Efficient Access Control Techniques for Distributed Wireless Communication Networks

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Preface

Due to the emergence of various services and applications on the Internet and the proliferation of smart and high performance mobile devices, the amount of mobile data traffic is explosively increasing in these years. The wireless local area network (LAN) system has been an important communication method as well as the cellular system to support such demands for mobile data communications. In order to support various services and applications having various quality of service (QoS) requirements, and to satisfy demands of various users, interworking of the cellular and wireless LAN systems are expected to take advantage of each system.

By fully utilizing the centralized control mechanism in the licensed spectrum, the cellular system is suitable to support high quality services and applications ensuring some QoS parameters. However the wireless LAN system which was originally designed to make data transmission in a best effort manner using the distributed control mechanism in the unlicensed frequency band needs to be improved to have better spectrum efficiency and functionality.

In this thesis, techniques to improve the efficiency of the medium access control (MAC) layer and to support various services and applications for the wireless network using distributed access mechanism are presented. In chapter 2, a rate switching algorithm for the wireless systems supporting multiple PHY rates is presented. The impact of the rate switching algorithm on the system capacity is analyzed based on the IEEE 802.11a system assuming a simple network model. A simple extension to improve the efficiency of the history based rate switching algorithm is proposed and its performance is evaluated. In chapter 3, a communication quality control scheme for the IEEE 802.11 wireless LAN is discussed. A procedure to ensure the QoS parameters such as bandwidth, delay and jitter is presented based on the point coordination function which is an optional procedure defined for the IEEE 802.11

MAC layer. It is shown that parameterized QoS will be possible in a managed environment. Besides, a mechanism to make prioritized transmissions based on the distributed coordination function is proposed. Results of computer simulations show that the proposed method successfully makes priority control based on the distributed access mechanism. In chapter 4, a reliable multicast protocol is presented. Although the broadcast nature of wireless communication channel is suitable for the multicast transmission, the multicast mechanism defined in the original IEEE 802.11 MAC does not ensure reliability due to lack of acknowledgment. It is desired that the reliability of the multicast transmissions should be improved without losing the efficiency. For this purpose, a representative acknowledgment scheme is proposed which defines a STA group based acknowledgment procedure. The performance of the proposed multicast scheme is evaluated by computer simulations and numerical analysis. Finally, a protocol for the inter-vehicle communication network is presented in chapter 5 as an ultimate style of distributed network. In order to improve the safety of the vehicle, reservation-ALOHA based communication protocol is proposed assuming a direct sequence spread spectrum (DSSS) physical layer (PHY). The data transmission protocol and the slot reservation algorithms are discussed.

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Contents

List of Figures

List of Tables

Abbreviations

AAA Authentication, Authorization, Accounting

ACK Acknowledgment

AP Access Point

APL Application

BE Best Effort

BEB Binary Exponential Backoff

BS Base Station

BSS Basic Service Set

CCK Complementary Code Keying

CCMP Counter mode with Cipher-block chaining Message authentication code Protocol

CFP Contention Free Period

CL Controlled Load

CoS Class of Service

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

CW Contention Window

DCF Distributed Coordination Function

DIFS DCF Inter-Frame Space

DSSS Direct Sequence Spread Spectrum

EDCA Enhanced Distributed Channel Access

FCS Frame Check Sequence

FHSS Frequency Hopping Spread Spectrum

IP Internet Protocol
IrDA Infra-Red Data Association
LAN Local Area Network
LLC Logical Link Control
MAC Medium Access Control
MCS Modulation and Coding Scheme
MIMO Multiple Input and Multiple Output
NACK Negative Acknowledgment
OFDM Orthogonal Frequency Division Multiplexing
OFDMA Orthogonal Frequency Division Multiple Access
PCF Point Coordination Function
PIFS PCF Inter-Frame Space
PLCP Physical Layer Convergence Protocol
PMD Physical Medium Dependent
QoS Quality of Service
RSSI Received Signal Strength Indication
RTS Request to Send
SIFS Short Inter-frame Space
SNR Signal to Noise Ratio
SNS Social Networking Service
STA Station
TKIP Temporal Key Integrity Protocol
TCP Transmission Control Protocol
UDP User Datagram Protocol
UP User Priority
V2I Vehicle to Infrastructure
V2V Vehicle to Vehicle
VLAN Virtual Local Area Network
VV Voice and Video
WLAN Wireless Local Area Network

Chapter 1

Introduction

1.1 Background

By the emergence of various services and applications provided over the Internet such as cloud computing, audio and video streaming and social networking, and due to proliferation of high performance smart mobile devices such as smartphones and tablets, the amount of data traffic transmitted over the Internet continues to grow. Figure 1.1 shows the results of research on the number of the Internet users and penetration rate of Internet services in Japan conducted by the ministry of internal affairs and communications (MIC), Japan [1]. The results show that more than one hundred millions of people use the Internet and penetration rate of the Internet service exceeds 80% in Japan in 2013.

Figure 1.2 shows the devices that people use for the Internet access which is the result of research conducted by MIC (Ministry of Internal Affairs and Communications) Japan in 2013. The devices are roughly categorized into PC, mobile terminals such as smartphones and tablets, and other data terminals such as game terminals and Internet TVs. As shown in Fig. 1.2, most of the users use both PC and mobile device for the access to the Internet.

1.1.1 Explosion of Mobile Data Traffic

The amount of mobile data traffic is rapidly increasing due to the recent proliferation of smartphones and tablets. Most of the smart mobile devices available in the market

