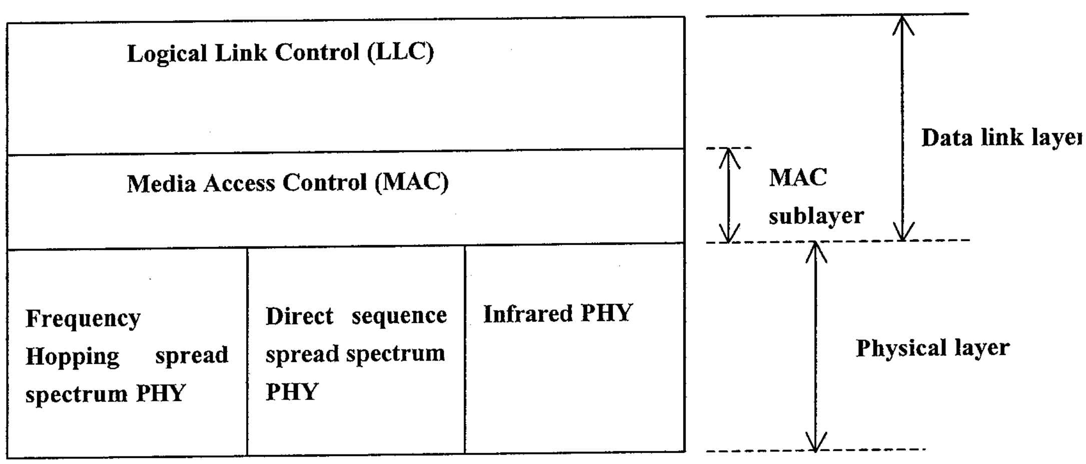


Figure 2.1 The IEEE802.11 standard protocol stack

As Figure 2.1 shows, IEEE 802.11 standard defines three types of physical layers, which are frequency hopping spread spectrum (FHSS), direct sequence spread spectrum (DSSS), and infrared (IR).

Data link layer is divided into two sub layers, which are the media access control (MAC) sub layer and the logical link control (LLC) sub layer.

Architecture of the 802.11 LAN

Normally, an 802.11 LAN is comprised of different functionality components, defined as follows:

(i). Wireless LAN station (STA)

The WLAN station [1] is the most basic component of a WLAN. A station is any device that complies with the 802.11 protocols, which include the physical layer and data link layer. Normally, the 802.11 protocols are integrated into a network interface card (NIC).

Typically, a station could be a portable computer, a PDA, or any other intelligent wireless terminal, so long as the station supports all the 802.11 station services, like authentication, privacy, and data transmission.

(ii). Basic Service Set (BSS)

The IEEE 802.11 standard defines the basic service set (BSS) [1] as the basic building block of an 802.11 wireless LAN. A BSS is comprised of a group of stations that are managed by a single coordination function, a DCF or PCF. The area where a BSS covered is called the Basic Service Area (BSA). A similar analogy is a cell in a cellular communication network. In a BSS, all stations can set up transmission with any other stations.

(iii).Independent Basic Service Set (IBSS)

The most basic WLAN topology is a set of stations [1], which recognize each other and are connected by the wireless media in a peer-to-peer fashion. This type of network topology is referred to as an independent basic service set (IBSS), which is the formal name of an ad hoc network in the IEEE 802.11 standard.

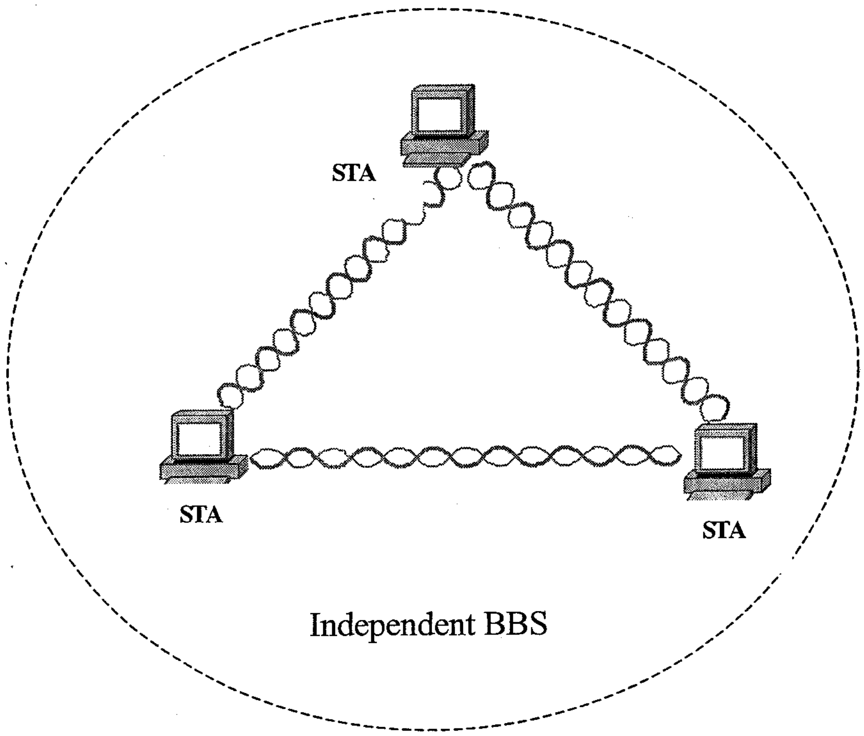


Figure 2.2 Sketch of an Independent BBS or Ad Hoc network

In an IBSS, mobile stations communicate directly with each other without using a centralized AP, through which all the traffic goes, as shown in Figure 2.2. Besides that, because there is no relay function in an IBSS, all stations have to be in the range of each other and communicate directly.

(iv). Infrastructure Basic Service Set

An infrastructure basic service set [1] is a BSS with an AP as shown in Figure 2.3. The AP is analogous to a switch in an IEEE 802.3 Ethernet network. It provides local relay function to all the stations in a BSS. All stations in a BSS have to communicate with the AP instead of communicating directly with each other. The appearance of AP not only increases the transmission range of each station, but also makes an effective hierarchy network.

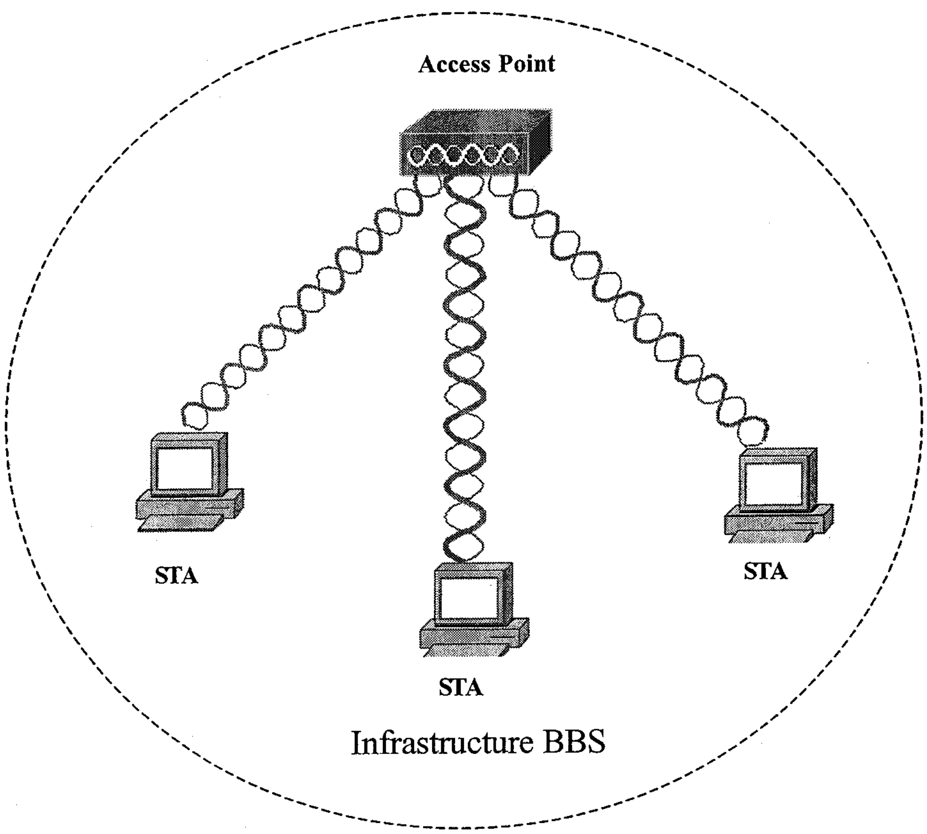


Figure 2.3 Sketch of an Infrastructure BBS network

(v). Distribution System (DS)

The distribution system (DS) [1] is a set of functions that integrate the multiple BSSs, which is implementation independent. In other words, the IEEE 802.11 standard doesn't have any restrictions on the type of a distribution system. For instance, a wired IEEE 802.3 Ethernet network, IEEE 802.4 token bus network, IEEE 802.5 token ring network, fiber distributed data interface (FDDI) metropolitan area network, or any other IEEE 802.11 wireless network could be the DS. Although the transmission medium of DS could be the same as that of the BSS, they are in different hierarchies. The DS is used to transmit data as a backbone in a network, whereas the BSS is just used to transmit data between STAs.

(vi). Extended Service Set (ESS)

An extended service set [1] is the combination of a set of infrastructure BSSs and a DS as shown in Figure 2.4. In an extended service set, the access points communicate with each other through DS and forward the traffic from one BSS to another to fulfill the communication between stations in different BSSs.

Here, the DS can be regarded as a backbone network that is responsible for MAC level transport using MAC service data unit (MSDU). On the other hand, the ESS is responsible for the control of the LLC sub layer among all the stations.

Typically, the access point determines: (i), the destination for traffic received from a BSS, (ii), whether traffic should be relayed back to a destination in the same BSS, (iii), forwarded on the distribution system to another AP, or sent to the wired network to a destination net in the extended service set.

Looking from outside, other users can consider the ESS and all its mobile stations as a single MAC layer network, whether all stations are mobile or not. Therefore, the ESS hides the mobility of the mobile stations from other users outside the ESS and allows other users that might not support the IEEE 802.11 protocol to communicate with wireless terminals in a WLAN.

An ESS can also provide Internet access for users with a device called portal. Actually, the portal is a logical gateway between the IEEE 802.11 network and other IEEE 802.X network. Similarly, the portal is analogous to a bridge. It not only provides range extension, but also the translation between different frame formats.

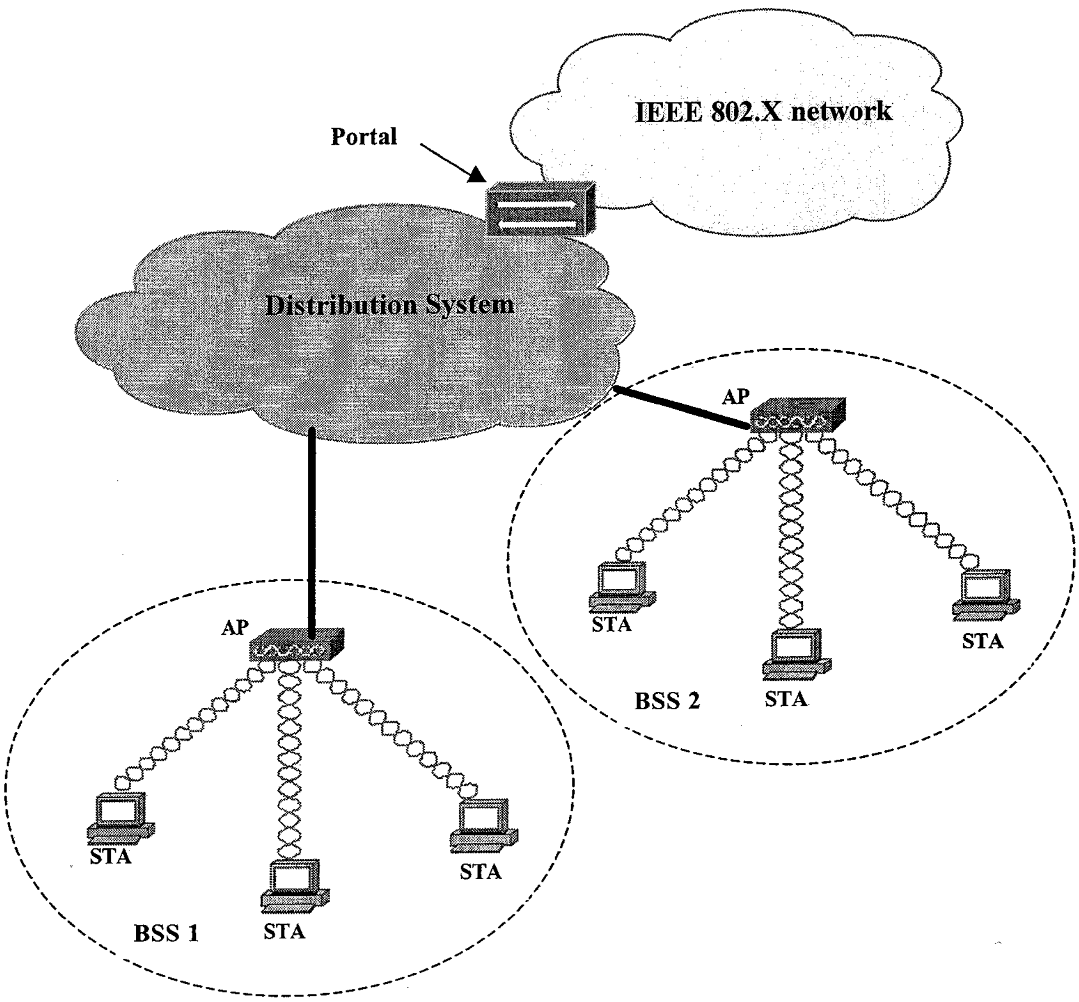


Figure 2.4 Sketch of an ESS network (including two BSSs, Distribution System, and Portal)

2.4 Physical Layer of the IEEE 802.11

The IEEE 802.11 physical layer (PHY) [1] is the interface between the MAC sub layer and the wireless media where frames are transmitted and received. The PHY provides three functions: (i), it provides an interface to upper layers and responses to the transmission and the reception of data; (ii), it uses signal carrier and different types of modulation schemes to transmit data frames over the media; (iii), it provides a carrier sense indication back to the MAC to verify activity on the media.

The 802.11 standard provide three different PHY implementations: frequency hopping spread spectrum (FHSS), direct sequence spread spectrum (DSSS), and IR. In what follows, we give an overview of these three techniques.

2.4.1 Spread Spectrum

Spread spectrum [1] is a transmission technique that uses large bandwidth to reduce the effect of interference from outside. In this, the transmitted signal is spread (in bandwidth) to reduce the peak power but maintaining the same total power. Two common spreading techniques can be used, which are FHSS and DSSS.

(i). Frequency Hopping Spread Spectrum (FHSS)

Frequency hopping utilizes a set of narrow channels and "hops" through all of them using a predetermined sequence. It utilizes the 2.4GHz industrial, scientific, and medical (ISM) band (i.e., 2.4000-2.4835GHz). In the United States, one can have a maximum of 79 channels specified in the hopping set. The first channel frequency is 2.402GHz, and all others are spaced with 1MHz, which is required by the FCC for the 2.4GHz ISM band. To avoid prolonged collision periods between different hopping sequences in a set, we define three different hopping sequence sets. Each set has 26 hopping sequences, which enable multiple BSSs to coexist in the same geographical area. This is very important for avoiding collision and enhancing the total throughput over the whole system. The minimum hop rate permitted is 2.5hops/s. Two-level Gaussian frequency shift keying (GFSK) is used to have the basic access rate of 1Mb/s. A logical 1 is presented by using frequency $F_c + f$ (F_c denotes the center frequency in modulation) and a logical 0 is presented by using frequency $F_c - f$. The other access rate of 2Mb/s uses four-level GFSK, in which 2 bits are encoded each time using four frequencies.

(ii). Direct Sequence Spread Spectrum (DSSS)

The principle of direct sequence is to spread a signal on a larger frequency band by multiplexing it with a signature or code to minimize ISI and background noise. To spread the signal, each bit is modulated by a code. At the receiver, the original signal is recovered by demodulating the received signal with the same code used at the transmitter. The 802.11 DSSS PHY also uses the 2.4GHz ISM band. The 1Mb/s basic rate is presented by using differential binary phase shift keying (DBPSK), and the 2Mb/s rate is presented by using differential quadrature phase shift keying (DQPSK). The bandwidth is divided into 11 sub-channels, 11MHz wide each. An 11-chip Barker sequence is used to spread data symbols. If DBPSK is used, the maximum channel rate is (11 chips/symbol)/(11MHz)=1Mb/s. Overlapping can be achieved by keeping the center frequencies of only two adjacent BSSs separated by at least 30MHz.

(iii). Infrared ray (IR)

The IR specification defines a wavelength range from 850 to 950 nm. Typically, the IR was designed for use in the case of line-of-site and reflected transmissions. For 1Mb/s, the infrared PHY uses a 16-pulse position modulation (PPM), where 4 data bits are mapped to 16 coded bits. For 2Mb/s, 4-PPM modulation is used where 2 data bits are mapped to 4 coded bits.

2.5 Medium Access Control Sublayer of IEEE 802.11

The MAC sublayer [1] provides many functions, including channel assignment procedures, protocol data unit (PDU) addressing, frame formatting, error checking, fragmentation, and reassembly. The channel assignment function operates quite fairly; all stations have to contend for the channel when transmitting a frame each time. There are two contention modes. One is contention period (CP); the other one is contention-free period (CFP). An AP controls the channel usage, so that stations don't need to contend for the channel. There three types of frames in IEEE 802.11 standard: (i) management, (ii) control, and (iii) data frame. The management frames are responsible for association and disassociation between stations and AP, timing and synchronization, and authentication and deauthentication. Control frames are responsible for handshaking during CP, for positive acknowledgement during the CP, and to end the CFP. Data frames are responsible for the data transmission during CP and CFP, and polling and acknowledgements during the CFP. The frame format in the IEEE 802.11 standard is shown in Figure 2.5.

From Figure 2.5, we know that the frame body MAC service data unit (MSDU) doesn't have a fixed length, which is comprised of a data payload and 7 octets for encryption/decryption if the optional wired equivalent privacy (WEP) protocol is used. The 48-bit MAC address is a unique address of each station. The 2 duration octets are used to show how much time the channel will be used by an ongoing transmission. The type field is used to indicate that a frame is a control, management, or data frame. The subtype field further indicates that the type of the frame, for example, a request to send or a clear to send control frame. The CRC field is used for error detection.

2	2	6	6	6	2	6	0-2312	4	Octet
Frame control	Duration conn. ID	Address	Address	Address	Sequence control	Address	Frame body	CRC	

2	2	4	1	1	1	1	1	1	1	1
Protocol version	Type	Subtype	To DS	From DS	Last fragment	Retry	Power mgt	More data	EP	EP

Figure 2.5 IEEE 802.11 standard frame format

2.5.1 Distributed Coordination Function

The DCF is a basic access function [1] in the IEEE 802.11 standard. It supports asynchronous data transmission on a best effort basis. According to the standard, all wireless stations must support the DCF. The DCF is only used in the ad hoc-type network, and it also can be used in an infrastructure network either running solely or combined with the PCF as shown below in Figure 2.6.

From the figure, we can see that the DCF is directly on top of the physical layer. It supports contention services where each station must contend for the channel when it has an MSDU to send. After one MSDU is transmitted, stations must contend for the channel again in order to provide fair services.

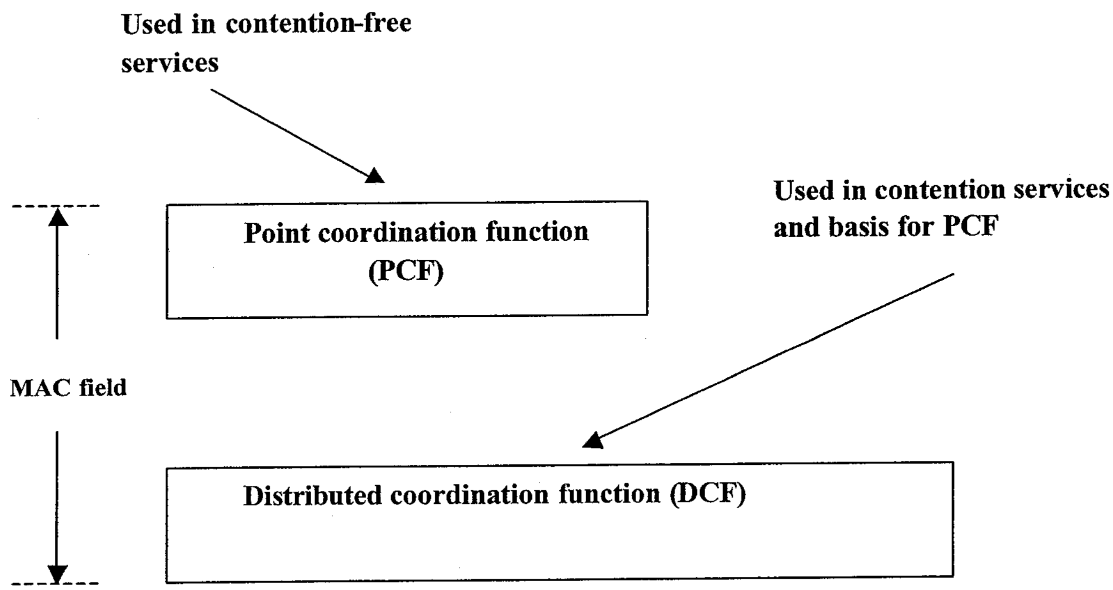


Figure 2.6 MAC architecture

The DCF is based on carrier sense multiple access with collision avoidance (CSMA/CA). CSMA/CD (collision detection) is not used here because a station cannot listen to the channel and transmit simultaneously. In the IEEE 802.11 standard, carrier sense is done at the air interface (physical carrier sensing), and at the MAC sublayer (virtual carrier sensing). Physical carrier sensing senses other users by studying all received packets, while virtual carrier sensing senses other current transmission by looking into the duration field.

When sending a MAC protocol data unit (MPDU), a source station writes the duration information in the MPDU, including MPDU header of request to send (RTS), clear to send (CTS), and data frames to perform the virtual carrier sensing. As we mentioned before, the MPDU contains a duration field that indicates the amount of time (in microseconds) that the channel will be used by this MPDU. Other stations that receive this MPDU use the information of the duration field to adjust their network allocation vector (NAV), which shows the amount of time that the station will wait until the current transmission is over and the channel can be sensed again. In other words, if a station's NAV is not zero, the channel will be considered to be busy. As a result, no action needs to be taken until the NAV decreases to zero.

There are three types of interframe space (IFS) time intervals specified in the IEEE 802.11 standard: from short to long, which are short IFS (SIFS), point coordination function IFS (PIFS), and distributed coordination function (DIFS). Obviously, stations waiting for an SIFS will have advantage over those waiting for a PIFS or DIFS when contending for the channel. That is to say, SIFS has the highest-priority access to the channel. In the basic access mode, a station senses the channel first. If the channel is idle, the station waits a DIFS period and then senses the channel again. If the channel is still idle, it sends its MPDU immediately. The destination station checks the CRC field of the received frame. If the check is correct, the destination station sends a positive acknowledgment frame (ACK) back to the source station, indicating that the transmission is successful. We should note that when the source station sends its MPDU, the duration field of MPDU is also filled with the amount of time that indicates how long the present transmission may take. All stations receiving this MPDU look into the duration field and write their NAV. Based on this amount, stations in the BSS know how long they should wait until the current traffic is over. The time amount in NAV should include an SIFS and an ACK following the data frame. Figure 2.7 below illustrates the process of a transmission:

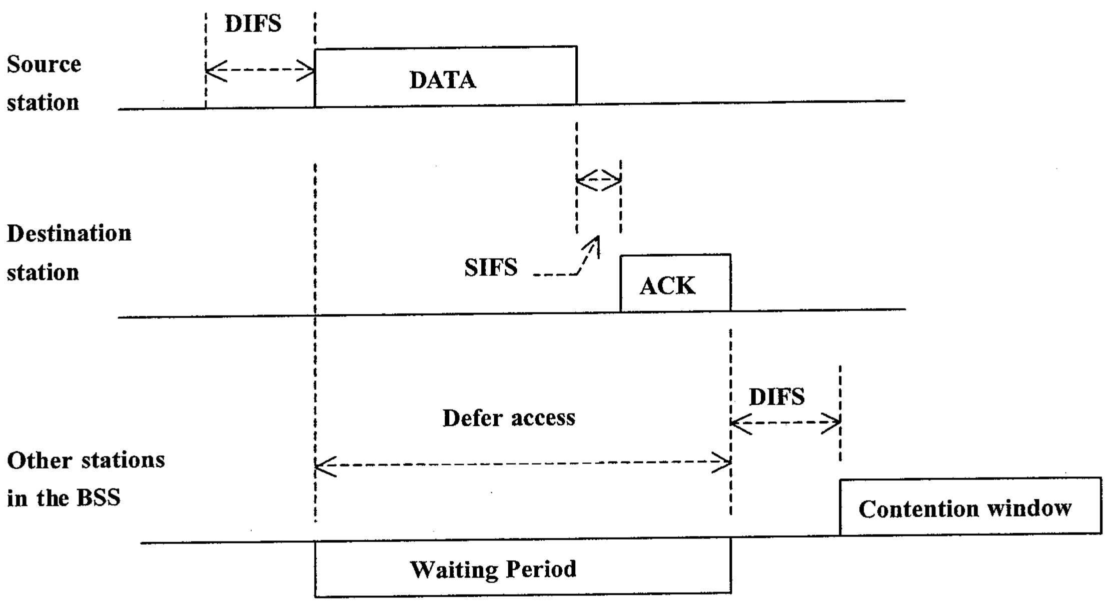


Figure 2.7 Transmission process without RTS/CTS

As we know, a source station in a BSS cannot hear its own data transmission. When a collision happens, the source station has no idea of the collision and keeps transmitting until finishing the MSPU. Under the circumstances that the MPDU is large, bandwidth wasting is considerable. As a result, a RTS/CTS mechanism is proposed to solve the bandwidth-wasting problem. One of an advantage of RTS/CTS mechanism is to let the RTS collide with the RTS instead of the collision between data frames when data traffic are heavy. Because the RTS and CTS are relatively small (RTS is 20 bytes and CTS is 14 bytes), compared to the data frame (maximum 2346 bytes), if collision does occur, the bandwidth wasting is relatively small.

The source station first transmits an RTS control frame, if it has some data frames to send, after successfully contending for the channel. All stations can hear this RTS and read its duration field. All stations other than the destination station will update their NAVs, according to the amount of time in the duration field. After receiving an RTS, the destination station sends a CTS frame back to the source station, after an SIFS period. All stations can hear the CTS, and all stations except the source station update their NAVs according to the duration field in the CTS. Upon successfully receiving the CTS, the source station waits an SIFS time and sends the data frame. The rest of process is like the case without RTS/CTS. Figure 2.8 shows this operation.

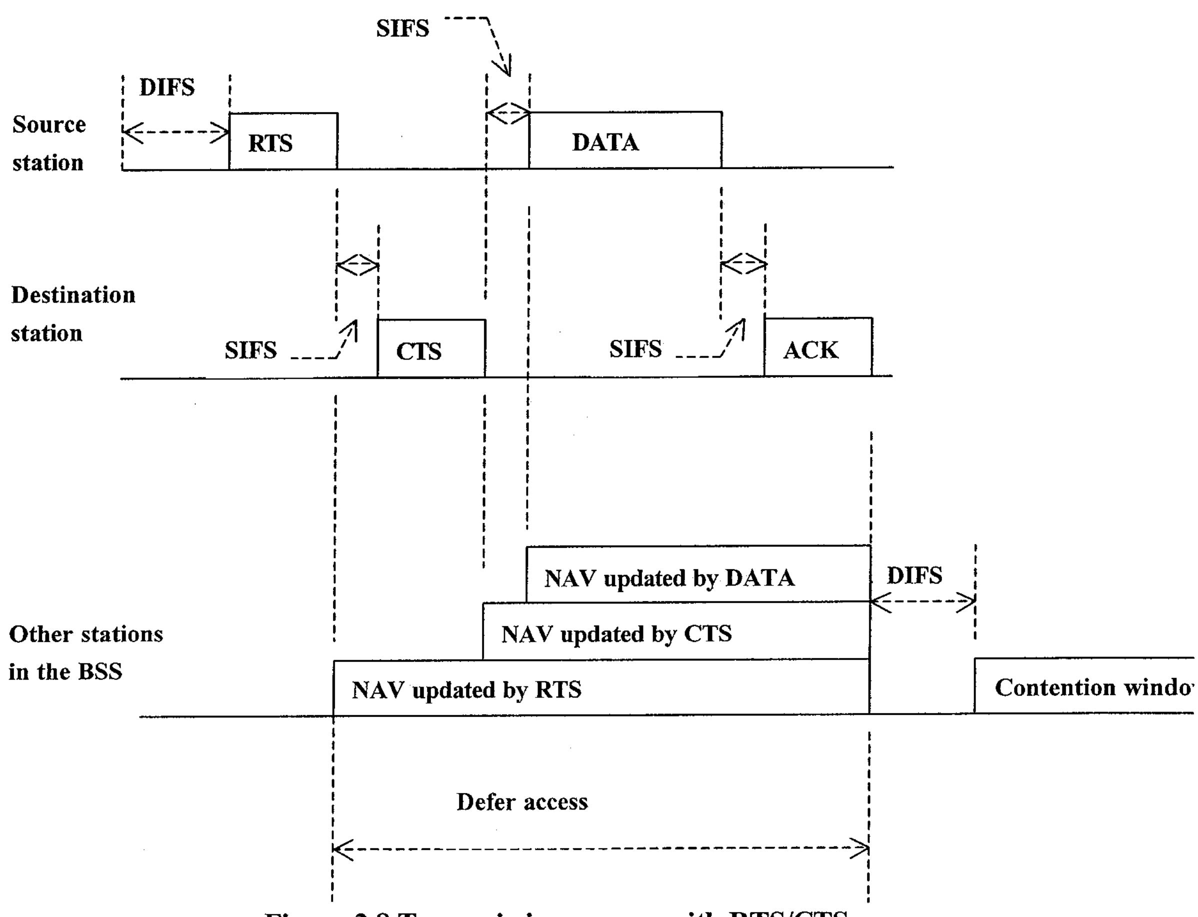


Figure 2.8 Transmission process with RTS/CTS

We should note that the stations, except the source and destination stations, update their NAVs when receiving the latest RTS, CTS, or DATA frame. This characteristic helps to solve the "hidden terminal" problem.

A station has the option to choose to never use the RTS/CTS mechanism, always use RTS/CTS mechanism, or use RTS/CTS when the length of MSDU exceeds a threshold value called RTS_Threshold (manageable parameter). As we mentioned before, instead of collision of MSDU, collision of RTS can save bandwidth. However, in a lightly-load system, the transmission of RTS and CTS causes additional overhead compared with no RTS/CTS mechanism.

A random backoff procedure does the job of the collision avoidance in CSMA/CA mechanism. Every station maintains a random backoff timer initially, which is T*Slot_Time long. Here, T is a random number between 0 and 7, and Slot_Time is defined by the physical layer, different Slot_Time corresponding to different physical layer implementations. When a station has a frame to send, it senses the channel. If the channel is busy, the station will wait until the channel is idle for a DIFS period; if the channel is idle, it will look into its backoff timer. If the timer is not zero, it will decrease the timer by 1 by waiting a Slot_Time period; if the timer is equal to zero, it sends its frame. In the period when a station decreases its backoff timer, if the channel becomes busy, the station will freeze its timer, and wait for the channel being idle again. Figure 2.9 shows this process.

A collision will occur when two or more timers are decreased to zero at the same time. In this case, these stations have to generate a new T, which is randomly between 0 and 15. For every transmission attempt, the backoff timer is $[2^{2+i} \times ran(a)] \times Slot\_Time$, where i is the number of consecutive times a station attempts to send an MPDU, $ran(a)$ is a uniform random variable in (0, 1), and $[x]$ means the largest integer less than or equal to x.

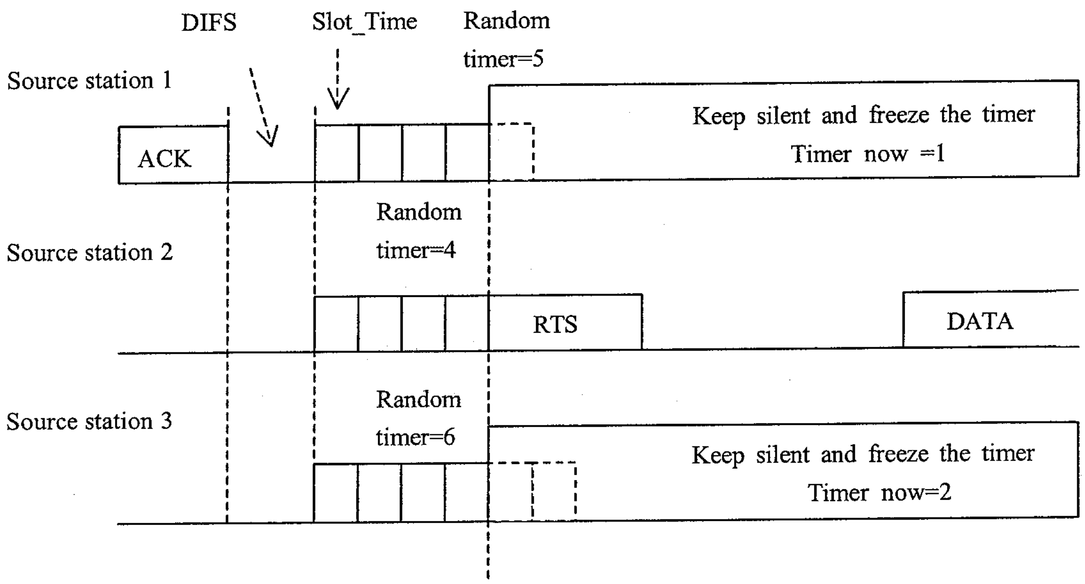


Figure 2.9 the procedure of CSMA/CA, I

In the IEEE 802.11 standard, the idle period after a DIFS period is defined as the contention window (CW). Figure 2.10 shows this process.

One of the advantages of this medium access method (shown in Figure 2.10) is that it balances the throughput among stations. It is fair because every station must contend for the channel for every MSDU transmission. All stations have equal priority to access the channel and transmit their frame. In other words, the DCF is not appropriate for real time traffic, like audio and video data, because it cannot guarantee a fixed delay for a system.

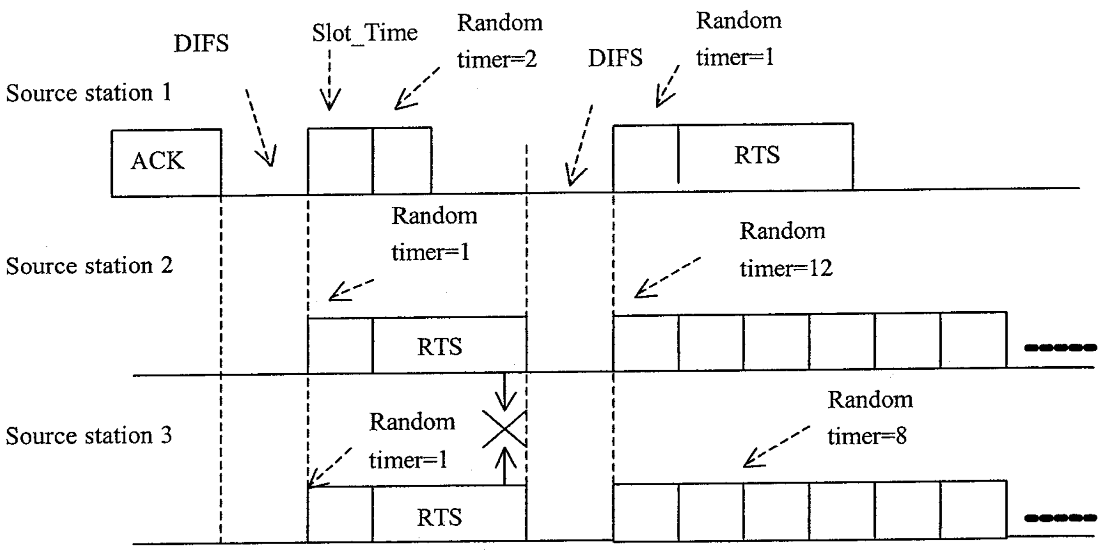


Figure 2.10 the procedure of CSMA/CA II

2.5.2 Point Coordination Function

The PCF is an optional function that offers contention-free frame transfer, and it is connection-oriented. In PCF, the point coordinator (PC) is performed by the AP within a BSS. Furthermore, it does the job of polling, and it provides the polled stations with contention-free access to the channel. A CF-aware station is a station that is capable of using a CF period (CFP). However, the method of how an AP creates a polling table and the polling sequence are not specified in the IEEE 802.11 standard.

The PCF has to coexist with the DCF and works on top of the DCF as shown in Figure 2.11. A CFP repetition interval is a unit in the PCF function. There are two different parts in a CFP repetition interval: one is contention-free period; the other one is contention period. A beacon frame is used at the beginning of the CFP. The function of the beacon frame is related to synchronization and timing as shown in Figure 2.11.

The duration of the CFP repetition is a manageable parameter that is always an integral number of beacon frames. CFP_Rate decides the duration of the CFP, and CFP_Max_duration determines the maximum size of the CFP. As a result, the duration of a CFP in any CFP repetition interval is determined by the AP. If the load is very light, the AP might shorten the duration of CFP and provide the CP in most of a CFP repetition interval. On the contrary, the AP can maximize the duration of CFP in order to provide services that have higher priority.

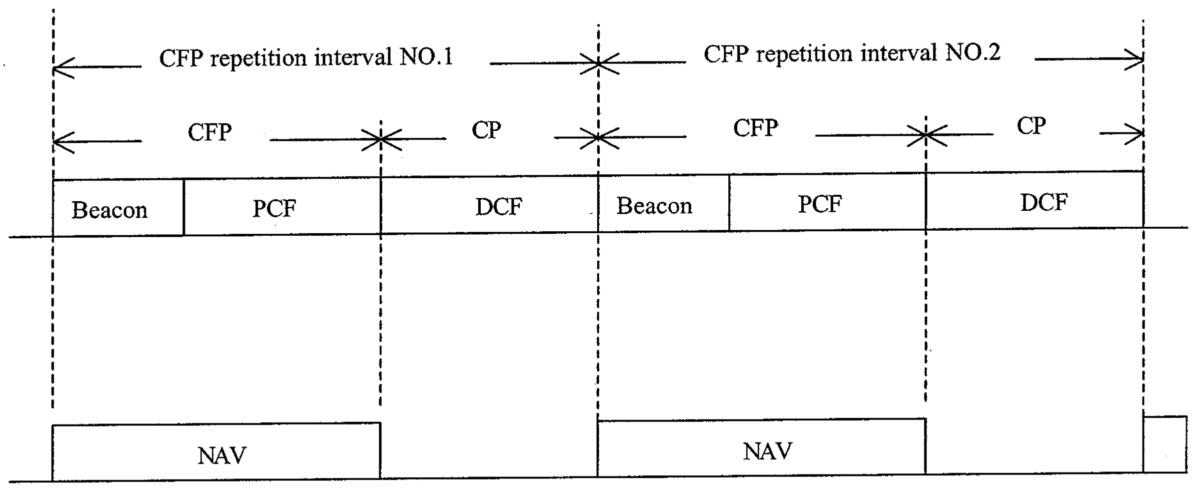


Figure 2.11 The structure of a CFP repetition interval

2.6 Summary

In this chapter, some fundamental knowledge in the IEEE 802.11 standard is introduced. In the following chapters, we will focus on DCF mode with RTS/CTS (used in ad hoc networks).