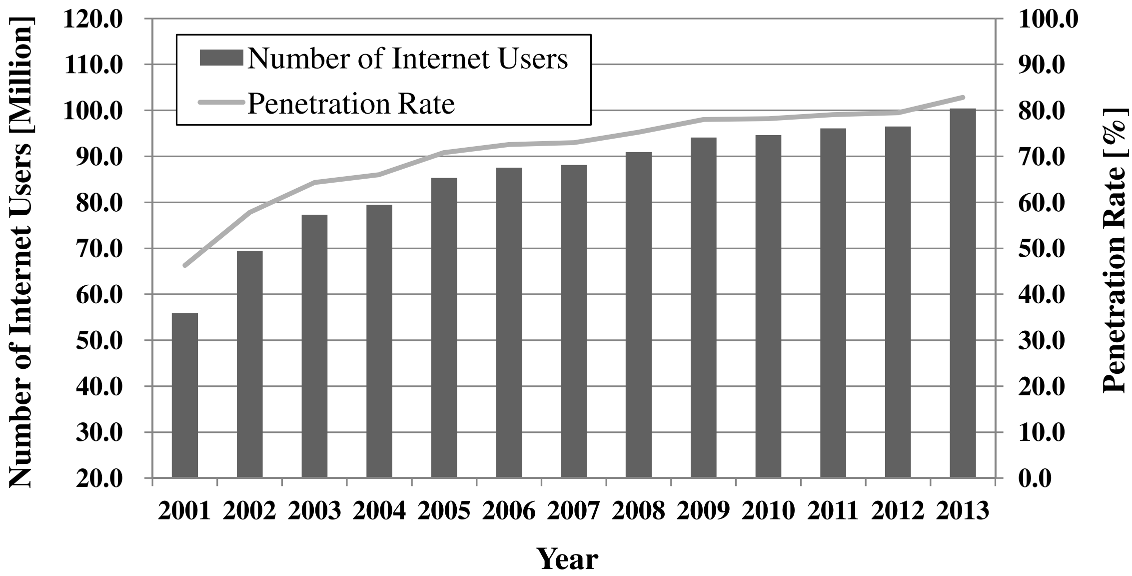

Fig. 1.1 Penetration of the Internet Service in Japan.

are equipped with wireless interfaces such as cellular, Bluetooth, IrDA, Wireless LAN and Global Positioning System (GPS). Among those interfaces, cellular and WLAN accesses are the primary interfaces for the Internet access. The cellular system has evolved from 3G (the 3rd generation cellular system) and HSPA (high speed packet access) to LTE (long term evolution) and LTE-Advanced systems. The WLAN system also has evolved from the IEEE 802.11a/b/g to the IEEE 802.11n, IEEE 802.11ac, and 802.11ad. As the speed of wireless interfaces and computing power of the processing unit become higher and higher, and more and more rich applications and services are actually used in the daily life of many people, the amount of the data from those smart mobile devices is significantly increasing.

According to the discussion in the radio policy vision council, it was reported that recent growth of the broadband data traffic is 1.6 times every year as shown in Fig. 1.3. The results of market research sponsored by Cisco Systems [2] show the similar results for the mobile data traffic. The Compound Average Growth Rate (CAGR) of the mobile data traffic from 2013 to 2018 will be 61% and there will be 15.9 Exabytes (1 Exabyte = 10^{18} Bytes)of such traffic transmitted from the mobile devices.

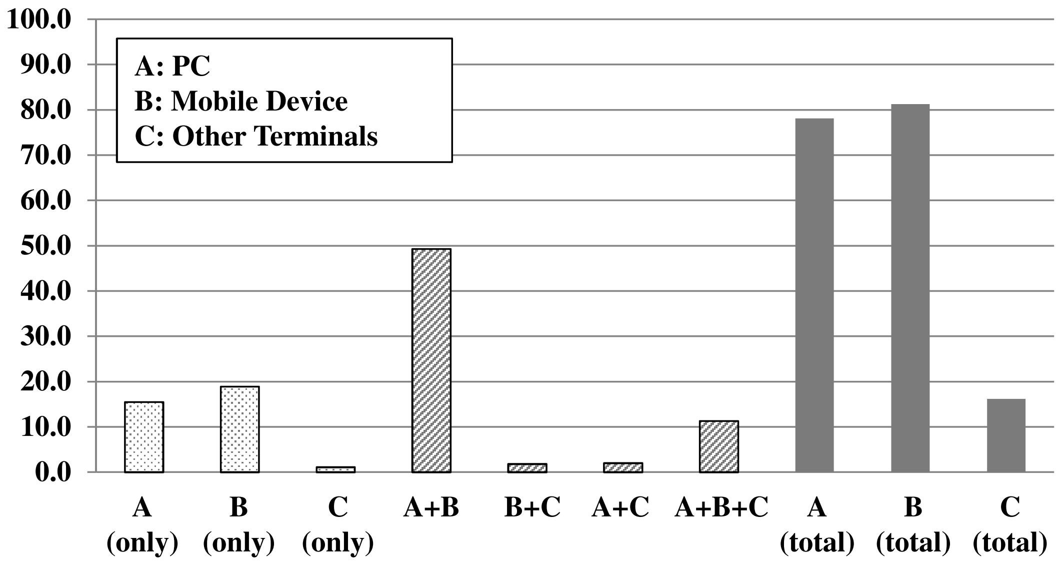

Fig. 1.2 Devices used for the Internet Access.

Mobile operators assume that the growth rate of mobile data traffic is twice every year which means that 1000 times mobile data traffic will be transferred over the network 10 years after. Therefore, how to accommodate this explosive amount of mobile data traffic is one of the biggest issues for the mobile communication industry. People are seriously talking about 5G, the fifth generation mobile communication systems and networks.

In the recent discussions of future mobile data communication services and systems, cellular data offloading is one of the biggest themes of the 5G system. According to the research conducted by Wireless LAN Business Promotion Council of MIC entitled "Future Trends of Mobile Traffic," 19.4% of total mobile data traffic is transmitted and received by the wireless LAN interface, i.e., offloaded to the wireless LAN, in 2012. In 2015, the amount of offloaded traffic is anticipated to be 64% of the entire mobile data traffic. This means that the wireless LAN is an important system which is expected to support the significantly increasing mobile data traffic by collaborating with the cellular system. However, the cellular system and the wireless LAN system have different characteristics and it is important to recognize what the differences are.