In the next chapter, basics of adaptive antenna array and adaptive algorithms will be discussed.

Chapter 3

Adaptive Antenna Arrays

3.1 Introduction

In this chapter, some basics relating to adaptive antenna array will be reviewed. In section 3.2, the definition of uniformly spaced linear array will be introduced. Section 3.3 will show the beamforming of antenna array. And space beampattern and element spacing will follow in Section 3.4. Finally, in Section 3.5 and 3.6, space adaptive arrays and adaptive algorithms will be discussed.

3.2 Uniformly Spaced Linear Array

An antenna array [12]-[15] consists of a set of antenna elements that are spatially distributed at known locations with reference to a common fixed point. There are many various geometries of antenna elements arrangement. If the centers of the elements of the array are aligned along a straight line, the array is called a linear array. If the space between array elements is equal, the array is called a uniformly spaced linear array. If the centers of array elements lie on a circle, the array is called a circular array. If the centers of array elements lie on a single plane, the array is called a planar array. We can see that linear array and circular arrays are special cases of the planar array. Figure 3.1 shows an M-element uniformly spaced linear array.

In the figure below, the elements are spaced by a distance of d. A plane wave arrives at the array from a direction θ off the array broadside, where θ is called the direction-of-arrival (DOA) of the received signal. The received signal at the first element can then be expressed as

$$z_1(t) = u(t)\cos(2\pi f_c t + \psi(t) + \alpha), \tag{3.1}$$

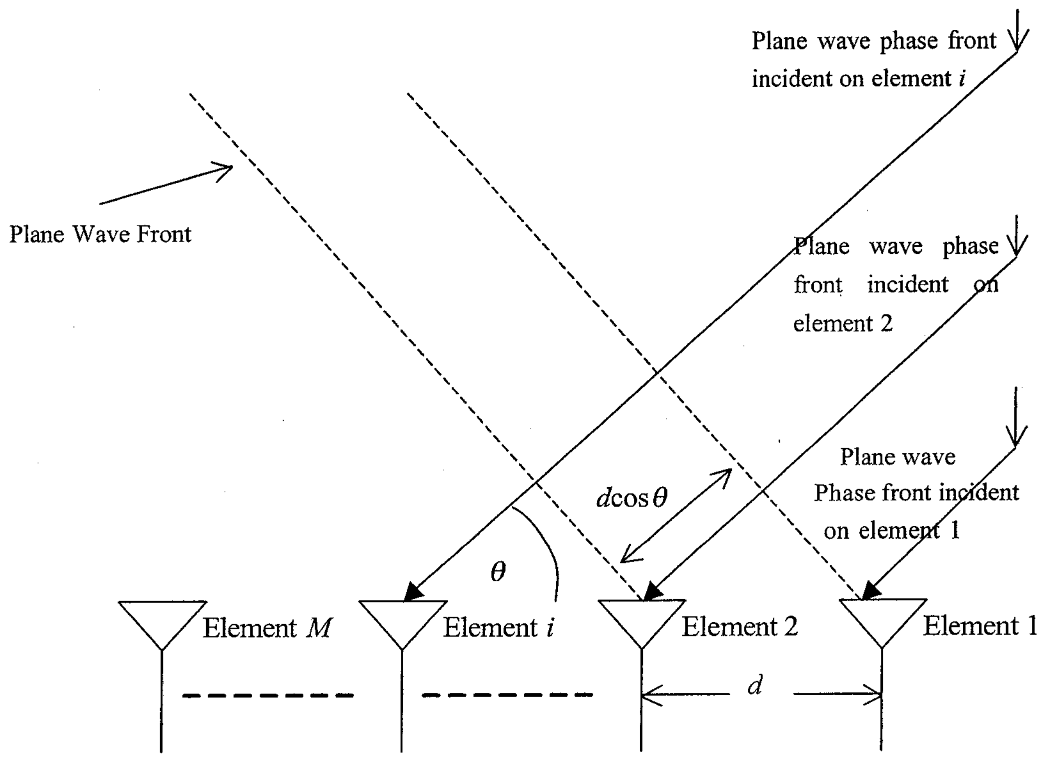


Figure 3.1 Picture of an M-element uniformly spaced linear array

where f_c is the carrier frequency of the modulated signal, $\psi(t)$ is the information carrying component, $u(t)$ is the amplitude of the signal, and α is a random phase. We also use the complex envelope representation of $z_1(t)$, given by

$$x_1(t) = u(t)\exp\{j(\psi(t) + \alpha)\}. \tag{3.2}$$

The received signal at the first element $z_1(t)$ in (3.1) and its complex envelope

$x_1(t)$ in (3.2) are related through

$$z_1(t) = \mathrm{Re}\left[x_1(t)\exp\{j(2\pi f_c t)\}\right], \tag{3.3}$$

where Re [•] stands for the real part of [•].

Now we assume that the first element is a reference point. Furthermore, we assume that the plane wave is generated far away from the array and transmitted through a non-dispersive medium. As a result, the received signals at other elements should be represented by a time-advanced or time-delayed version of the signal received at the first element. From Figure 3.1, we note that the plane wave arrived at the second element should travel $d\cos\theta$ distance more than that arrived at the first element. The time delay relates to this distance can be represented by

$$\tau = \frac{d\cos\theta}{c}, \tag{3.4}$$

where c is the speed of light. Therefore, the received signal at the second element can be expressed as

$$z_2(t) = z_1(t-\tau) = u(t-\tau)\cos(2\pi f_c(t-\tau) + \psi(t-\tau) + \alpha). \tag{3.5}$$

If the carrier frequency f_c is large, compared to the bandwidth of the received signal, then the modulating signal may be regarded as quasi-static during time intervals of order τ and equation (3.5) can be reduced to

$$z_2(t) = u(t)\cos(2\pi f_c t - 2\pi f_c \tau + \psi(t) + \alpha). \tag{3.6}$$

The complex envelope of $z_2(t)$ is therefore

$$\begin{aligned} x_2(t) &= u(t)\exp\{j(-2\pi f_c \tau + \psi(t) + \alpha)\} \\ &= x_1(t)\exp\{-j(2\pi f_c \tau)\}. \end{aligned} \tag{3.7}$$

Now, we know that the time delay of the signal at the different antenna elements leads to a phase shift represented by $\exp\{-j(2\pi f_c \tau)\}$. Combining equation (3.4) and (3.7), we get

$$x_2(t) = x_1(t)\exp\left\{-j(2\pi f_c \frac{d\cos\theta}{c})\right\}$$

$$= x_1(t)\exp\left\{-j(\frac{2\pi}{\lambda} d\cos\theta)\right\}, \tag{3.8}$$

where λ is the wavelength of the radio wave. Hence, the complex envelope of the received signal at the ith element may be presented by

$$x_i(t) = x_1(t)\exp\left\{-j(\frac{2\pi}{\lambda}(i-1)d\cos\theta)\right\} \qquad i = 1,\ldots,M. \tag{3.9}$$

Using vector notation, we define

$$X(t) = \begin{bmatrix} x_1(t) \\ x_2(t) \\ \vdots \\ x_M(t) \end{bmatrix}, \tag{3.10}$$

and

$$Q(\theta) = \begin{bmatrix} 1 \\ e^{-j\frac{2\pi}{\lambda}d\cos\theta} \\ \vdots \\ e^{-j\frac{2\pi}{\lambda}(M-1)d\cos\theta} \end{bmatrix}. \tag{3.11}$$

Combining (3.10) and (3.11), we get the vector form of (3.9).

$$X(t) = Q(\theta)x_1(t). \tag{3.12}$$

The vector $X(t)$ is defined as the array input vector or the illumination vector, and $Q(\theta)$ is called the steering vector. In this case, the steering vector is only a function of the angle-of-arrival. However, in general, the steering vector is also a function of the individual element response, the array geometry, and signal frequency.

In the above discussion, the bandwidth of the impinging signal expressed in equation (3.9) is assumed to be much smaller than the reciprocal of the propagation time across the array. Any signal satisfying this condition is referred to as narrowband; otherwise it is referred to as wideband. In most of the discussion that follows, the signal is assumed to be narrowband.

We can extend the case (3.12) to a more general one. Assume that there are q narrowband signals $s_1(t),\ldots,s_q(t)$, using frequency f_c, impinging on the array with a DOA θ_i, $i = 1,2,\ldots,q$. These signals could be uncorrelated as signals come from different users, or they could be correlated as signals come from multipath but from just one user. However, at the receiver, the received signal can be expressed as

$$X(t) = \sum_{i=1}^{q} Q(\theta_i)s_i(t) + N(t), \tag{3.13}$$

where

$$Q(\theta_i) = \begin{bmatrix} 1 \\ e^{-j\frac{2\pi}{\lambda}d\cos\theta_i} \\ \vdots \\ e^{-j\frac{2\pi}{\lambda}(M-1)d\cos\theta_i} \end{bmatrix} \tag{3.14}$$

and $N(t)$ denotes the $M \times 1$ noise vector at the array elements. In matrix notation, equation (3.13) becomes

$$X(t) = Q(\Theta)s(t) + N(t) \tag{3.15}$$

where $Q(\Theta)$ is the $M \times q$ matrix of the steering vectors

$$Q(\Theta) = \lfloor Q(\theta_1) \quad \cdots \quad Q(\theta_q) \rfloor \tag{3.16}$$

and

$$s(t)=\begin{bmatrix} s_1(t) \\ \vdots \\ s_q(t) \end{bmatrix}. \tag{3.17}$$

Finally, equation (3.15) represents a common narrowband input data model.

3.3 Beamforming of Antenna Array

Beamforming is a process that is used to form the original beam pattern from different received signals at a receiver. When dealing with the received signals, we often find that the signals are usually impaired by the presence of interfering signals. If the desired signal and the interfering signal occupy the same frequency, normally, the temporal filtering cannot separate the desired signal from the interference. On the other hand, because we know that the received signal is composed of different transmitters from different locations, the spatial filtering can be used to separate the desired signal from interference. Therefore, it is necessary to collect the data over a spatial aperture.

Normally, a beamformer linearly combines the spatially received signals from the sensors of antenna array. The process of combining is almost the same as the one finite impulse response (FIR) filter does in time domain. Figure 3.2 is a block diagram of narrowband beamformer, and we only discuss the narrowband beamformer in this thesis.

In Figure 3.2, the output at time k, $y(k)$ is given by a linear combination of the data at the M sensors:

$$y(k)=\sum_{i=1}^{M} w_i^* x_i(k), \tag{3.18}$$

where * denotes complex conjugate. We use the complex envelope representation of the received signal before, therefore, both $x_i(t)$ and w_i are complex. The weight w_i is called the complex weight.

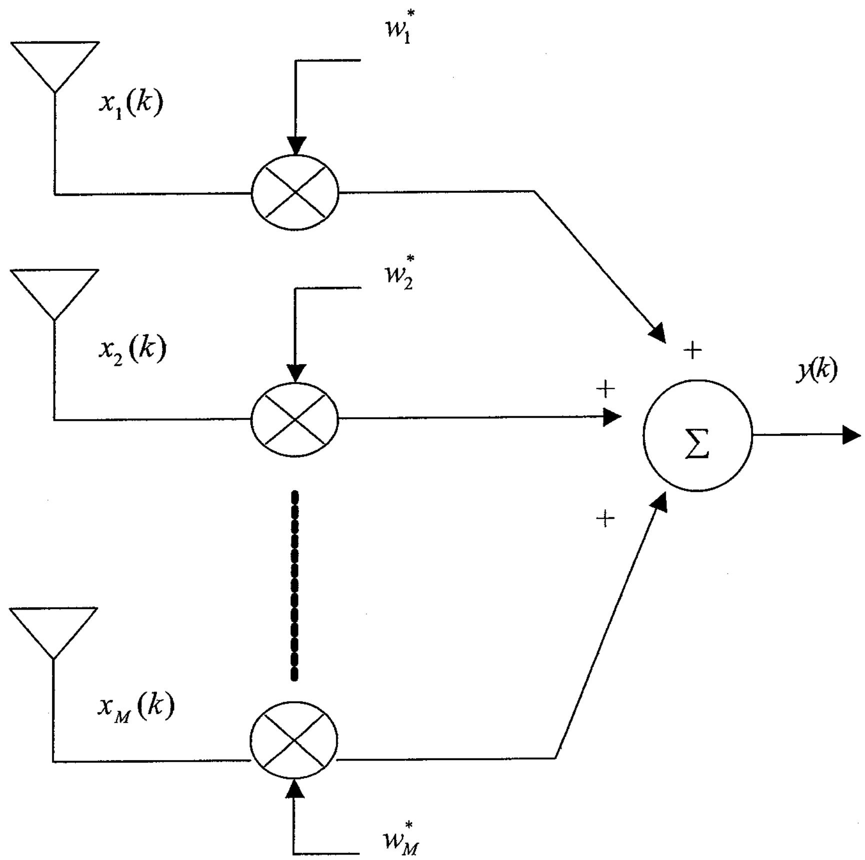


Figure 3.2 A narrowband beamformer model

Equation (3.18) can also be expressed in vector form as:

$$y(k) = w^H x(k), \tag{3.19}$$

where

$$w = \begin{bmatrix} w_1 \\ w_2 \\ \vdots \\ w_M \end{bmatrix} \tag{3.20}$$

and H denotes the Hermitian (complex conjugate) transpose. The vector w is called the complex weight vector.

Different from a narrowband beamformer, a wideband beamformer samples the propagating wave field in both space and time and is often used when signals of significant frequency extent (broadband) are of interest.

A wideband beamformer is shown in Figure 3.3. The output in this case may be expressed as

$$y(k)=\sum_{i=1}^{M}\sum_{l=0}^{K-1} w_{i,l}^{*}x_i(k-l), \tag{3.21}$$

where K-1 is the number of delays in each of the M sensor channels. Let

$$w=[w_{1,0,\ldots,}\, w_{1,K-1,\ldots,}\, w_{M,0,\ldots,}\, w_{M,K-1}]^T \tag{3.22}$$

and

$$x(k)=[x_1(k),\ldots,x_1(k-K+1),\ldots,x_M(k),\ldots,x_M(k-K+1)]^T, \tag{3.23}$$

where T denotes transpose. Equation (3.21) may also be expressed in vector form as in equation (3.19). In this case, both w and x (k) are $MK\times 1$ column vectors.

Comparing Figure 3.2 with Figure 3.3, we see that a wideband beamformer is more complex than the narrowband beamformer. Since both types of beamformers may share the same data model, we will concentrate on the narrowband beamformer in the following discussion.

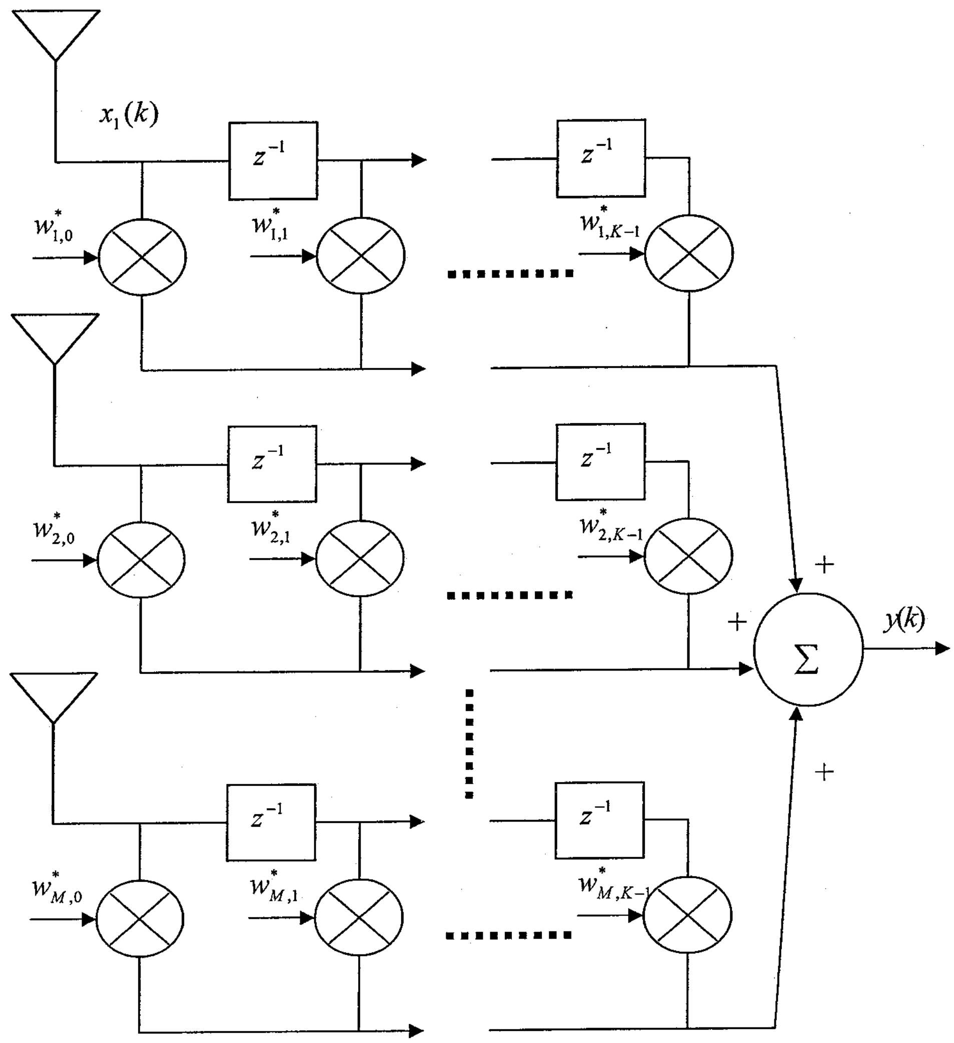


Figure 3.3 A wideband beamformer samples the signal in both space and time

3.4 Space Beampattern and Element Spacing

To explain the beampattern and element spacing problem in uniform linear arrays, we can compare the restoration problem of FIR filters in time domain with the beamforming problem in the space domain. Here, beampattern and element spacing of an antenna array may be considered as counterpart of the magnitude response of an FIR filter and the sampling period of a discrete time signal, respectively.

Consider a signal $x(t)$ given by

$$x(t) = \sum_{i=1}^{q} a_i \exp\{j(2\pi f_i t + \phi_i)\} + n(t), \tag{3.24}$$

where f_i, a_i, and ϕ_i are the frequency, amplitude, and phase, respectively, of the ith sinusoid. We sample the signal with a sampling period T_s unrelated to the frequency of the unknown sinusoid and let $x(l)$ denote the signal at time instant lT_s. We then have

$$x(l) = \sum_{i=1}^{q} a_i \exp\{j(2\pi f_i (lT_s) + \phi_i)\} + n(lT_s). \tag{3.25}$$

The sampled signal is applied to an FIR filter with $M-1$ delay units, as shown in Figure 3.4.