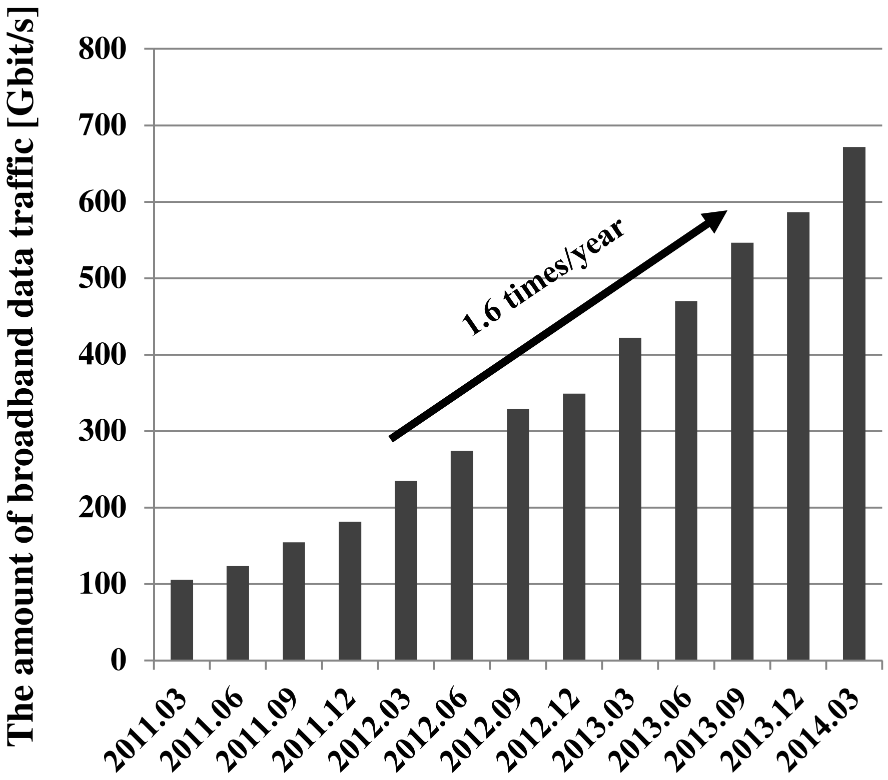

Fig. 1.3 Growth of Broadband Data Traffic.

1.1.2 Characteristics of Cellular and Wireless LAN Systems

Although the cellular and the WLAN systems use the radio frequency bands to communicate, those systems have different properties. Here, those characteristics are reviewed from the view point of the frequency bands and associated access control schemes.

Figure 1.4 shows the frequency assignment for the cellular and WLAN systems in Japan. The cellular systems use the licensed frequency bands from 700 MHz to 2100 MHz while the WLAN systems use the frequency of 2.4 GHz band and 5 GHz bands.

The cellular systems use the licensed frequency bands. A licensed mobile network operator can exclusively use the assigned frequency bands. In such frequency bands, reliable and efficient services can be offered by designing the service area appropri-

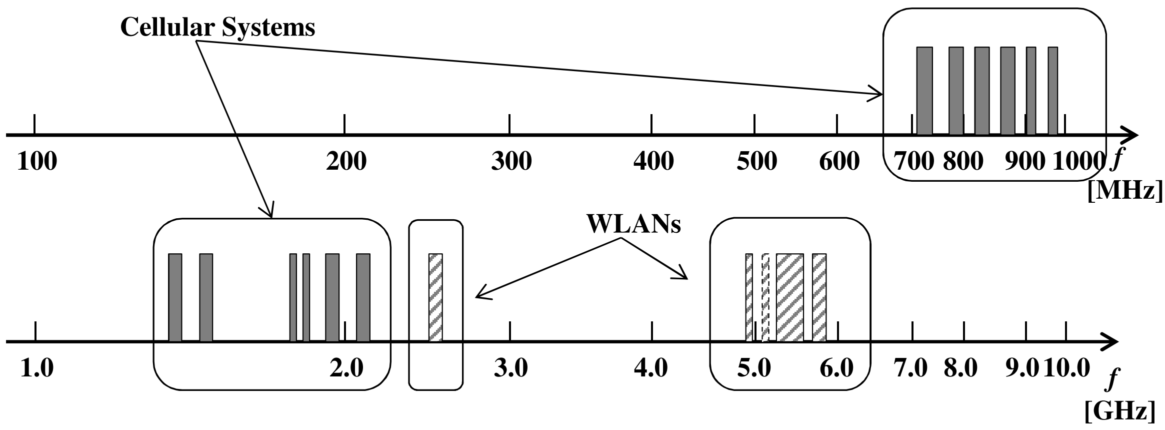

Fig. 1.4 Frequency Assignment to cellular and WLAN systems in Japan.

ately based on interference calculation, and by using centralized control methods for medium access, resource management, etc. Therefore, the licensed frequency bands are ideal medium for the mobile network operators to provide their services. From the user's point of view, however, it is necessary to have a contract with an operator for a subscription of the communication service and to pay for subscription fee.

The WLAN systems use the unlicensed frequency bands. One of the benefits of using the unlicensed frequency band is that anyone can install and operate a wireless network using certified devices. Therefore, WLAN is suitable for building a wireless network in a private space such as homes and offices where the user sometimes needs to change the configuration and/or topology of the network. The cost effectiveness is also an advantage of WLAN in building the wireless network. The price of the device is reasonable and most of the mobile handheld devices available in the market are equipped with the WLAN interface by default. For the wireless devices using the unlicensed spectrum, it is required to share the frequency resource with other systems and/or devices operating on the same channel when they are close to each other. For this purpose, the wireless devices that use the unlicensed spectrum employ the distributed access control mechanism. For the case of the WLAN, CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance) is adopted as the basic channel access procedure which is described in the next section. Use of distributed access control scheme makes frequency utilization rate lower than the centralized system and makes it difficult to ensure quality of service (QoS) parameters such as delay, jitter and bandwidth.

1.1.3 Importance of Unlicensed Frequency Band

The cellular and the WLAN systems have both advantages and disadvantages. As discussed previously, the cellular system can provide efficient and reliable services using the licensed spectrum while the WLAN system can provide a wireless access method in flexible and cost effective way by using the unlicensed spectrum. In the future wireless communication services, therefore, it is desired that the cellular and the WLAN systems collaborate each other to support various services and applications in a reliable and efficient manner at a low cost.

However, the WLAN system using distributed control mechanism in the unlicensed spectrum makes transmission in a best effort manner. It needs to be improved to support various applications and services collaborating with the centralized system such as cellular. Before looking into the problems of the distributed access control system, background of distributed wireless networks are reviewed next.

1.2 History of Distributed Wireless Networks

Studies on computer network in early '60s introduced a new communication style called packet switching. J. C. R. Licklider of MIT envisioned a globally interconnected set of computers and through which everyone could quickly access data and programs from any site for social interaction through the network. The concept of such a computer network can be seen in the Internet today.

During the research and development of computer networks, important concept of distributed network has been established. In this section, history of the distributed network is reviewed based on the typical implementations of ARPANET, ALOHA network and the IEEE 802.11 WLAN.

1.2.1 ARPANET

The ARPANET was the first packet based data communication network sponsored by the advanced research project agency of the United States of America and is regarded as the origin of the Internet.

Before the ARPANET, all kinds of communications including voice and data are conducted in the circuit switched manner, i.e., a dedicated communication path is established between the end points for each communication demand. In the packet networks, data is processed into packets which are data segments with source and destination addresses and other control information attached, and transmitted over a shared communication medium.

The initial ARPANET comprised of four host computers in University of California Los Angeles, Stanford Research Institute, University of California Santa Barbara and Utah University grew into the Internet which is interconnection of independent and autonomous networks. According to the "A Brief History of the Internet" on the web site of the Internet Society [3], it is noted that "robustness and survivability, including the capability to withstand losses of large portions of the underlying networks" was emphasized in the work of developing the Internet after the ARPANET to cope with the unreliable and poor quality of the communication links at that time.

In order to implement those features, the network should not assume centralized control entity. Therefore, the distributed control has become the basis for the Internet.

1.2.2 ALOHAnet and Random Access Protocols

The benefit of packet based communication is exploited by the systems using the shared communication medium. The first packet based wireless network was developed by the ALOHA project of Hawaii university under the leadership of Norman Abramson in 1970 to connect central computer and remote consoles located in other islands [4, 5].

The original version of ALOHAnet used two distinct frequency channels in the UHF band and employed a hub/star configuration. The hub machine broadcasts data to all client terminals on the network using the “outbound” channel while various kinds of client machines send data packets to the hub using the “inbound” channel. The inbound channel, therefore, can be regarded as a random access channel. Use of a short acknowledgment packet upon correct reception of a data frame on the random access channel made it possible for the client machines to know whether the transmission was successful. If no acknowledgment was received after transmission of a data packet, the client machine recognized that the transmission was unsuccessful. The protocol used in the ALOHAnet, called pure-ALOHA or simply ”ALOHA”, is summarized as follows.

- A client is allowed to start transmission whenever it is ready to do so.
- Upon detection of an unsuccessful transmission, the client retransmits the data packet later after a random deferral time.

The primary importance of the ALOHAnet was that the use of shared communication medium in which data transmissions including retransmissions are carried out in a distributed manner. The performance analysis of the ALOHA protocol made it recognized that the collisions of packets always cost high. And then many researches have been conducted to improve the performance of the random access protocol.

The slotted ALOHA is a revised version of the pure ALOHA protocol in which transmission of a packet is synchronized to the slot which reduced collision probability of the transmitted packets and improved throughput by twice. Reservation ALOHA is extension to the slotted ALOHA protocol that allows cyclic use of the slot in a time frame. In order to reduce the collision probability, so called ”listen before talk” rule was adopted in the distributed access control mechanism and the Carrier Sense Multiple Access (CSMA) protocol was developed. The CSMA protocol has become a basis for the MAC protocol of the Local Area Network (LAN) in which data is exchanges over a shared medium. The Ethernet protocol adopted Carrier Sense Multiple Access with Collision Detection (CSMA/CD) while the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) protocol was employed in the wireless LAN which is described next.

1.2.3 IEEE 802.11 Wireless LAN

The IEEE 802.11 is the wireless LAN standard developed by the working group (WG) 11 of the IEEE 802 Local and Metropolitan area network Standards Committee (LMSC) [8] which specifies physical (PHY) and MAC layers, and related management entities as shown in Fig. 1.5.

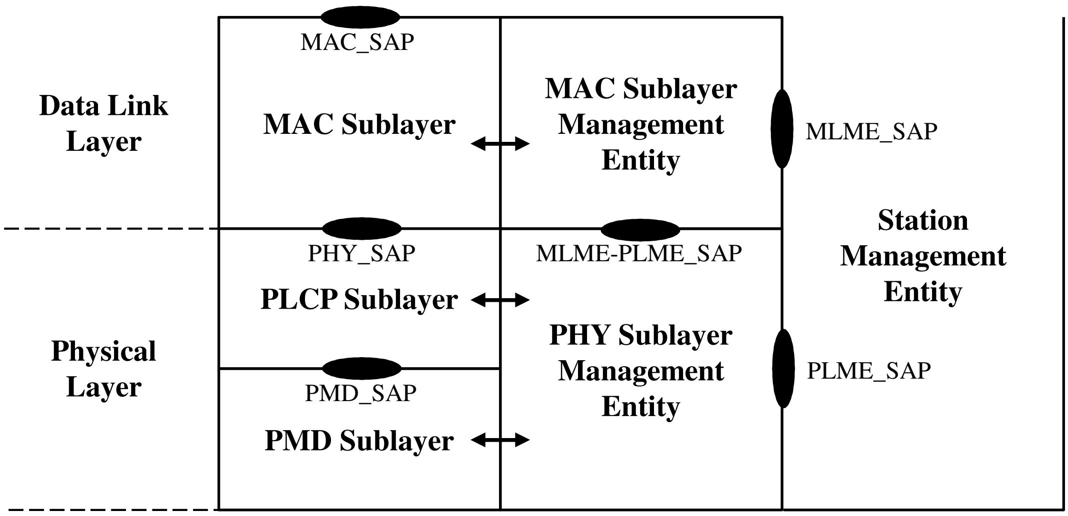

Fig. 1.5 Reference Model of the IEEE 802.11 WLAN.
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1.2.3.1 The IEEE 802.11 PHY

The IEEE 802.11 PHY layer consists of two sublayers, i.e. the Physical Layer Convergence Protocol (PLCP) and the Physical Medium Dependent (PMD) sublayers. The PLCP sublayer specifies the PHY convergence function which adapts the capability of the PMD system to the PHY service. The PMD sublayer defines the characteristics and the method of transmitting and receiving the data between two or more STAs through a wireless medium.

The initial version of the IEEE 802.11 standard specified three kinds of physical layers, i.e. DSSS PHY, Frequency Hopping Spread Spectrum (FHSS) PHY, and Infrared (IrDA) PHY. The DSSS and FHSS PHYs were designed to use the unlicensed spectrum of the 2.4 GHz Industrial, Scientific, and Medical (ISM) band. All of these PHY layers offered data rates of 1 and 2 Mbit/s.