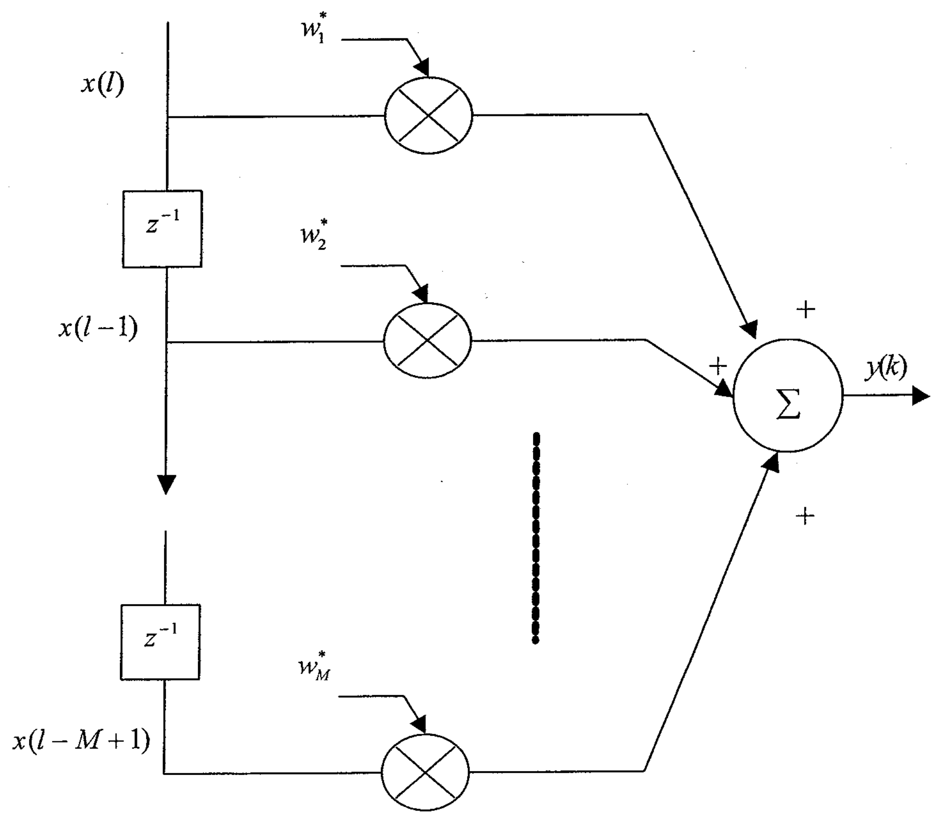


Figure 3.4 An FIR filter in time domain

At time instant lT_s, the filter input and the $M-1$ outputs of the delay units may be expressed as

$$x(l) = \sum_{i=1}^{q} a(f_i)s_i(l) + n(l), \tag{3.26}$$

where $x(l) = [x(l), x(l-t),..., x(l-M+1)]^T, n(l) = [n(lT_s),..., n((l-M+1)T_s)]^T$

$$a(f_i) = \begin{bmatrix} 1 \\ e^{-j2\pi T_s f_i} \\ \vdots \\ e^{-j2\pi(M-1)T_s f_i} \end{bmatrix}, \tag{3.27}$$

and

$$s_i(l) = a_i \exp\{j(2\pi f_i(lT_s) + \phi_i)\}. \tag{3.28}$$

Comparing equations (3.26), (3.27) with equations (3.13), (3.14), we can see that in a narrowband ULA, there is a correspondence between the normalized element spacing, $\frac{d}{\lambda}$, and the sampling period, T_s, in the FIR filter. Also $\sin\theta_i$ can be related to the temporal frequency f_i of the FIR filter input.

As the ULA and the FIR filter share the same characteristics, the Nyquist sampling theorem applied to the FIR filter in time domain may be applied to the ULA in space domain. In the ULA case, the sampling rate corresponds to the inverse of the normalized element spacing, and the highest frequency corresponds to 1 (since $\sin\theta_i$ is always less than 1). From the Nyquist sampling theorem, to avoid spatial aliasing, we should have

$$\frac{1}{\frac{d}{\lambda}} \geq 2 \times 1, \tag{3.29}$$

which is equivalent to

$$d \le \frac{\lambda}{2}. \tag{3.30}$$

As a result, the element spacing of an antenna array should always be less than or equal to half of the carrier wavelength. However, in order to achieve the maximum diversity, we cannot make the element spacing very small. The reason is that the closer we place two antenna elements the more the mutual correlation we get from these two elements. Therefore, the distance between elements must be large enough to avoid significant correlation. In practical cases, the distance between two elements is usually equal to a half wavelength so that we can minimize the spatial aliasing and mutual correlation at the same time.

The frequency response of an FIR filter with tap weights $w_i^*, i = 1,\ldots,M$ and a sampling period T_s is given by

$$H(e^{j2\pi f}) = \sum_{i=1}^{M} w_i^* e^{-j2\pi f T_s (i-1)}, \tag{3.31}$$

where $H(e^{j2\pi f})$ denotes the response of the filter to a complex sinusoid of frequency f. In the time domain, if we want to extract the signal with frequency f_i, we need to find a set of complex weights so that the frequency response of the filter has a higher gain at f_i and lower gains at other frequencies. Similarly, in the space domain, since f and T_s correspond to $\sin\theta$ and $\frac{d}{\lambda}$, respectively, to get the beamformer response,

$$g(\theta) = \sum_{i=1}^{M} w_i^* e^{-j\frac{2\pi}{\lambda}(i-1)d\cos\theta} \tag{3.32}$$

where $g(\theta)$ represents the response of the array to a signal with DOA equal to θ. Therefore, if we want to extract signals arriving from different directions with θ_i, we should find a set of weights that achieve higher gain at direction θ_i and lower gains at other directions.

Simply, the array response may also be expressed in vector form as

$$g(\theta) = w^H Q(\theta), \tag{3.33}$$

where w and $Q(\theta)$ are defined in equations (3.11) and (3.20). The beamformer response may also be viewed as the ratio of the beamformer output to the signal at the reference element when a single plane wave is incident on the array.

The beampattern is defined as the magnitude of $g(\theta)$ and is given by

$$G(\theta) = |g(\theta)|. \tag{3.34}$$

Using $G(\theta)$, we may define the normalized beamformer response as

$$g_n(\theta) = \frac{g(\theta)}{\max\{G(\theta)\}}, \tag{3.35}$$

where $g_n(\theta)$ is known as the normalized radiation pattern or the array factor.

The spatial discrimination capability of a beamformer depends on the size of the spatial aperture of the array. The absolute aperture size is not important; rather its size in wavelengths is the critical parameter. To illustrate this point, let us consider a uniform linear array with equal weight for each element. From equation (3.32), we get the beamformer response

$$g(\theta) = \sum_{i=1}^{M} e^{-j\frac{2\pi}{\lambda}(i-1)d\cos\theta}$$

$$= \frac{1-e^{-j\frac{2\pi}{\lambda}Md\cos\theta}}{1-e^{-j\frac{2\pi}{\lambda}d\cos\theta}} \tag{3.36}$$

$$= \frac{\sin(\pi \frac{Md}{\lambda} \cos\theta)}{\sin(\pi \frac{d}{\lambda} \cos\theta)} e^{-j\pi \frac{(M-1)d}{\lambda} \cos\theta} . \tag{3.37}$$

The beampattern of this equal-weight beamformer is shown in Figure 3.5. The polar coordinate plot of the beampattern is shown in Figure 3.6. In Figure 3.5, the normalized beampattern gain is expressed in dB.

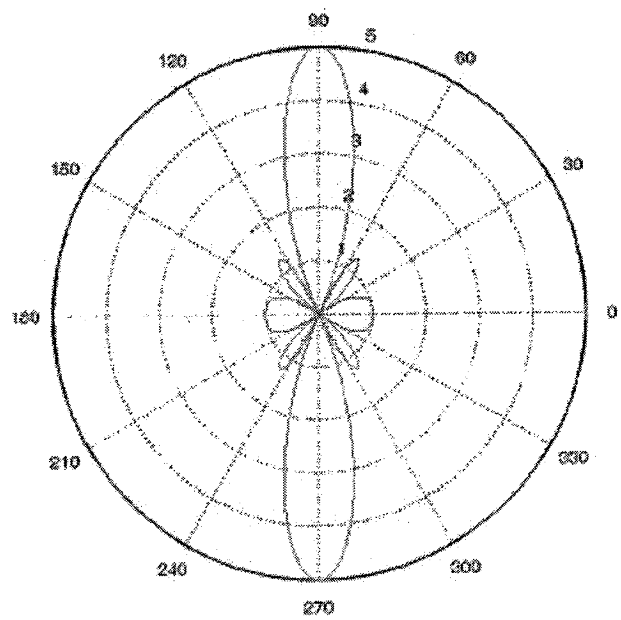


Figure 3.5 Beampattern of an equal-weight beamformer Number_of_element=4, Element_space=$\lambda/2$

From equation (3.37), we know that the null-to-null beamwidth, θ_{BW} of the array is determined by

$$\pi \frac{Md}{\lambda} \sin\theta = \pi . \tag{3.38}$$

The solution of equation (3.38) may also be expressed as

$$\theta_H = \arcsin(\frac{\lambda}{Md}) , \tag{3.39}$$

where θ_H is the peak-to-null beamwidth, or half of the null-to-null beamwidth. The null-to-null beamwidth is then obtained by

$$\theta_{BW} = 2\theta_H = 2\arcsin(\frac{\lambda}{Md}). \tag{3.40}$$

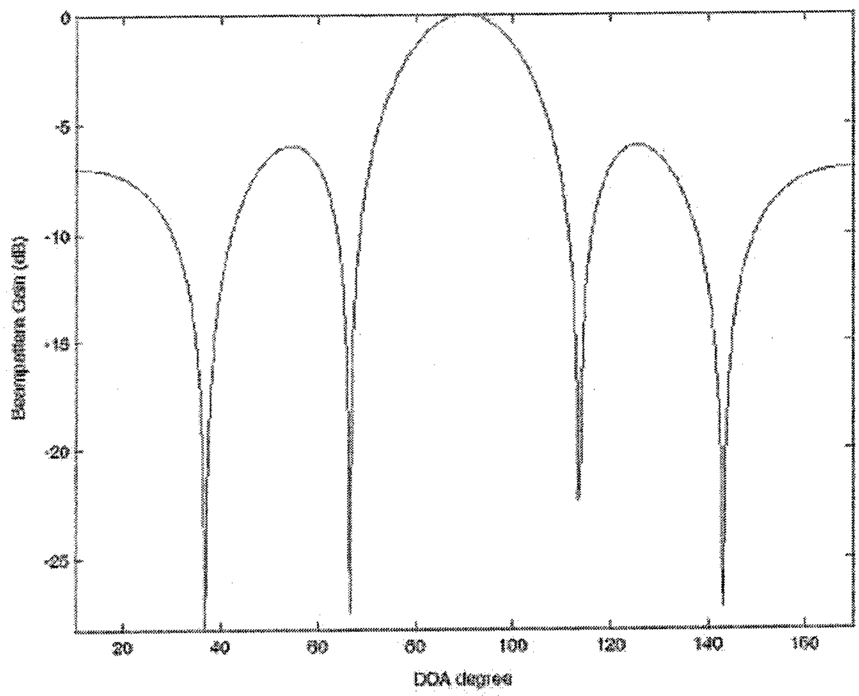


Figure 3.6 Beampattern of an equal-weight beamformer in polar form

From equation (3.40), we know that the beamwidth, which is the width of the main lobe of the beampattern, is inversely proportional to $\frac{Md}{\lambda}$. Therefore, if the aperture size in wavelengths is large, the beamwidth of the array will be small, and the beamformer will have a high spatial discrimination capability. For a general beamformer with unequal weights $w_{iue}, i = 1,2,...,M$, the weight w_{iue} may be viewed as the product of the equal weight series and a periodic weight series:

$$w_{iue} = w_{ie}\widetilde{w}_{iue}, \qquad i = -\infty,...,1,2,...,\infty, \tag{3.41}$$

where

$$w_{ie} = \begin{cases} 1 & i = 1,2,\ldots,M \\ 0 & otherwise \end{cases} \tag{3.42}$$

and

$$\widetilde{w}_{(i+kM)ue} = w_{iue}, \quad i = 1,2,\ldots,M \quad and \quad k = -\infty,\ldots,1,2,\ldots,\infty. \tag{3.43}$$

Since in time domain, the frequency response of a periodic time series contains a delta function, similarly, in space domain, the beampattern corresponding to the weight series $\widetilde{w}_{iue,}$ will have a high angle resolution. Furthermore, the resulting beampattern corresponding to the unequal weight w_{iue} thus can be viewed as the convolution of the equal-weight beampattern with a beampattern having high angle resolution. Further the beamwidth (or equivalently, the aperture size in wavelength) of the equal-weight beamformer determines the spatial discrimination capability of a general beamformer. Therefore, in addition to minimizing the mutual correlation effect between the array elements, maximizing the aperture size is another reason to keep the element spacing of the array near a half wavelength.

From Figure 3.6, we see that the beampattern of the array is symmetrical about the axis connecting the array elements. This symmetry of the beampattern is inherent in the uniform linear array. From equation (3.32), we know that $g(\theta) = g(\pi - \theta)$, thus the beampattern of the uniform linear array is symmetrical. Arrays with non-linear spacing or with other geometry, such as the circular array, do not exhibit this kind of symmetry.

3.5 Adaptive Antenna Arrays

In a mobile communication system, due to the movement of mobile stations, normally, DOAs of received signals are time varying. In this case, a fixed-weight beamformer cannot work properly against these time-varying factors. Therefore, we need an adaptive antenna array to handle this problem. An adaptive antenna array [12]-[17] is one kind of system that has the ability to adjust its beampattern and other parameters adaptively to get a best gain at the desired direction. In this case, the antenna array is referred to as adaptive beamformers, or smart antennas. A simple narrowband adaptive array is shown in Figure 3.7.

In Figure 3.7, the complex weights $w_1^* ..., w_M^*$ are controlled by a processor. The processor controls the antenna weights using an adaptive algorithm. Most adaptive algorithms are derived by first creating a performance criterion, and then generating a set of iterative equations to adjust the weights so that the performance criterion is met. There are some performance criteria, which are most useful, minimum mean squared error (MMSE), maximum signal-to-interference-and-noise ratio (SINR), maximum likelihood (ML), minimum noise variance, minimum output power, maximum gain, etc [13]-[18]. These criteria are normally expressed as cost functions, which are typically inversely associated with the quality of the signal at the array output. When the cost function is minimized or maximized (depending on the choice of the cost function), the performance criterion is met and the algorithm is said to have converged.

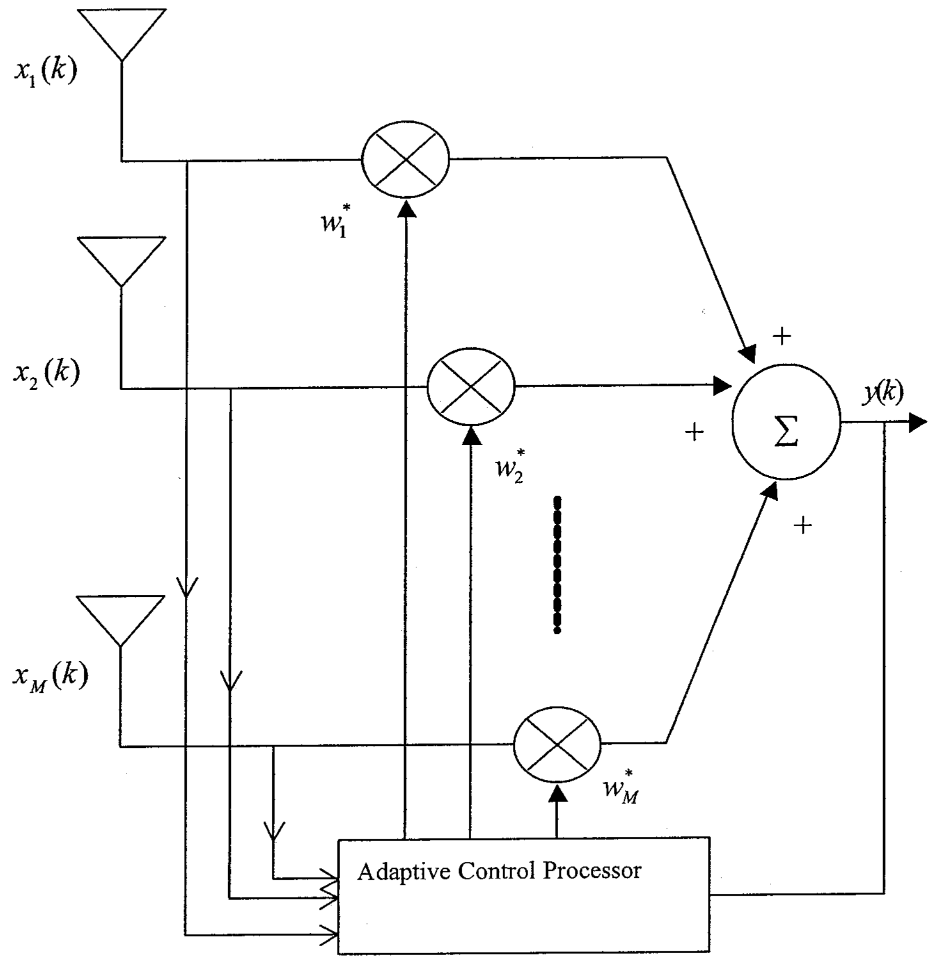


Figure 3.7 A simple narrowband adaptive array

For one adaptive array, there may exist several adaptive algorithms that could be used to adjust the weight vector. The choice of one algorithm over another is determined by various factors:

(i). Rate of convergence: This is defined as the number of iterations required for the algorithm, in response to stationary input, to converge to the optimum solution. A fast rate of convergence allows the algorithm to adapt rapidly to a stationary environment of unknown statistic.

(ii). Tracking: When an adaptive algorithm operates in a nonstationary environment, the algorithm is required to track statistical variations in the environment.

(iii) Robustness: In one context, robustness refers to the ability of the algorithm to operate satisfactorily with ill-conditioned input data. The term robustness is also used in the context of numerical behavior.

(iv) Computational requirements: Here the issues of concern include: the number of operations (i.e., multiplications, divisions, and additions/subtractions) required to make one complete iteration of the algorithm; the size of memory locations required to store the data and the program; the investment required to program the algorithm on a computer or a DSP processor.

3.6 Adaptive Algorithms

Adaptive beamforming algorithms may be categorized into two classes according to whether a training sequence is used or not. One class of these algorithms is known as the non-blind adaptive algorithm in which a training sequence is used to adjust the array weight vector. The second class is based on blind adaptive algorithms in which no training is needed

In a non-blind adaptive algorithm [15]-[18], [21]-[24], a training signal, $d(t)$, which is known to both the transmitter and receiver, is sent during the training period. The adaptive array in the receiver uses the training signal to compute the optimal weight vector, W_{opt}. After the training period, data is sent and the adaptive array uses the weight vector computed previously to process the received signal. If the channel and the interference characteristics remain constant from one training period until the next, the weight vector W_{opt} will contain the information of the channel and the interference, and their effect on the received signal will be compensated at the output of the array.

In a blind adaptive algorithm [15]-[18], a training signal is not required. Instead, some known properties of the desired received signal are used. Most of the blind algorithms may be categorized into the following three classes or combinations of them: algorithms based on estimation of the DOAs of the received signals [15]-[18], algorithms based on property-restore techniques and algorithms based on the discrete-alphabet structure of digital signals [21]-[22]. Constant modulus algorithm (CMA) and least squares constant modulus algorithm (LSCMA) belong to the algorithms based on property-restore techniques [15]-[18].