The IEEE 802.11a standard introduced Orthogonal Frequency Division Multiplexing (OFDM) PHY in the 5 GHz band. Using the 20 MHz of the channel bandwidth, the IEEE 802.11a devices offer the data rates of up to 54 Mbit/s. The IEEE 802.11b extended the DSSS PHY in 2.4 GHz band to have data rate close to the 10 Base-T interface of the Ethernet. By adding complementary code keying (CCK) modulation scheme, the IEEE 802.11b offered the data rates of 5.5 and 11 Mbit/s. The 802.11g further extended the 802.11b standard by applying the OFDM PHY defined in the 802.11a standard to the 2.4 GHz band.

The IEEE 802.11n standard ratified in 2009 extended the maximum data rate to be 600 Mbit/s by defining the Multiple-Input and Multiple-Output (MIMO) technique and by optionally adding 40 MHz channel bandwidth for the OFDM PHY in both 2.4 GHz and 5 GHz bands.

The latest standard of IEEE 802.11ac further extended the channel bandwidth to be 80 MHz, non-contiguous 80+80 MHz, and 160 MHz. The downlink multi-user MIMO was also introduced to enhance the network throughput to aggregate the STAs having single antenna.

1.2.3.2 The IEEE 802.11 MAC

Figure 1.6 shows the original architecture of the IEEE 802.11 MAC sublayer which consists of two coordination functions. The Distributed Coordination Function (DCF) provides contention services while Point Coordination Function (PCF) is optional and is required for the contention-free services.

The use of unlicensed spectrum required the wireless LAN system to have the capability of coexistence with other systems operating on the same channel in proximity. Therefore the DCF which is based on the CSMA/CA protocol was developed for the basic channel access protocol which is a realization of “Listen-Before-Talk” behavior.

In the DCF procedure, all stations (STAs) including access points (APs) are always sensing the channel and recognize that the channel is idle if no transmission is detected for a period specified by DCF Inter-Frame Space (DIFS) as shown in Fig. 1.7. When

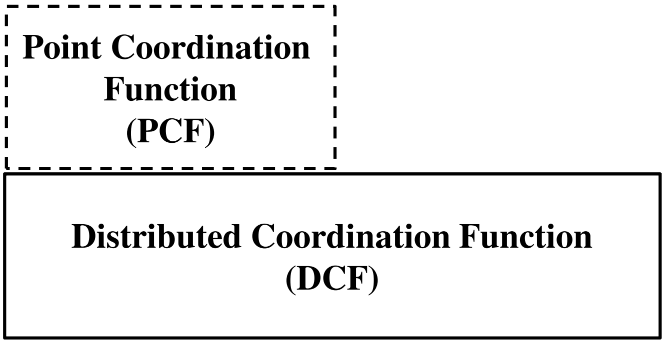

Fig. 1.6 The original IEEE 802.11 MAC architecture.
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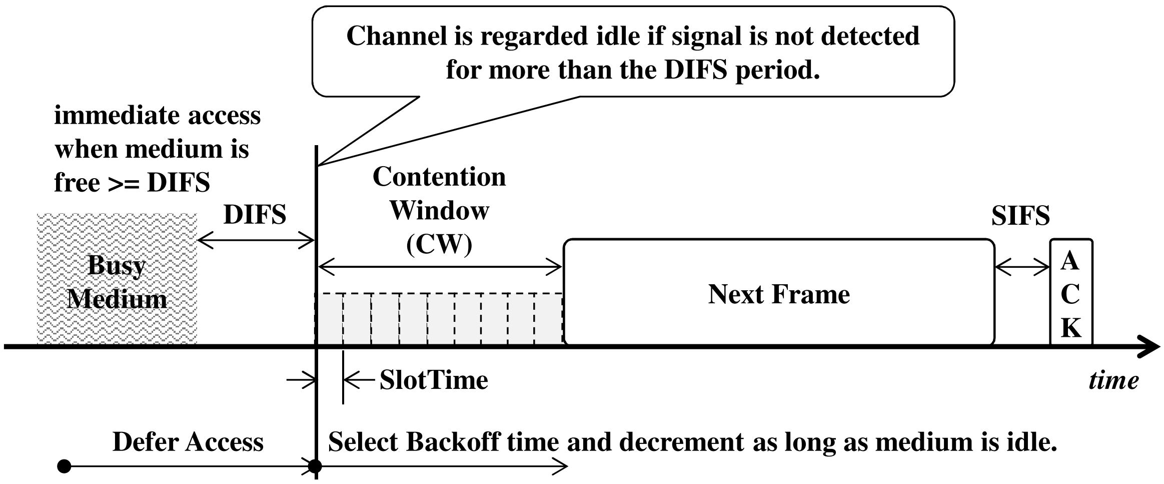

Fig. 1.7 Basic Access Procedure of the IEEE 802.11 original MAC.

a STA has data to transmit the STA immediately starts transmission of the data if the channel is idle. If the channel is busy, however, the STA waits until the channel becomes idle and generates a random number which is uniformly distributed between 0 to CW_{min} to set the backoff counter. And then the STA decrements the backoff counter value every after the SlotTime by 1 as long as the channel is idle and starts transmission when the backoff counter reaches 0. If the channel becomes busy before the backoff counter reaches 0, the STA suspends count down and waits until the channel becomes idle again and repeats the above procedure.

The recipient of a unicast frame needs to send an acknowledgment frame to the transmitter if the frame is received correctly. If no response was returned for a unicast transmission, the transmitter recognizes that the transmission was unsuccessful and tries to retransmit it. However, in the case of retransmission, the range of generating a backoff counter value was doubled to reduce the collision probability of the retransmitted frame. This is the basic channel access procedure of DCF which is suitable for making transmissions in a best effort manner. In [10], Bianchi proposed a simple and accurate analytical model of DCF and provided results of performance analysis.

After the original IEEE 802.11 standard was ratified in 1997, the 802.11 WG continues standardization activity to achieve higher data rates and support various functionalities. The IEEE 802.11e defined QoS support mechanism by defining Hybrid Coordination Function (HCF) which is extended and integrated coordination function of the conventional DCF and PCF providing prioritized and parameterized QoS, respectively.

The IEEE 802.11i standard enhanced security features and defined two security protocol called Temporal Key Integrity Protocol (TKIP) which is still based on RC4 and Counter mode with Cipher-block chaining Message authentication code Protocol (CCMP) based on Advanced Encryption Standard (AES) developed by the National Institute of Standards and Technology (NIST) to replace the conventional Wired Equivalent Privacy (WEP) scheme.