3.6.1 Least-Mean-Squares Algorithm

A significant feature of the LMS algorithm is its simplicity; it does not require measurements of the pertinent correlation functions, nor does it require matrix inversion. We can describe the LMS algorithm by the following equations [23]:

$$y(k) = w^H(k)x(k) \tag{3.44}$$

$$e(k) = d(k) - y(k) \tag{3.45}$$

$$w(k+1) = w(k) + \mu x(k)e^*(k), \tag{3.46}$$

where $e(k)$ is the prediction error, and μ is the step size which controls the convergence rate and stability of the algorithm.

The LMS algorithm is a member of a family of stochastic gradient algorithms since the instantaneous estimate of the gradient vector is a random vector that depends on the input data vector $x(k)$. The LMS algorithm requires about 2M complex multiplications per iteration, where M is the number of weights (elements) used in the adaptive array. The response of the LMS algorithm is determined by three principal factors: (1) the step-size parameter, (2) the number of weights, and (3) the eigen value of the correlation matrix of the input data vector.

3.6.2 Recursive Least squares Algorithm

Unlike the LMS algorithm which uses the method of steepest descent to update the weight vector, the recursive least squares (RLS) algorithm [23] uses the method of least squares to adjust the weight vector. In the method of least squares, we choose the weight vector $w(k)$, so as to minimize a cost function that consists of the sum of error squares over a time window. In the method of steepest-descent, on the other hand, we choose the weight vector to minimize the ensemble average of the error squares.

In the exponentially weighted RLS algorithm, at time k, the weight vector is chosen to minimize the cost function

$$\varepsilon(k) = \sum_{i=1}^{k} \lambda^{k-i} |e(i)|^2 , \tag{3.47}$$

where λ is a positive constant close to, but less than one, which determines how quickly the previous data are de-emphasized. In a stationary environment, however, λ should be equal to 1, since all past and present data should have equal weight. The RLS algorithm is obtained by minimizing the cost function in (3.47). The RLS algorithm can be described by the following equations

$$k(k)=\frac{\lambda^{-1}P(k-1)x(k)}{1+\lambda^{-1}x^{H}(k)P(k-1)x(k)} \tag{3.48}$$

$$\alpha(k)=d(k)-w^{H}(k-1)x(k) \tag{3.49}$$

$$w(k)=w(k-1)+k(k)\alpha^{*}(k) \tag{3.50}$$

$$P(k)=\lambda^{-1}P(k-1)-\lambda^{-1}k(k)x^{H}(k)P(k-1) \tag{3.51}$$

The initial value of $P(k)$ can be set to

$$P(0)=\sigma^{-1}I \tag{3.52}$$

where I is the $M\times M$ identity matrix, and σ is a small positive constant.

An important feature of the RLS algorithm is that it utilizes information contained in the input data, extending back to the instant of time when the algorithm is initiated. The resulting rate of convergence is therefore typically an order of magnitude faster than the simple LMS algorithm. This improvement in performance, however, is achieved at the expense of a large increase in computational complexity. The RLS algorithm requires $4M^2+4M+2$ complex multiplications per iteration[15]-[18], where M is the number of weights used in the adaptive array.

3.6.3 Constant Modulus Algorithm (CMA)

Some communication signals such as phase-shift keying (PSK), frequency-shift keying (FSK), and analog FM signals have a constant envelope. This constant envelope may be distorted when the signal is transmitted through the channel. The constant modulus algorithm (CMA)[15]-[18] adjusts the weight vector of the adaptive array to minimize the variation of the envelope at the output of the array.

The CMA minimizes the cost function

$$J(k) = E\left[\left||y(k)|^p - 1\right|^q\right]. \tag{3.53}$$

The convergence of the algorithm depends on the coefficients p and q in the equation above. Usually, the cost function J with $p = 1, q = 2$, or $p = 2, q = 2$ is used:

$$J(k) = E\left[\left||y(k)| - 1\right|^2\right]. \tag{3.54}$$

Using the method of steepest-descent, and replacing the gradient vector with its instantaneous estimate, we can update the weight vector by

$$w(k+1) = w(k) - \mu x(k)\left(y(k) - \frac{y(k)}{|y(k)|}\right)^*, \tag{3.55}$$

where μ is the step-size parameter. Now, we can describe the steepest-descent CMA (SD-CMA) by the following three equations

$$y(k) = w^H(k)x(k) \tag{3.56}$$

$$e(k) = y(k) - \frac{y(k)}{|y(k)|} \tag{3.57}$$

$$w(k+1) = w(k) - \mu x(k)e^*(k). \tag{3.58}$$

We see that the CMA is very similar to the LMS algorithm, and the term $\frac{y(k)}{|y(k)|}$ in CMA plays the same role as the desired signal $d(t)$ in the LMS algorithm. However, the reference signal $d(t)$ must be sent from the transmitter to the receiver and must be known for both if the LMS algorithm is used. The CMA algorithm does not require a reference signal to generate the error signal at the receiver.

In summary, since non-blind algorithms use a training signal, during the training period, data cannot be sent over the radio channel. This reduces the spectral efficiency of the system. However, blind algorithms use the noisy signal to form the weight vector of the antenna array. As a result, the BER (bit errors probability) of blind algorithms is poor compared to non-blind algorithms in general [15]-[18], [21]-[22].

In this thesis, we use LMS algorithm to update the weights of the antenna array.

3.7 Summary

In this chapter, some concepts in adaptive antenna array are discussed, including ULA, beamforming technique, beampattern, and adaptive algorithms.

In the next chapter, our simulation model and the new proposed MAC layer protocol using adaptive antenna array will be developed.

Chapter 4

System Model and Protocol Description

4.1 Introduction

In this chapter, our system model and protocol description will be defined. The chapter will be divided into two parts: section 4.2 will propose the system model and some preliminaries; section 4.3 will focus on the details of the proposed protocol description.

4.2 System Model and Preliminaries

In our development, we consider a simulation model where we have 10 stations randomly appearing in a 200meter $\times$ 200meter area, as shown in Figure 4.1.

We assume that station 1, 2, 3, 4, and 5 are stations that have data from upper layer to send. On the contrary, station 6, 7, 8, 9, and 10 are stations that receive data from other stations. Furthermore, we fix the combinations of source and destination stations during the simulation period. The combinations are station 1 transmitting to station 6, station 2 to station 7, station 3 to station 8, station 4 to station 9, and station 5 to station 10.

With different random seeds in Matlab, we can generate different scenarios, which means that stations are randomly distributed in space. Theoretically, the more random seeds we consider the more accurate simulation results we can get. However, due to limitation of time, we randomly pick up 25 random seeds, and get the average result over these scenarios.

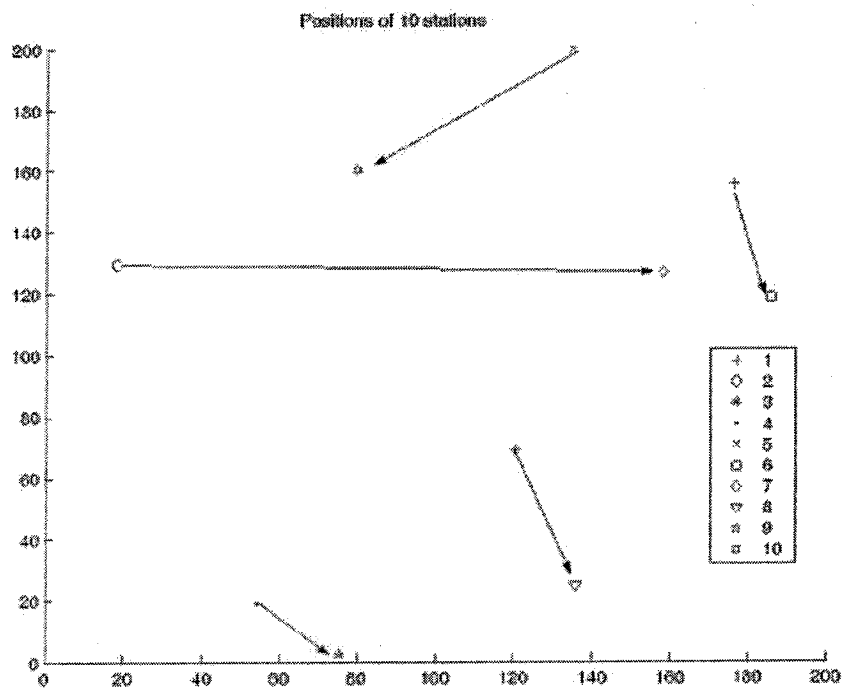


Figure 4.1 Snapshot of a 10-station network topology.

Similar to [4] and [5], we assume that each station is equipped with a Global Positioning System (GPS) to determine its position. Without loss of generality, and to enable the study of out-of-range problem, we assume that the radio range of each station to be 100 meters. All stations are assumed to be equipped with directional antennas, and these directional antennas are able to operate in two modes: omni and directional modes. In our protocol, the directional mode only used when a station transmits a data frame. The omni mode is always used when a station receives signals.

Furthermore, due to the power limitations of these stations, we assume that the radio range for both the omni and the directional modes of operation to be the same, which is 100 meters.

Different from the works in [2]-[5], we take the interference arising from the antenna array side lobes into consideration. This interference cannot be ignored, especially in situations where two receiving stations are close to each other, or when the number of active stations is large.

We assume that there are two FDMA channels to use. One channel is dedicated to control signals between the transmitting and receiving stations, referred to as the control channel. This channel is mainly used for the transmission of RTS, CTS, and ACK frames. The second channel, referred to as the data channel, is used for data transmission between stations.

One key parameter of the proposed MAC protocol is that RTS and CTS frames can be used to transmit positions and antenna weights information. The knowledge of real time positions and weights plays an important role in our protocol and simulation. Distinguishing from the simulations in [2]-[5], instead of only taking the destination station into consideration, we obtain the real time weights from the position information of all active stations in the radio range. Furthermore, we use the RTS and CTS to update the real time weights at every transmitting station during the transmission of data frames.

Before a station sends an RTS to initiate a transmission, it will send a Request for address (RFA) frame, which is newly defined in our protocol. The mission of RFA is to request the position information from stations that receive the frame. Upon receiving the RFA, stations use Request for address ACK (RFA_ACK) to carry the position information and send it back to the stations that sent RFA.

In our protocol, we also define a function called Virtual Carrier Sense (VCS). The goal of the VCS is to better check the status of the channel and return accurate channel status information. The VCS is a very important part in our proposed protocol since every transmitting station needs the result returned by the VCS to decide whether it can transmit frames or not. Therefore, from the whole system point of view, the introduction of VCS makes simultaneous transmissions better and smoother.

A VCS includes two parts: weight-training program and BER decision program. To better illustrate the VCS, let's use an example as shown in Figure 4.2.

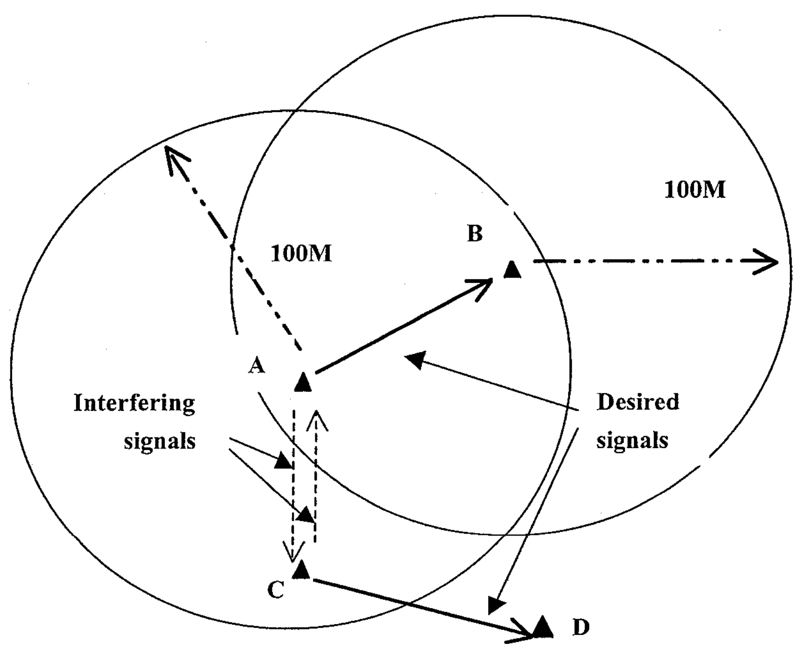


Figure 4.2 Illustration of virtual carrier sense function

Assume that station A is transmitting data to station B, and station C wants to transmit to station D shortly after station A begins its transmission. If station C has knowledge of positions of all stations, it will perform the VCS to check if the channel is idle for it to use.

Note that, although knowing that station A has access to the channel, station C still has a chance to transmit its data frame. The reason is that all transmitting stations in our model use adaptive antenna array to transmit data frames. Therefore, simultaneous transmissions over a single channel are possible, depending on the positions of transmitting and receiving stations.

In the first part of VCS, weight-training program, station C uses the position information to form a simulation model in its memory. In this off-line virtual model, station C assumes that station A^* and C^* (here, A^* and C^* are virtual stations in memory of station C) are using adaptive antenna array to transmit a 1,000-long training bits, and we chose the LMS as adaptive algorithm to obtain the antenna weights. Because the positions and the training bits are known by station C in advance, after the training period, station C gets the optimum weights of station A^* and C^* easily.

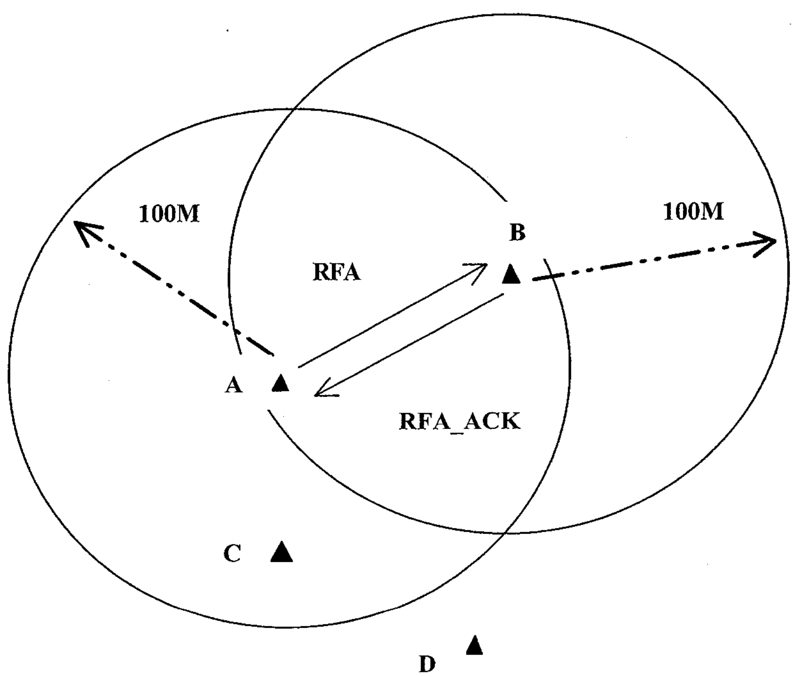
In the second part of VCS, BER decision program, station A^* and C^* transmit another 100,000-long (can be adjusted) training bits using these optimum weights. Also because positions and training bits are known in advance, station C obtains two BER values based on the optimum weights. Then, station C compares these two BERs to a pre-set BER_threshold (defined below). If and only if these two BERs are both lower than the BER_threshold, station C should consider the channel as idle; otherwise, station C considers that the channel is busy.

The goal of the pre-set BER_threshold is to provide a minimum quality of service (Qos) requirement that every transmission should meet. Usually, the pre-set BER_threshold is set by application programs in the upper layers. In our simulation, it is set to 10^{-5}.

In this thesis, we assume that the time of taking a VCS operation is small and can be ignored.

4.3 Protocol Description

The proposed MAC protocol works as follows.

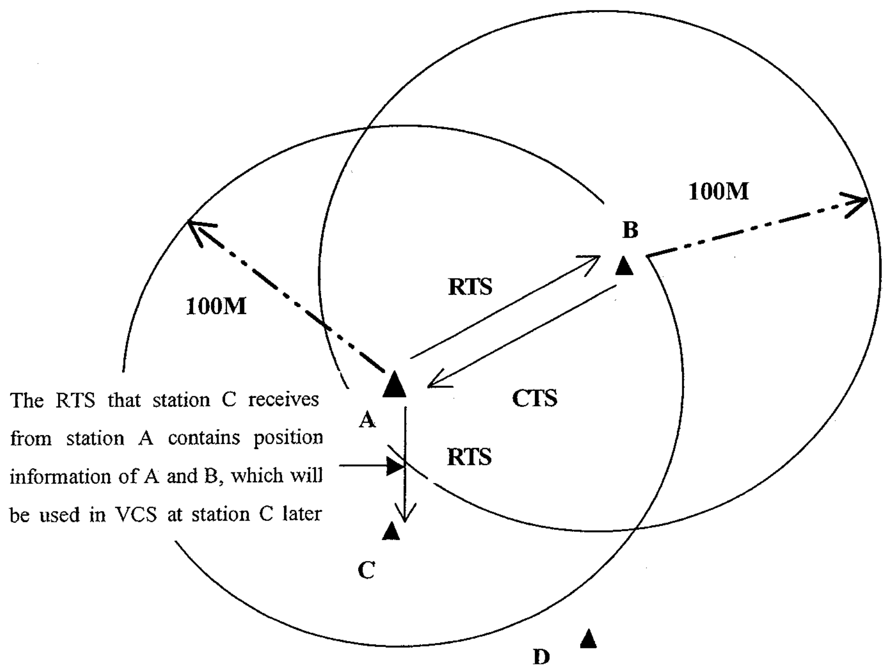


⟶Transmissions on the control channel ------>Transmissions on the data channel

Figure 4.3 Illustration of 4-station scenario I.

1), Considering the simple 4-station scenario in Figure 4.3, station A transmits to station B, and station C transmits to station D. Assuming that station A first accesses the control channel. Having obtained this access, station A sends an RFA using its omni mode (using a single antenna element) on the control channel. After receiving the RFA, station B answers with its position in an RFA-ACK frame.

2), Having received this information, station A writes the position information of A and B into its RTS frame, combined with other information (specified in the IEEE 802.11 standard), and sends it back to station B using the omni mode and on the control channel (see Figure 4.4).

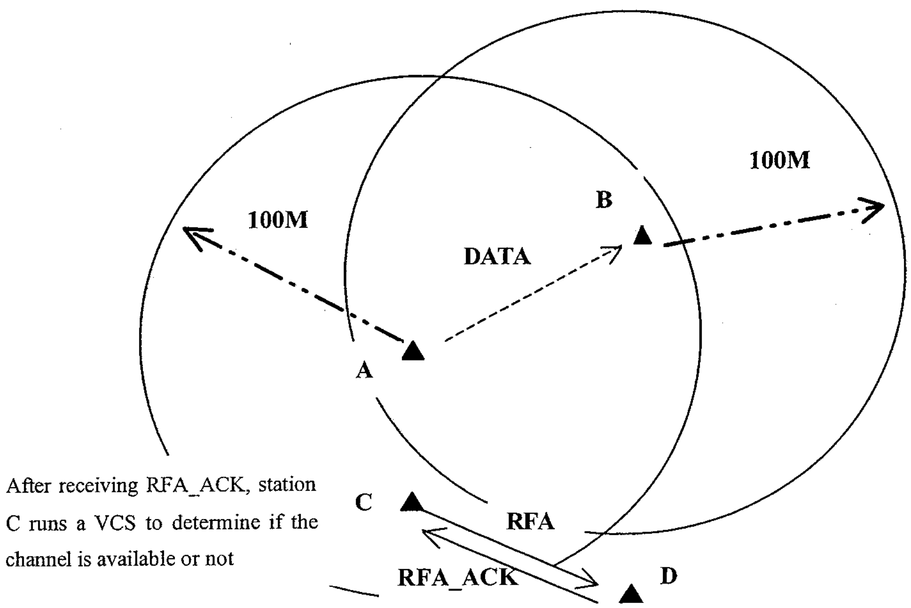


——>Transmissions on the control channel ------>Transmissions on the data channel

Figure 4.4 Illustration of 4 station-scenario II.