There are many other important amendments for the IEEE 802.11 standard such as the Inter-basic service set (BSS) transition protocol defined in the IEEE 802.11r, the mesh networking defined in the IEEE 802.11s, the interworking mechanism with the external network defined in the IEEE 802.11u. Table 1.1 summarizes the published standard and amendment of the IEEE 802.11 WLAN.

There are still on going activities in the IEEE 802.11 WG as listed in Table 1.2. Details can be found on the web site of the IEEE 802.11 WG [9].

Table. 1.1 The IEEE 802.11 Published Standards.

Standard	Description
IEEE 802.11-1997	The original wireless LAN standard. PHY: DSSS, FHSS and IrDA MAC: DCF (mandatory) and PCF (Optional)
IEEE 802.11a	OFDM PHY in the 5 GHz
IEEE 802.11b	High Speed PHY in the 2.4 GHz band.
IEEE 802.11d	Multi-Regulatory Operation.
IEEE 802.11e	MAC enhancement for QoS support.
IEEE 802.11g	Extended Rate PHY in 2.4 GHz band.
IEEE 802.11h	Spectrum Managed 802.11a.
IEEE 802.11i	Security enhancement.
IEEE 802.11j	Japanese 5 GHz Wireless Access System.
IEEE 802.11k	Radio Resource Measurement.
IEEE 802.11n	High Throughput PHY and MAC for 2.4 and 5 GHz bands.
IEEE 802.11p	Wireless Access in the Vehicular Environment.
IEEE 802.11r	Fast Inter-BSS Transition Protocol.
IEEE 802.11s	Mesh Networking.
IEEE 802.11u	Interworking with External Networks.
IEEE 802.11v	Wireless Network Management.
IEEE 802.11w	Protected Management Frames.
IEEE 802.11y	Public Safety in the United States.
IEEE 802.11z	Direct Link Setup.
IEEE 802.11aa	Video Transder Streams.
IEEE 802.11ac	Very High Throughput PHY and MAC below 6 GHz.
IEEE 802.11ad	Very High Throughput PHY and MAC in 60 GHz.
IEEE 802.11ae	Prioritized Management Frames.
IEEE 802.11af	Operation in the TV White Space.

Table. 1.2 The IEEE 802.11 Standardization Activities.

Activity	Description
IEEE 802.11 TGah	Sub 1 GHz operation.
IEEE 802.11 TGai	Fast Initial Link Setup.
IEEE 802.11 TGaj	Chinese Millimeter Wave
IEEE 802.11 TGak	Transit Links within Bridged Networks
IEEE 802.11 TGaq	Pre-Association Service Discovery.
IEEE 802.11 TGax	High Efficiency WLAN.

1.3 Challenges of Distributed Wireless Networks

As shown in Fig. 1.8, there are many services and applications available on the Internet each of which has unique characteristics. Many people enjoy the streaming audio and video, contents sharing, social networking services (SNS), and many others in their daily lives using smart mobile devices. Nowadays, most of the mobile terminals are equipped with both cellular and WLAN interfaces.

The cellular and the WLAN systems are expected to collaborate together taking advantage of each systems to satisfy users' demands. Ideally, the mobile data communication services should be provided without making the user to be aware of the communication means or medium.

When thinking collaboration of cellular and WLAN systems, the cellular system can provide high quality services because of the nature of the centralized control mechanism. On the other hand, the distributed access control mechanism employed in the WLAN system is originally designed to make data transmissions in a best effort manner and it needs to be improved to support various kinds of services and applications. Therefore, the improvement of the distributed access control mechanism is the key enabler for cellular and WLAN collaboration.

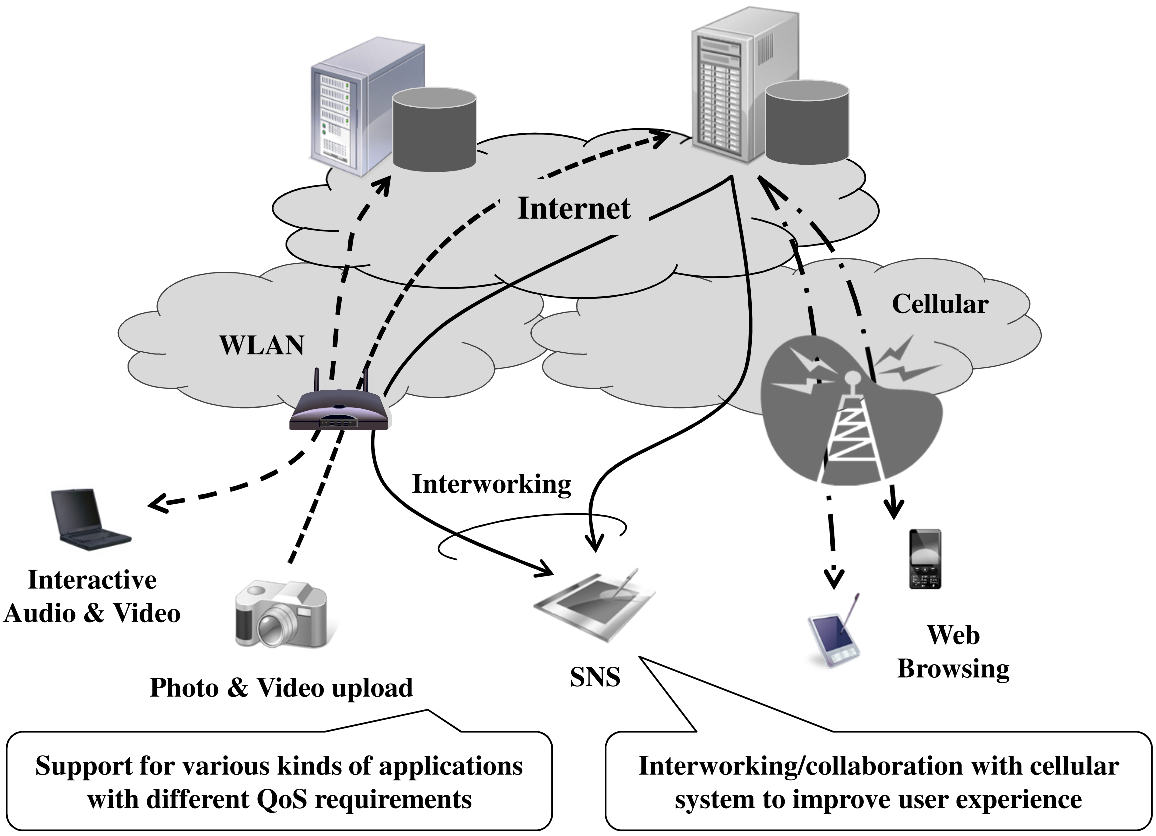

Fig. 1.8 Requirements for the distributed wireless network

1.4 Contribution of This Thesis

In order to enhance the performance and functionalities of the distributed access mechanism, studies on following items have been conducted and described in the subsequent chapters which are summarized in Fig. 1.9.

Improved Management and Cross-Layer Optimization

In order to optimize the performance of the system, there are many control and management mechanisms. In chapter 2, a rate switching algorithm and its impact on the system capacity is studied based on the IEEE 802.11a WLAN system. This is one of the control and management schemes for cross-layer optimization.

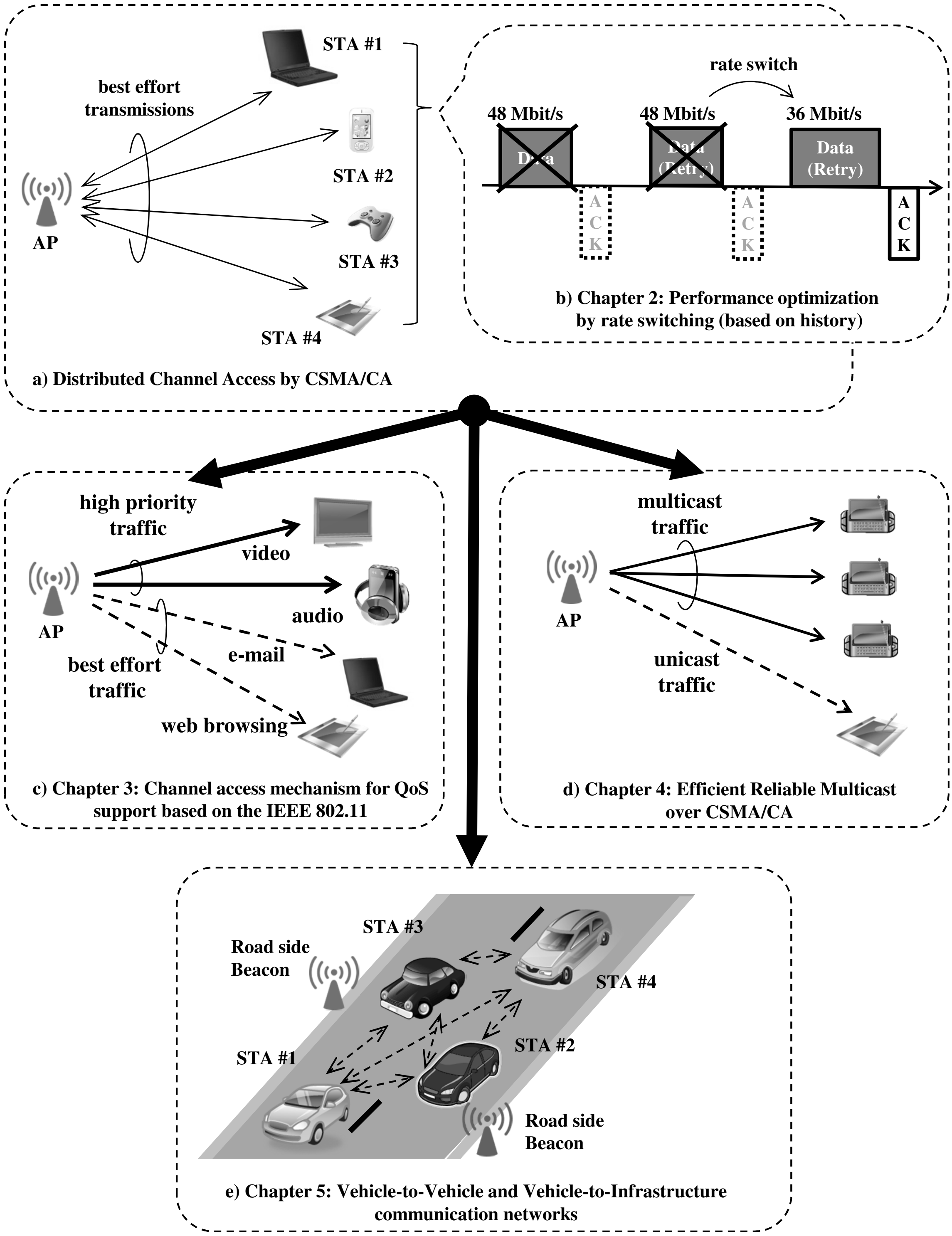

Fig. 1.9 MAC Functionalities over the Distributed Access Control.

Recent wireless systems basically support multiple PHY rates, and inappropriate choice of the transmission rate significantly affects the throughput performance and the system capacity. Although the rate switching algorithm itself is out of the scope of the IEEE 802.11 WLAN standard, it is important to choose an appropriate transmission rate in a given environment for the high efficiency.

The typical rate switching algorithm used in the WLAN system is a history based rate switching which is simple and effective method that does not require any explicit feedback for the rate selection. A simple modification to the typical rate switching algorithm is proposed to improve the throughput and system capacity.

Enhancement of Functionalities

Communication Quality Control

Chapter 3 discusses the communication quality control schemes for the IEEE 802.11 WLANs. In order to improve user experience, data for some kinds of applications need QoS support to ensure communication parameters such as bandwidth, delay and jitter. The cellular system is suitable for this purpose because of the centralized control mechanism. The WLAN system, however, needs to be enhanced to support this feature based on the distributed access control mechanism.

The IEEE 802.11 WLAN standard specifies a centralized access mechanism based on the polling scheme called PCF on the top of the distributed access control. Although the effectiveness of PCF is limited within a managed or interference-free environment, use of PCF is one way of supporting QoS for the WLAN system. In this case, resource management scheme in the AP becomes important.

Due to lack of inter-BSS coordination mechanism, it will be difficult to support QoS in the overlapped BSS (OBSS) environment using the PCF. In this case, a simple priority control scheme based DCF is more appropriate. Therefore, a mechanism to enhance the DCF procedure to make priority control is also proposed.