Upon receiving the RTS, station B also writes the position information of A and B into its CTS frame, and sends it back to station A. Since station A knows the position of B, after receiving the CTS, station A sends its data frame using the directional mode on the data channel.

3), When stations A and B finish transmission in the control channel (transmission between A and B on data channel is still in progress), the NAV of station C decreases to zero. If station C has data to send to station D, it then sends an RFA to D on the control channel, and D answers back with an RFA-ACK, as shown in Figure 4.5.

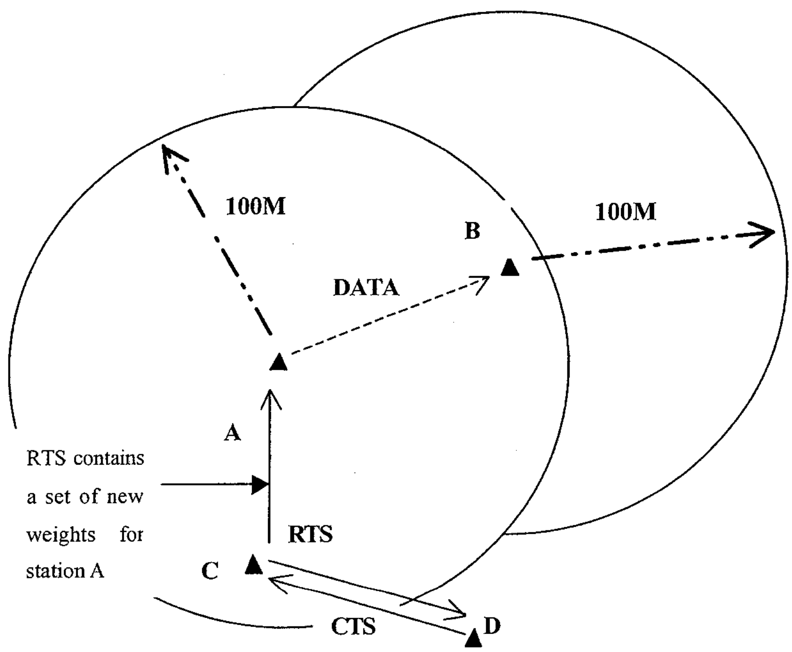


——> Transmissions on the control channel ------> Transmissions on the data channel

Figure 4.5 Illustration of 4 station-scenario III.

4), Note that, now station C knows the position information of the stations within it neighborhood (i.e., range) because it has read the position information in A's RTS and in D's RFA-ACK. With this knowledge, station C performs a VCS. Comparing with a Pre-set BER_threshold, station C will know whether to send a data frame using the directional mode on the data channel or not. If the VCS shows that the data frame transmitted from C may cause failure to ongoing transmissions (or station D could not receive data in a good condition) station C will keep silent.

If the VCS shows that the participation of station C and D doesn't interfere with any other transmissions, station C will keep the weights of C and A in its memory. Note that, in [3]-[6] the weights of antennas are fixed during transmissions. But in our protocol, the VCS procedure generates the best weights for station C and A based on the real time positions of the active users.



——→Transmissions on the control channel - - - - -→Transmissions on the data channel

Figure 4.6 Illustration of 4 station-scenario IV.

5), If station C passes the VCS, it writes the weights of station A into an RTS and sends the RTS using its omni mode. When station D receives the RTS, it sends a CTS back to station C, as shown in Figure 4.6. At the same time, station A also receives this RTS, which contains a set of new antenna weights of A. Station A immediately updates the old weights with the new ones, and then keeps transmitting data. Note, we previously assumed that this adaptation time is small and can be ignored.

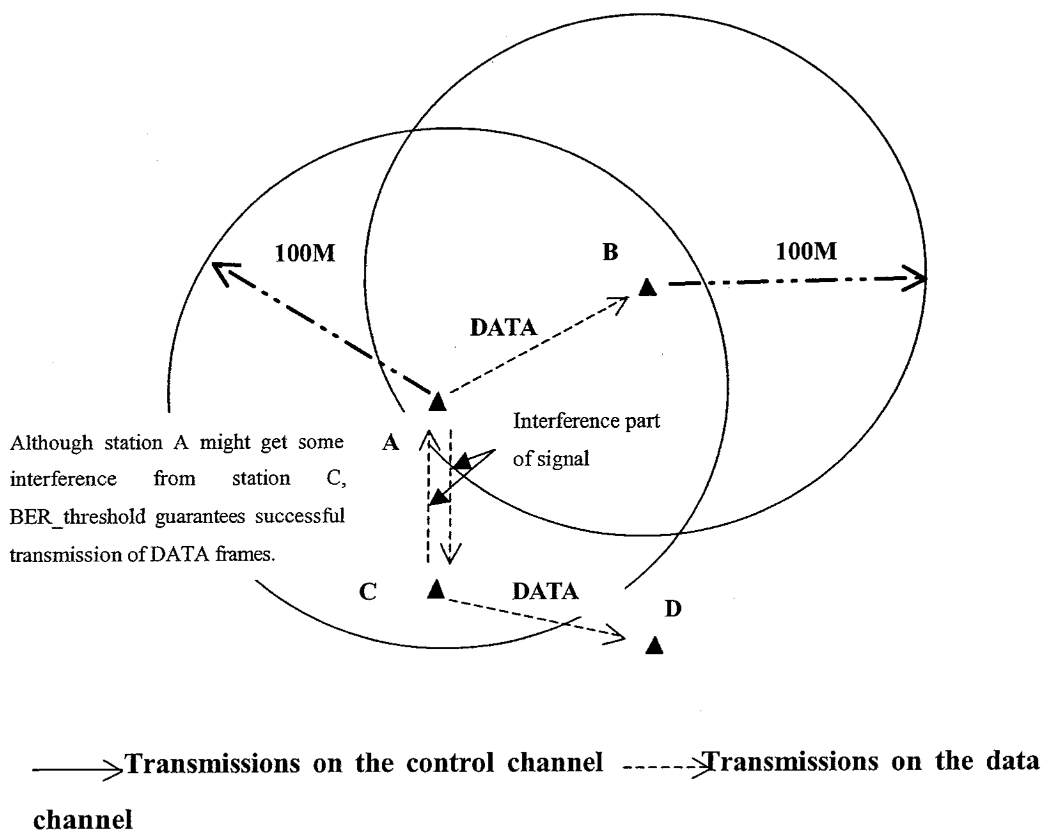


Figure 4.7 Illustration of 4 stations scenario V.

6), Finally, along with station A, station C sends the data frame using the directional mode on the data channel, as shown in Figure 4.7. The BERs of the received data at stations B and D are guaranteed by the value of the Pre-set BER threshold.

In order to illustrate the operation of the proposed protocol, a flow chart is shown in Figure 4.8. It is a flow chart of a 4-station scenario that is slightly different than the scenario shown in Figure 4.3-4.7. In Figure 4.8, 4 stations E, F, G, and H are distributed on a straight line from left to right. The distance between every neighboring station is 100M. Furthermore, we assume that the radio range of each station is 100M. That means every station can communicate with its nearest neighbors only. Station E transmits to station F, and station G transmits to station H.

If we use the IEEE 802.11 MAC layer protocol in this scenario, after receiving an RTS from station E, station F sends a CTS back to station E. However, not only station E but also station G can hear this CTS. As a result, station G has to keep silent during the transmission between E and F according to the protocol.

In our protocol, as shown in Figure 4.8, when station E and F finish their transmission on the control channel, station G can use the control channel to transmit control frames. Besides that, if it passes the VCS, it can transmit data frame simultaneously with station E on the data channel. Therefore, the throughput performance of the overall system is increased when our protocol is used.

4.4 Summary

We introduced our system model and some preliminaries, and further developed the details of the proposed protocol. In the next chapter, simulation results and discussions using our proposed protocol will be given.

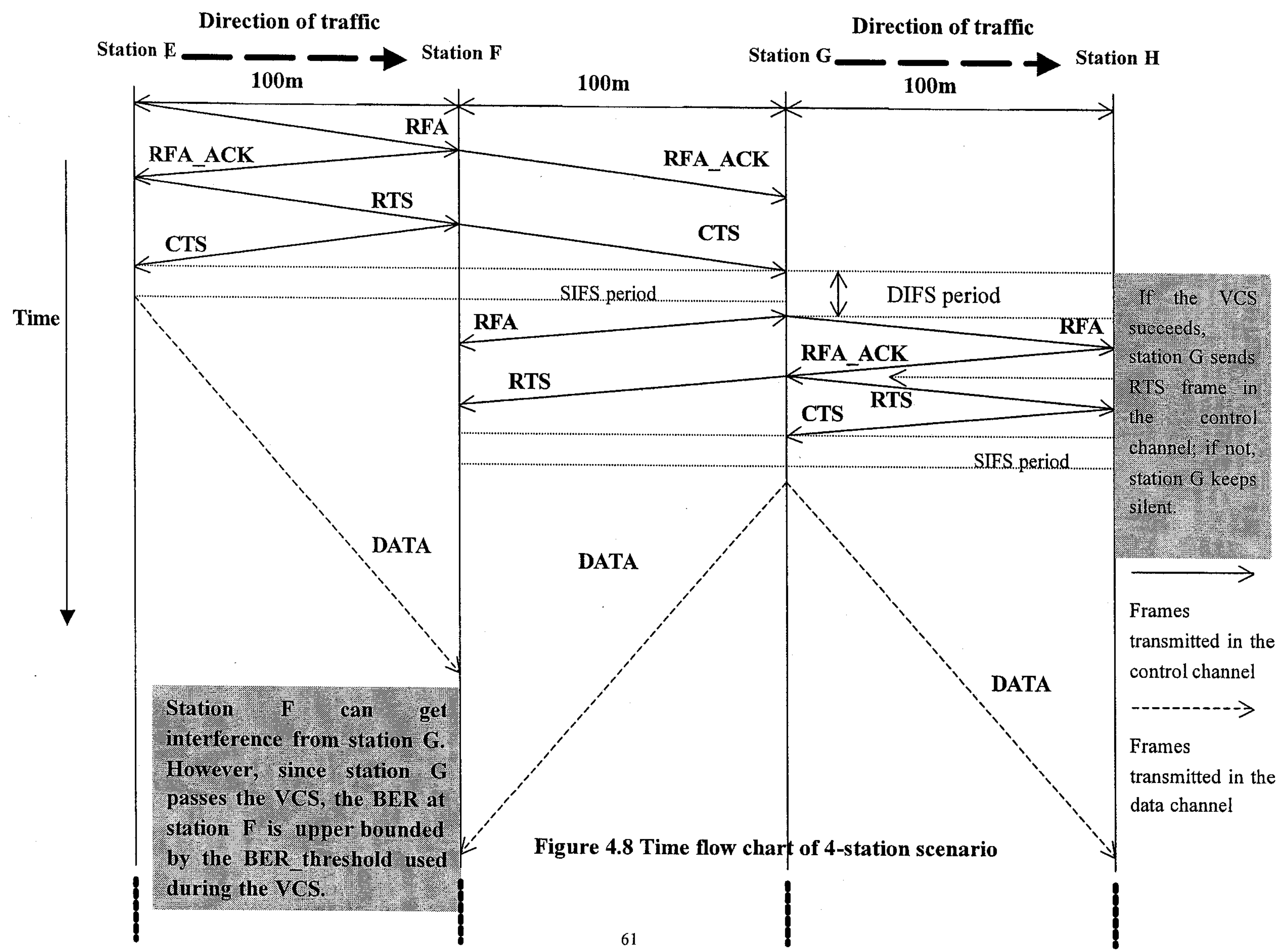


Figure 4.8 Time flow chart of 4-station scenario

Chapter 5

Simulation Results and Discussion

5.1 Introduction

In this chapter, simulation results and relevant discussions are presented. Section 5.2 includes simulation parameters. Section 5.3 provides a beampattern comparison between DMAC and proposed protocol. In Section 5.4 and 5.5, simulation results using random and two specified topologies will be shown.

5.2 Simulation Parameters

In the following two sections, we consider an ad hoc network with randomly generated topology reflecting the random movement of 10 active stations in a $200M \times 200M$ area. A snapshot of this topology is depicted in Figure 4.1. Based on this snapshot, station 7 is out-of-range relative to station 2 (distance between stations>100M). The remaining simulation parameters are listed as follows:

- Without loss of generality, we consider stations 1,2,3,4, and 5 to be the transmitting stations and stations 6-10 as receiving ones.
- As shown in Figure 4.1, we consider a transmission scenario as follows, station $1 \rightarrow 6$, station $2 \rightarrow 7$, station $3 \rightarrow 8$, station $4 \rightarrow 9$, and station $5 \rightarrow 10$.

- We randomly generate 25 different topologies, and evaluate the average throughput over these 25 network topologies.
- The transmission range of all stations in both the omni and directional modes is fixed to 100M.
- The transmission rate is taken to be 1Mbps.
- The transmitted frame length is 4000bits, and for each scenario, the simulation time is equal to 1 second.
- The length of both the RFA and RFA-ACK frames is set to 20 bits.
- The throughput is defined as the total system throughput, which is the number of all successful received frames (control frames are not included).
- As a benchmark, we compare the performance of the proposed protocol with the DMAC [5] and IEEE 802.11[1].

Since the main focus of this work is to investigate the performance of the proposed protocol using directional antennas, issues related to the fairness of the IEEE 802.11 are not treated in this thesis. Therefore, all frames from the upper layers are stored in buffers (for each station) at the beginning of the simulation time.

As mentioned earlier, we employ a simple single-hop routing protocol. In this routing protocol, if the destination node is out-of-range of the source node, the transmission can only take place through a single hop (i.e., through one intermediate node within the range of the two nodes). Note that, one can employ more efficient routing protocols to solve the out-of-range problem. However, here we chose to use such a simple routing protocol to simplify the simulation model. Also to allow for a fair comparison, the same routing protocol is also incorporated into both the DMAC and the IEEE 802.11.

In simulating the DMAC, the definition of the DNAV angle (same as sub-channel angle) is shown in Figure 5.2. We assume that the effective angle is the angle of the main lobe of a beampattern when a certain number of antennas are used. Therefore, DNAVs are fixed for every station when the number of antennas is fixed. Then based on this effective angle, we define the angle of DNAV as half of the effective angle. When the angle of each DNAV is determined, the total number of DNAV is fixed as well.

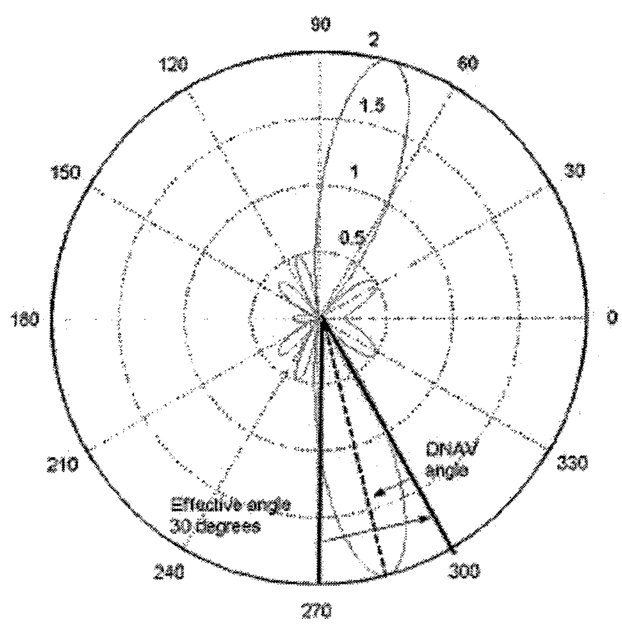


Figure 5.1 Definition of effective angle and DNAV angle. Num_of_antenna=5

For example, as shown in Figure 5.2, the effective angle is 30°, when using 5 antennas. So, the angle of each DNAV is equal to half the effective angle, which is 15°. Hence, in a 360°-degree-circle, the original NAV defined in the IEEE 802.11

standard is divided into $360^\circ/15^\circ$, 24 DNAVs. Furthermore, in this thesis, the distribution of these DNAVs is fixed during the simulation. That means DNAV_1 will indicate the idle/busy status of the sub_channel_1, which is from 0° degree to 15° degree; DNAV_2 will indicate the status of the sub_channel_2, which is from 15° degree to 30° degree, and so on.

Instead of occupying two sub-channels (effective angle is 30°), a transmission should occupy three sub-channels since the angle of destination is random (not to be 0°, 15°, or 30° exactly). For example, if the destination station is at the 21° degree of source station, a transmission from source to destination will occupy sub-channel 1, 2, and 3, and it will write three DNAV tables (to be busy) of other stations in its radio range. In this case, the spatial bandwidth of a sub-channel, which is 15° bandwidth wasting. Based on this model, one can see that no matter how the signal-to-noise ratio (SNR) is increased, the throughput stays at a maximum limited value. This will later be demonstrated when we present the simulation results and comparisons of the proposed two-channel MAC protocol compared to the standard IEEE 802.11 and DMAC protocol.

5.3 Beampatterns in DMAC and Proposed Protocol

As mentioned before, in our protocol, when a station wants to transmit data, it not only performs a physical carrier sense [1], but also a VCS defined in section 4.2. After finishing the VCS, the station obtains: (i) the weights of all active transmitting stations; (ii) the signal that indicates whether it is allowed to access the channel or not based on these weights.

We know from [2]-[5], that the weights of the antenna array of the transmitting station are only determined by the position of the receiving station. In this thesis, we call this type of weight-training method: separate training, because it doesn't include interference stations during the weight-training process. For example, as shown in Figure 4.1, the weights of station 1 are only determined by the position of station 6.

However, in our weight-training process, which is defined as together training, instead of only taking the transmitting station into consideration, we get the weights from the position information of all active stations in the radio range (including interference stations), which is more accurate. Also in Figure 4.1, the weights of station 1 might be determined not only by the position of station 6 but also by the position of station 2 and 7, if station 7 is in the range of station 1 and station 2 keeps transmitting to station 7. As a result, we have two different types of beampatterns due to two different types of weight-training methods.

Two beampatterns used in [2]-[5] and our proposed protocols are shown below respectively, when using the same topology shown in Figure 4.1. In the first step, we take two pairs of the stations into consideration, which are station 1 to 6 and station 2 to 7, when we form the beampatterns using separate training and together training. Furthermore, in order to better study the effect of interference on the beampatterns, we make an assumption that there is no more radio range limit when a source station transmits signals to its destination station (still, only among these four stations). Thus, we get Figure 5.2 and 5.3.

From Figure 4.1, we can measure that station 6 and 7 are at 286° and 230° of station 1. Therefore, 286° is the direction of desired signal, and 230° is the direction of the interfering signal to station 7. The main lobes of two transmitting beampatterns shown in Figure 5.2 and 5.3 both point towards the direction of 286° as well.

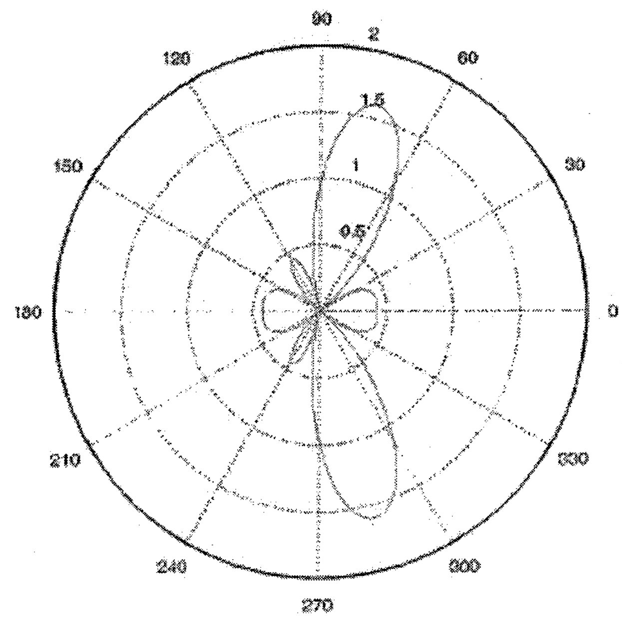


Figure 5.2 Beampattern of station 1, using separate training in DMAC, separate training beampattern of station1, Num_of_antenna=4, training_pair=2, angle_of_training: [286° 230°]

A distinct difference between these two beampatterns is that in together training, we eliminate the interference gains toward 330°-30° and 150°-210° compared to separate training. Furthermore, the maximum amplitude of the desired signal toward 286° is 2.8 compared to 1.6 in separate training. Hence, the beampattern obtained from together training is obviously better than the one from separate training. This makes sense since together training uses real time information (positions of all active stations) as opposed to separate training.

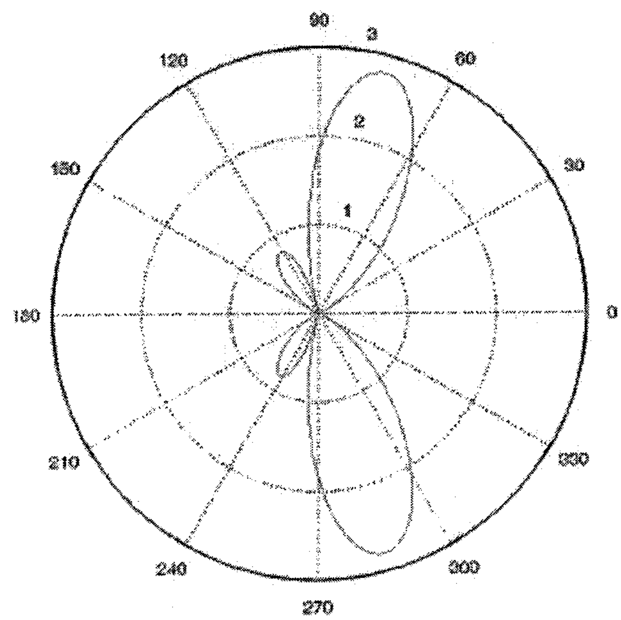


Figure 5.3 Beampattern of station 1, using together training in our protocol together training beampattern of station1, Num_of_antenna=4, training_pair=2, angle_of_training: [286° 230°]

Now, let us further study the BER performance of these two training methods without noise. We use a simple simulation model based on Figure 4.1. In this model, we consider an extreme case that there is no limit on the radio range of every station. For example, the signals of station 1 can not only be heard by station 6, but also be heard by other stations like station 7, 8, 9, and 10; station 2 can be heard not only by its destination station 7, but also by station 6, 8, 9, and 10, as so on. Therefore, each receiving station (station 6-10) will receive one desired signal plus four undesired interfering signals. Now, we calculate the BER at each receiving station, and get the average BER as shown in Figure 5.4.

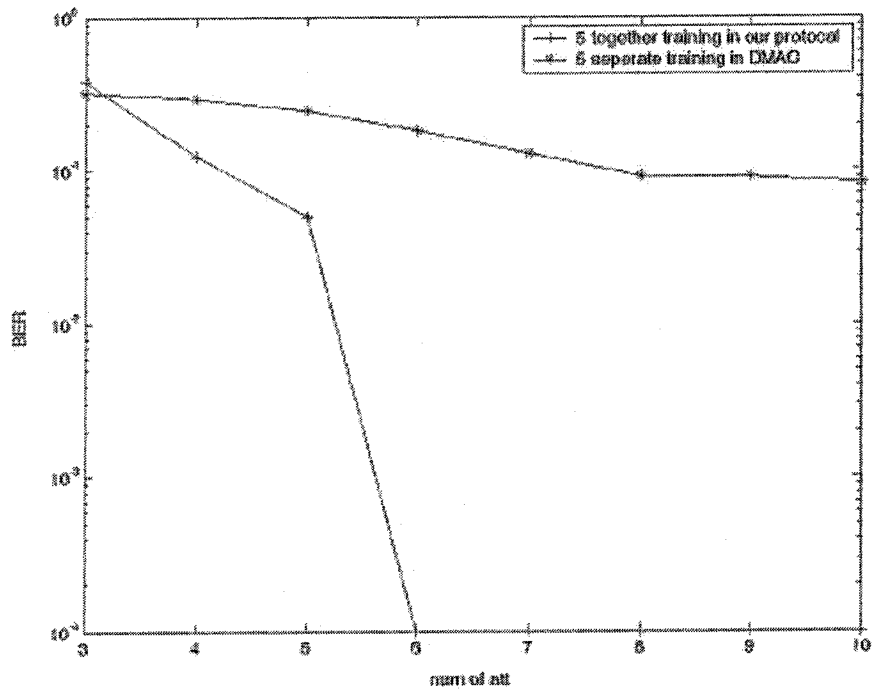


Figure 5.4 BER comparison between separate training and together training at different number of antennas (without noise)

From Figure 5.4, we can see very clearly the large performance gap between these two training methods. When separate training is used, the BER performance stays between 10^0-10^{-1} no matter how much we increase the number of antennas. On the other hand, the BER is lower than 10^{-4} when 6 antennas are used as shown in Figure 5.4. Also we can see that, in the together training case, the BER performance connects to the number of antenna very closely. Although this figure shows that together training is more superior to separate training only in the topology shown in Figure 4.1, in general, it is the same case based on the simulation results that will be presented later on. The conclusion we make here is that if we have the knowledge of where the other stations are, then we can use this knowledge to improve the system performance. As shown in section 4.2, the new created frame RFA, RTS, and CTS do the job of exchange of position information.

5.4 System Performance Evaluation using Random Topology

For the random scenarios, we study the average system throughput for two different load distribution cases. In the first case, we consider a network where all participating stations are transmitting at a same fixed load (from upper layers), and the load can be adjusted from 0.1Mbps to 1Mbps, as shown in Figure5.5.

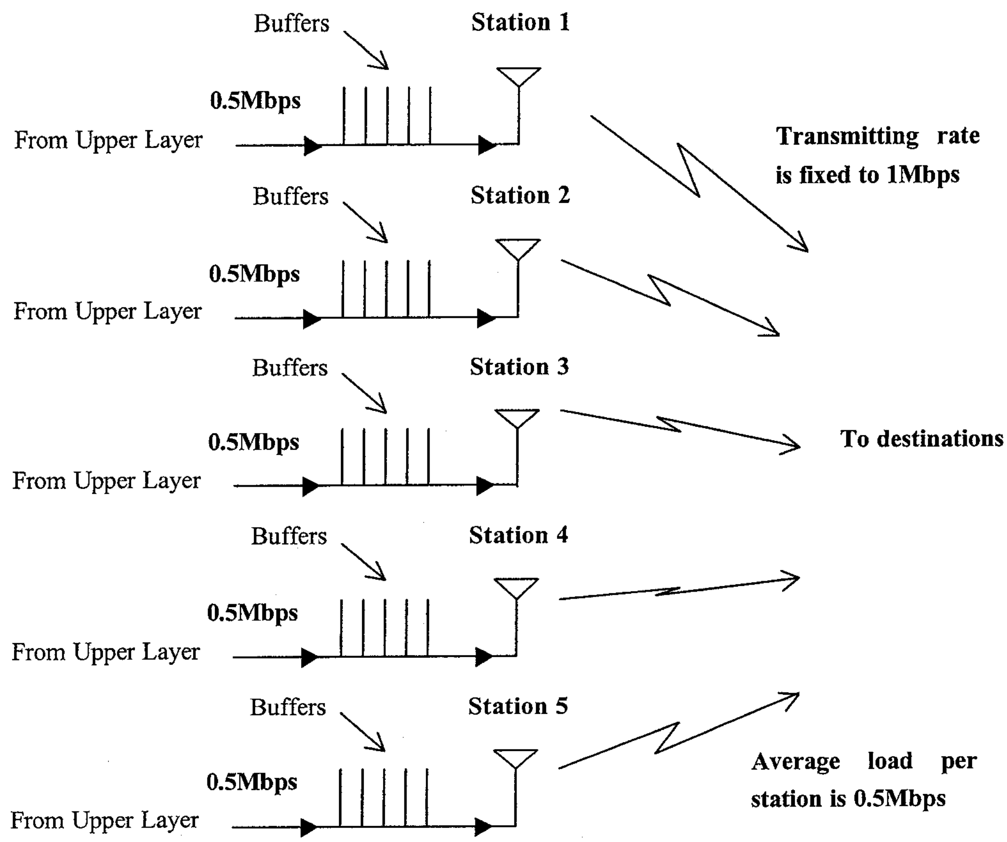


Figure 5.5 Fixed load case illustrations

In the second case, shown in Figure5.6, we consider a scenario in which the transmitting stations send their data according to a uniform distribution at different transmission rates. That is we set a simple uniform load pattern here. For example, if the mean of uniform distribution load per station is 0.5Mbps, the loads over five transmitting station 1, 2, 3, 4, and 5 will be 0.3, 0.4, 0.5, 0.6 and 0.7Mbps respectively.

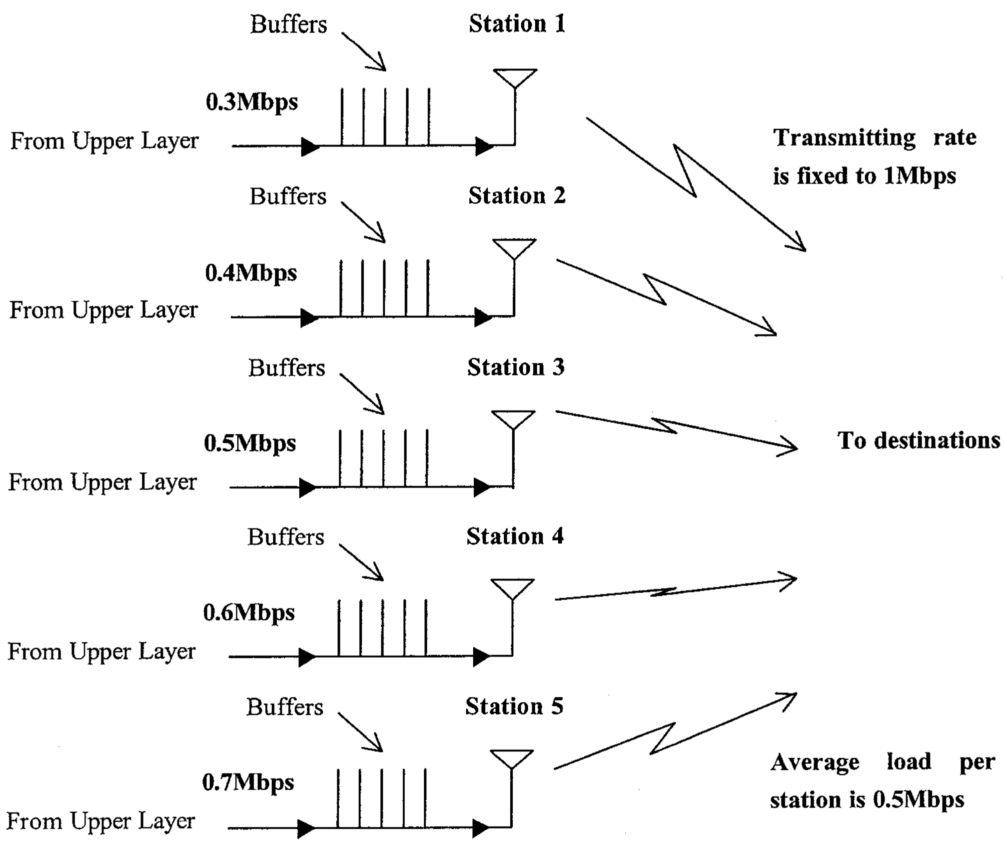


Figure 5.6 Uniform distribution load case illustrations

Finishing the load distribution models, we will present the system throughput using different protocols (the IEEE 802.11 standard, DMAC [5], and proposed MAC protocol) at different situations, which include different number of antennas, different SNRs, different topologies, and different load patterns.

Figure 5.7 shows results for the total achieved throughput at different fixed loads (per station). The advantage of the proposed protocol, over DMAC and IEEE 802.11 of enabling simultaneous transmissions between stations and avoiding the hidden terminal problem is quite clear in highly loaded systems.

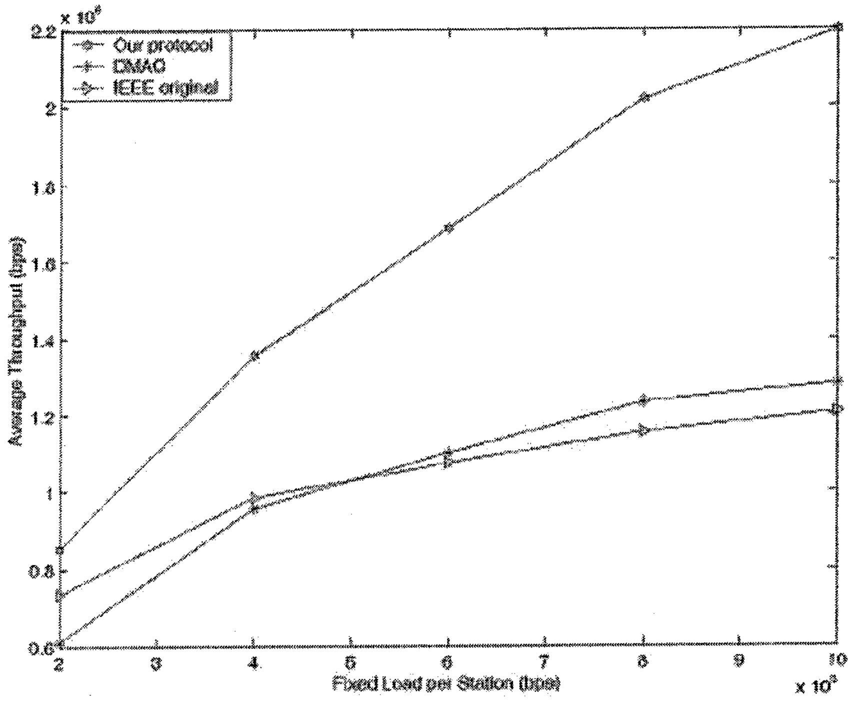


Figure 5.7 Average throughput of the proposed MAC protocol, DMAC, and IEEE 802.11 at different fixed loads (per station) for 25 random topologies. SNR=20dB, frame_length=4000 bits, Num_of_antenna=5 (not applied to IEEE 802.11).

A second remark from these results is that all three protocols perform similarly when the assigned load-per-station is light. The reason is that in both DMAC and IEEE 802.11 when the load-per-station is light, and even though some stations have to wait until the channel is clear, they still have time to finish their transmissions. On the other hand, when the load-per-station becomes large, those waiting stations will not have enough time to finish their transmissions. Note that our protocol, with its larger simultaneous transmissions and robustness against the hidden-terminal problem, will be able to perform better than the other two protocols especially in heavily loaded systems.

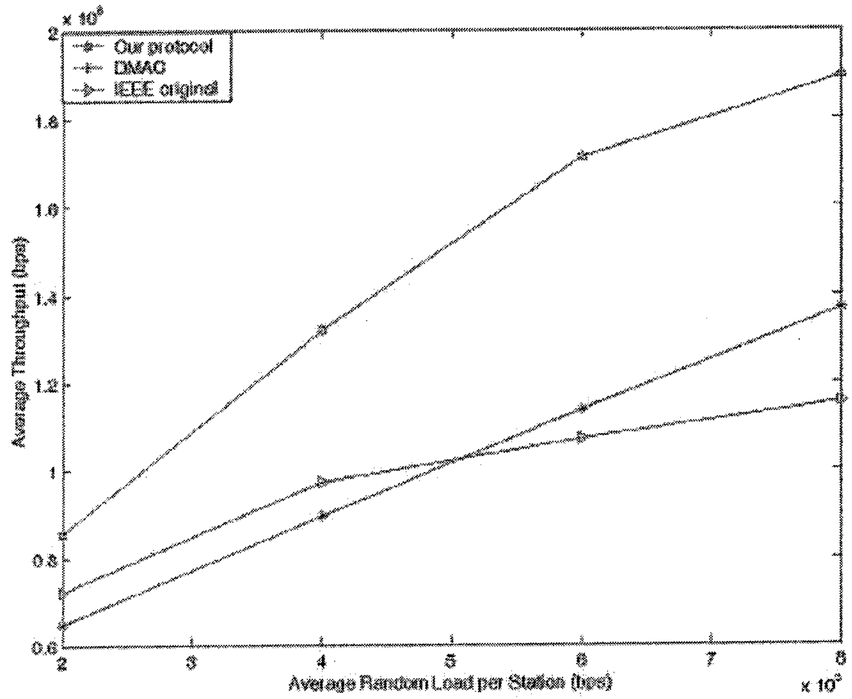


Figure 5.8 Average throughput of the proposed MAC protocol, DMAC, and IEEE 802.11 at different uniform distribution loads (per station) for 25 random topologies. SNR=20dB, frame_length=4000 bits, Num_of_antenna=5 (not applied to IEEE 802.11).

In Figure 5.8, we see that when the load per station is random, the performance of the whole system slightly degrades compared to the results in Figure 5.7. The reason is that in some scenario, when stations with larger loads are close to each other, local traffic jam may occur. As a result, these stations may not have enough time to finish their transmissions. Therefore, we can conclude that the average fixed load pattern will offer a larger throughput gain than the case of uniform distribution loads.

In Figures 5.9 and 5.10, the number of antennas decreases to 3. The performance of the system is almost the same in both fixed load and uniform distribution load case. However, we notice that in fixed load case, when the load per station is 1Mbps, the 5-antenna system offers 0.2Mbps higher throughput than the 3-antenna case.

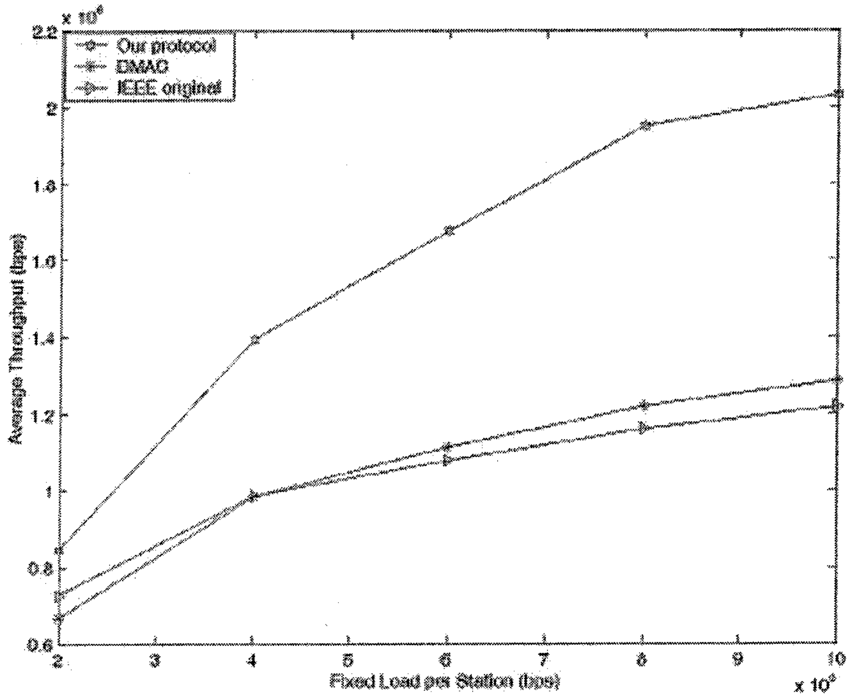


Figure 5.9 Average throughput of the proposed MAC protocol, DMAC, and IEEE 802.11 at different fixed loads (per station) for 25 random topologies. SNR=20dB, frame_length=4000 bits, Num_of_antenna=3 (not applied to the IEEE 802.11).

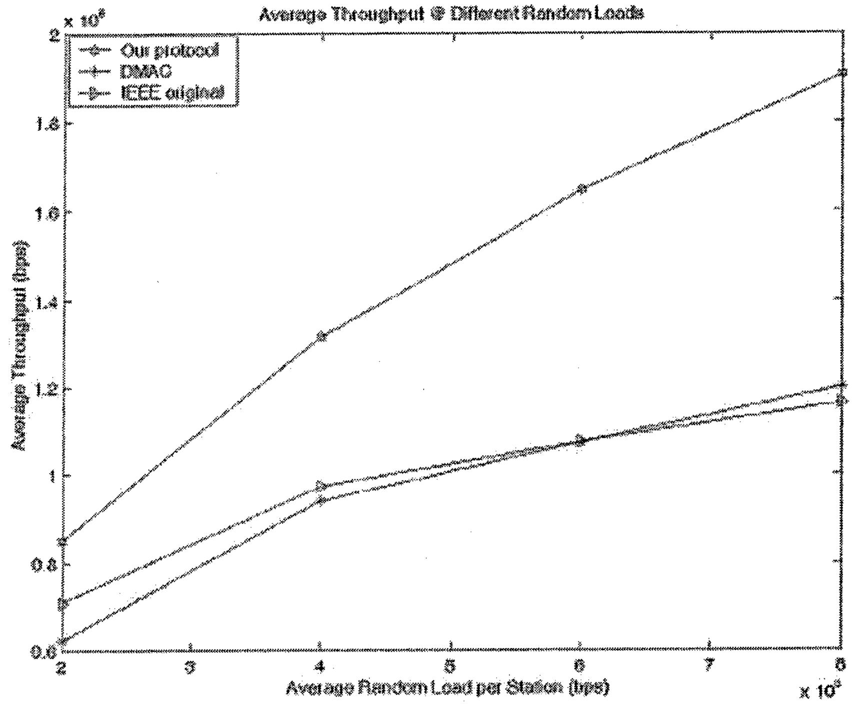


Figure 5.10 Average throughput of the proposed MAC protocol, DMAC, and IEEE 802.11 at different average uniform distribution load (per station) for 25 random topologies. SNR=20dB, frame_length=4000 bits, Num_of_antenna=3 (not applied to IEEE 802.11).

The same conclusion can be drawn for the DMAC protocol. In general, we notice that the number of antennas plays an important role in improving the total system throughput for the proposed MAC protocol. Also, based on our results, we foresee that the more antennas we use the larger the gap we get between our proposed protocol and DMAC protocol.

5.5 System Performance Evaluation using two Fixed Topology

In order to take an insight look of the proposed two-channel protocol, the throughput performance of a particular topology (shown in Figure 4.1) is studied as follows.

We will present the system throughput using different protocols (the IEEE 802.11 standard, DMAC [5], and proposed MAC protocol) at different situations, which include different number of antennas, different SNRs, and different load patterns.

Here we study the performance of the network topology in Figure 4.1. As shown in Figure 4.1, every receiving station is in the range of its corresponding transmitting station except for station 7 (the limit of radio range is 100M). Given this topology, we study the performance of the network in terms of the total throughput. In Figure 5.11, we compare the throughput performance versus the signal-to-noise ratio (SNR) of the proposed two-channel MAC protocol with IEEE 802.11 and DMAC.

Our proposed protocol offers a substantial throughput gain compared to the other protocols. This large throughput gain is mainly due to the fact that our protocol allows more simultaneous transmissions during the simulation time. For example, DMAC and IEEE 802.11 allow two pairs of stations to transmit at a time based on the topology shown in Figure 4.1. Therefore the maximum throughput is less than 2Mbps under full load condition (meaning load per station=1Mbps, which is the same as the transmitting rate, 1Mbps). On the other hand, our protocol allows four pairs of stations to transmit simultaneously yielding to 3.6Mbps throughput under the same condition. We also notice that no matter how much the SNR is increased (beyond 10dB), the throughput of DMAC stays at 1.8Mps. The reason behind this is that using the DNAV to identify whether the channel is available for transmission or not is not accurate compared to our VCS. In addition to the DNAV problem, the unheard RTS/CTS problem also limits the performance of DMAC.

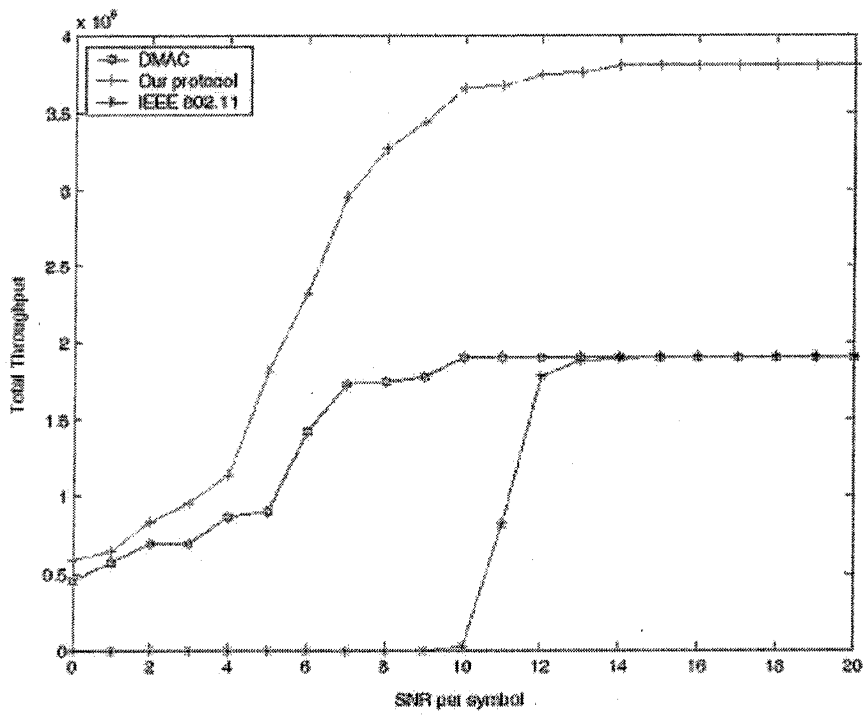


Figure 5.11 Performance comparison based on the topology in Figure 4.1, load per station=1Mbps, Num_of_antenna=5

Next, we examine the effect of varying the number of antenna elements on the system throughput at a fixed SNR of 10dB and 1Mbps load per station. This is shown in Figure 5.12 using the topology shown in Figure 4.1.

We can see that both the proposed protocol and DMAC nearly achieve the same throughput when the number of antennas is less than 5 elements. As opposed to DMAC where the maximum achieved throughput is reached using 4 antenna elements, our results show that increasing the number of antennas using the proposed protocol can significantly improve the overall system. This is simply due to the advantage of using the VCS that forms more accurate weights than DMAC, leading to a 3.6Mbps throughput achieved with 5 antenna elements. The relatively poor 1.8Mbps maximum throughput of DMAC is due to the bandwidth-waste problem of the DNAV discussed earlier in section 5.2, and the limited number of simultaneous transmission (two pairs in this case).

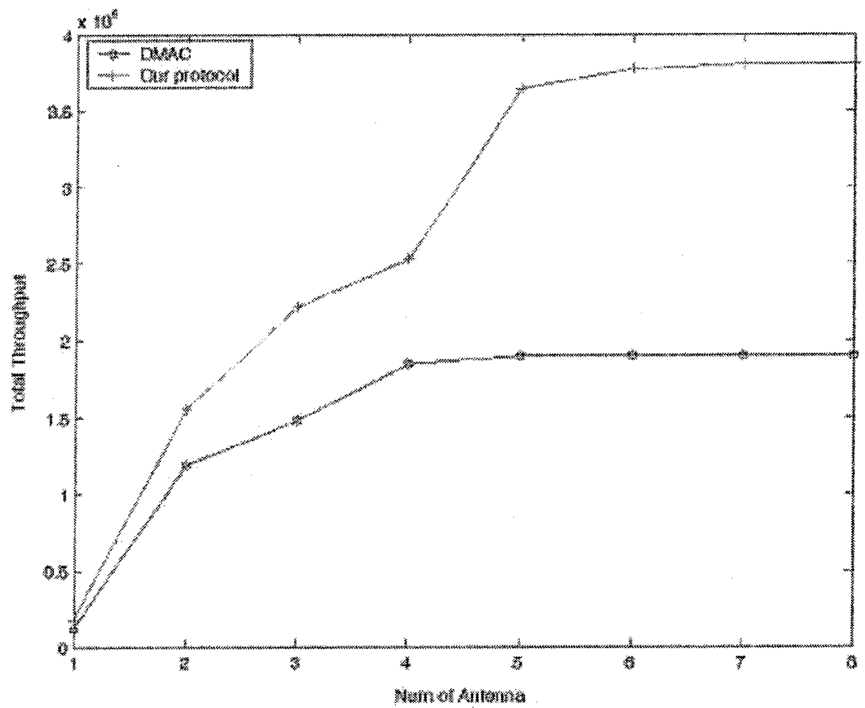


Figure 5.12 Effect of number of antennas on the total throughput based on Figure 4.1, SNR=10dB, load per station=1Mbps.

We notice that due to our definition of throughput, the maximum system throughput, cannot be multiple of the transmitting rate, 1Mbps, even if stations transmit and receive smoothly. The reason is that control frames are not counted into the system throughput. Besides that, the waiting periods like SIFS and DIFS [1] also affect the system throughput. Therefore, the simulation results are usually multiples of 0.9Mbps, 3.6Mbps in our proposed protocol and 1.8Mbps in the DMAC. Since the transmission rate is fixed to 1Mbps, the maximum traffic throughput should be 4Mbps. According to the definition of throughput, the 3.6Mbps throughput is counted without any control frames. Therefore, a 0.4Mbps or 0.1Mbps per station bandwidth is used for control frames and waiting periods like SIFS.

As a final study on the topology of Figure 4.1, Figure 5.13 shows the results for the total achieved throughput at different loads (per station).

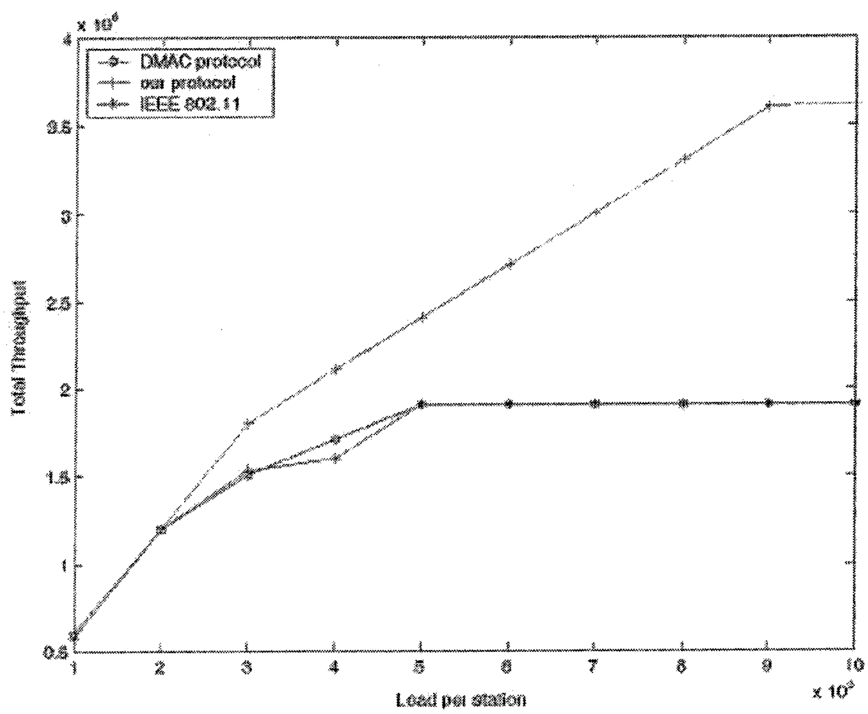


Figure 5.13 System throughput comparison at different loads based on Figure 4.1, SNR=20dB, Num_of_antenna=5.

Comparing Figure 5.13 with Figure 5.7, we see the difference is that under full load condition, the maximum throughput in Figure 5.13 is around 3.6Mbps; whereas in Figure 5.7 is only around 2.2Mbps.

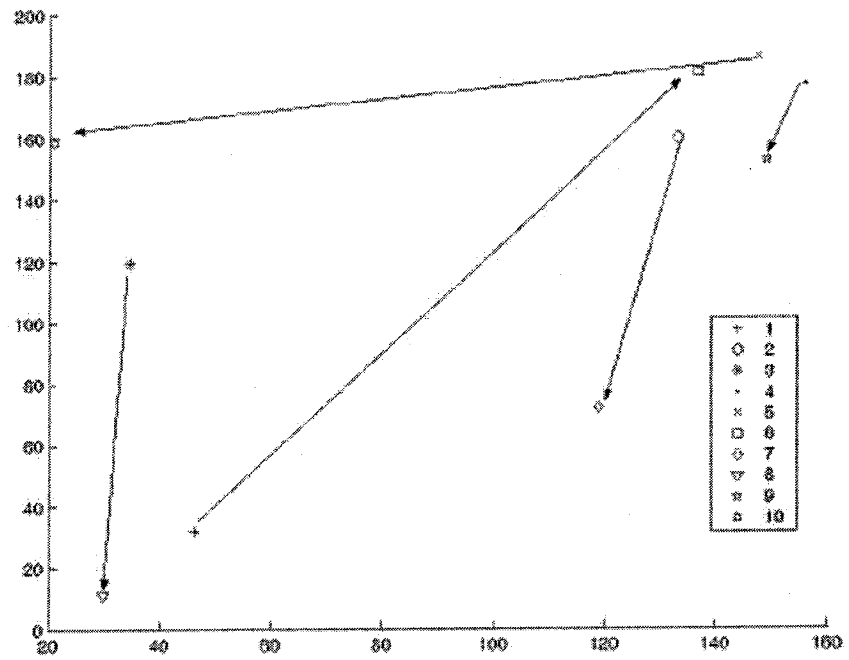


Figure 5.14 Snapshot of another 10-station network topology.

Considering another random positions scenario as in Figure 5.14, only two pairs of stations are in the transmission range (station 2 to 7 and station 4 to 9). Based on this random topology, Figure 5.15 shows throughput results at different system loads, a SNR equal 20dB and using 3-element array.

As seen in Figure 5.15, the maximum throughput of the IEEE 802.11 and the DMAC under full load conditions are almost the same, which is approximately 0.9Mbps. In other words, only one pair of stations is transmitting during the simulation period. On the other hand, the maximum throughput achieved using our protocol is around 1.9Mbps. This throughput can be simply justified since in this case, two simultaneous transmissions take place during the simulation period.

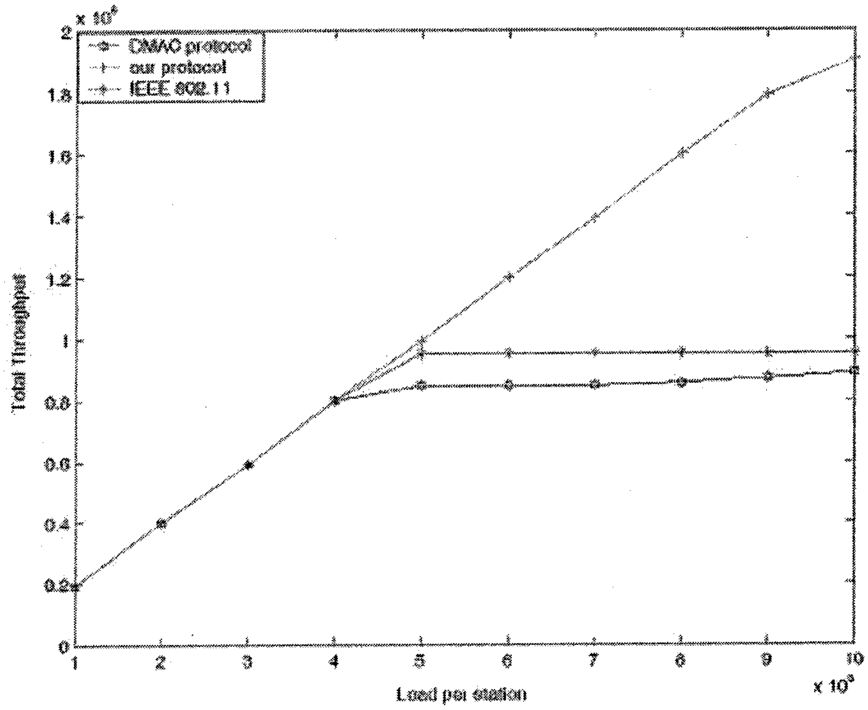


Figure 5.15 Total throughput comparison at different loads based on Figure 5.14, SNR=20dB, Num_of_antenna=3.

Based on our investigations, we have obtained more results for different (random) network topologies where all results confirm the superiority of our two-channel protocol over both IEEE 802.11 and the DMAC.

5.6 Summary

In this chapter, we presented simulation parameters, beampattern for DMAC and proposed protocol, and system performance evaluations of the IEEE 802.11 standard, DMAC and proposed protocol.

In next chapter, a conclusion of the thesis and future work will be provided.

Chapter 6

Conclusion and future work

6.1 Conclusion

In this thesis, we proposed a two FDMA channel MAC layer protocol for use in ad hoc networks equipped with directional antennas. In what follows, we state our conclusions.

(i). The hidden-terminal problem that is common in ad hoc networks, when using adaptive antenna arrays, can be avoided if the position information of the interfering users is known before transmission (the new frame RFA does this job).

(ii). In order to obtain better weights on transmission stations, position information can also be used to form new sets of weights, and these weights give better BER performance than the case where we don't have prior information of positions.

(iii). The results have clearly shown that large throughput gains can be achieved using the proposed two-channel protocol compared to the DMAC and the IEEE 802.11 protocols.

(iv). By studying 25 random topologies and two specified fixed topology, we also noticed that the performance of the system depends mainly on the topology of the network.

(v). We have shown that using a simple one-hop routing protocol, as a solution to the out-of-range problem, with our two-channel MAC protocol can bring the total throughput to higher levels.

(vi). Compared to the DMAC and the IEEE 802.11 protocols, although the proposed protocol requires more bandwidth (two FDMA channels) it has the potential to obtain large throughput gains when N data channels are further introduced in our scenario.

6.2 Future Work

In the performance evaluation section, we show the average result of 25 random cases. However, in order to get a more accurate average result, a larger number of random cases should be used in our future simulations. Besides that, instead of using our simple routing protocol, having a good routing protocol is also very necessary when handling the out-of-range problem. We further speculate that using a good routing protocol will improve our system throughput significantly when many data are needed to forward more than one hop before getting to their destination.

We use two channels in our protocol, one control channel and one data channel. As the concept of multichannel shown in [6], we plan to design a multichannel system with one control channel and N data channels to maximize the utilization of the control channel. In our simulation model, the control channel is idle in most of the simulation time. The use of N data channel will increase the number of simultaneously transmission and as well the number of control frame, which is transmitted on control channel.

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