The effect of PCF based QoS support scheme and DCF based priority control scheme is evaluated by the computer simulations and the applicability of those techniques are discussed.

Reliable Multicasting

Multicast is an efficient way to deliver the same set of data to a group of users which goes together with the broadcast nature of the wireless medium. One of the challenges in a multicast communication is to ensure the reliability because multicast communication, in general, does not have the acknowledgment procedure.

In chapter 4, we will discuss the reliable multicast protocol for the WLAN system. To add the reliability to the multicast transmissions, an acknowledgment procedure is added to the multicast sequence. The point here is not all of the STAs return the acknowledgment for the received multicast data, but an selected STA from a group of STAs sends an acknowledgment. The performance of the proposed scheme is compared with the other reliable multicast protocol.

Support for Emerging Applications

As an example of the ultimate style of the distributed wireless networks, inter-vehicle communication network and the MAC protocol is discussed in chapter 5. An inter-vehicle communication network has some unique characteristics.

Assuming that the primary objective of inter-vehicle communications is to improve the safety by providing information of other vehicles in proximity, we have derived requirements to design the MAC protocol and proposed a Reservation-ALOHA based MAC protocol for the inter-vehicle communication network. The performance of the proposed protocol is evaluated by the computer simulations.

Chapter 2

Rate Switching Algorithm for IEEE 802.11 Wireless LANs

2.1 Overview

The IEEE 802.11 wireless LAN systems such as the IEEE 802.11a, 802.11b, 802.11g and 802.11n support multiple transmission rates in the PHY layer. In such cases, a station chooses the data rate to be used for each frame transmission by rate switching algorithm. Despite the rate the switching algorithm has a great impact on the throughput performance, the algorithm itself is out of the scope of the standard and developers implements their own algorithms. In this chapter, general rate switching algorithm of the IEEE 802.11 wireless LAN that is actually used in many products is introduced and its effect on the performance is evaluated. Moreover, new rate switching algorithm is proposed to improve the throughput performance. Although the IEEE 802.11a PHY is assumed in this chapter, the proposed algorithm can be applied to any wireless systems supporting multiple transmission rates at the PHY layer. Throughput performance and system capacity of the IEEE 802.11a WLAN is analyzed based on the basic frame exchange sequence of the IEEE 802.11 MAC protocol. It is shown that the IEEE 802.11 MAC will have much better throughput and system capacity with the proposed algorithm.

2.2 Introduction

The IEEE 802.11 working group has developed wireless LAN standards. The original IEEE 802.11 standard [11] published in 1997 defined three kinds of PHY layers, namely FHSS and DSSS which operate in 2.4 GHz ISM band and IrDA, and all of those three PHYs offered transmission rates of 1 Mbit/s and 2 Mbit/s. The IEEE 802.11b standard extended the original DSSS PHY and specified additional data rates of 5.5 and 11 Mbit/s using CCK modulation. The IEEE 802.11a standard introduced OFDM PHY for the 5 GHz band and specified eight data rates of 6, 9, 12, 18 24, 36, 48 and 54 Mbit/s. The IEEE 802.11g standard made OFDM PHY available in 2.4 GHz band maintaining backward compatibility with the IEEE 802.11b standard. The IEEE 802.11n specified the MIMO OFDM PHY and extended the data rates up to 600 Mbit/s. All these amendment is contained in the latest version of the IEEE 802.11 wireless LAN standard.

Recently, the IEEE 802.11 working group has developed the IEEE 802.11ac and IEEE 802.11ad standards [12, 13] which offer data rates of more than 1 Gbit/s in 5 GHz and 60 GHz bands, respectively. All those PHY layers specify multiple data rates.

Compared to the maximum transmission rate, however, the actual throughput is not so high. For example, the maximum throughput of a user using 802.11a device is about 30 Mbit/s even though the highest transmission rate of 54 Mbit/s is available. One of the reasons for this is an overhead of the MAC protocol. The CSMA/CA protocol employed as the basic medium access control method of the WLAN MAC layer requires frame transmissions in the manner of "Listen-Before-Talk." Therefore, in addition to the ACK procedure and optional Request-to-Send/Clear-to-Send (RTS/CTS) handshake, the time duration for a STA to determine that the channel is in idle and the contention window to avoid collisions are the overhead in a frame exchange sequence. The data frame format also introduces overhead by adding control and address fields, security encapsulation and frame check sequence (FCS).

Besides, there are management procedures which incur additional overhead. One of the reasons to decrease the throughput is rate switching algorithm. The transmission rate for a frame is dynamically determined by the transmitting STA considering the channel quality between the transmitter and receiver pair. A higher transmission rate will be selected when the channel quality is good enough. On the other hand, a lower transmission rate will be chosen to maintain connectivity and/or reliability when the channel condition is not so good. Although the transmission rate for a specific frame is supposed to be determined by the rate switching algorithm of the MAC layer, the algorithm itself is out of the scope of the standard, and a manufacturer can implement its own algorithm considering the requirements from the market segment of interest. Since the rate switching algorithm can be used as one of the differentiating technology of the product, not so many actual rate switching algorithms have been disclosed.

There are some studies on the rate switching algorithm for the IEEE 802.11 wireless LANs that discuss QoS related issues [14], [15]. However, there are not many works that discussed the impact of the rate switching algorithm on the system capacity [16], [17]. In these papers, methods of maintaining a certain level of communication quality have been discussed when switching to a lower modulation and coding scheme (MCS). There are not so many works that evaluate the impact of the rate switching algorithm on the throughput performance. In this chapter, the effect of the rate switching algorithm on the throughput is discussed, and then, a simple modification to improve throughput performance is presented. Throughput performances and the system capacities using the conventional and proposed rate switching algorithms are analyzed. It is shown that higher system capacity can be expected by using the proposed rate switching algorithm.

Rest of this chapter is organized as follows. An example of rate switching algorithm actually used in the IEEE 802.11 wireless LAN is presented in section 2.3. The proposed rate switching algorithm is explained in section 2.4 and analysis of throughput performance is presented in section 2.5. Section 2.6 concludes this work.

2.3 An Example of Rate Switching Algorithm for the IEEE 802.11 WLANs

In this section, a basic idea and typical rate switching algorithm for the IEEE 802.11 WLAN is introduced.

2.3.1 Basic ideas for rate switching

If the PHY layer supports multi-rate capability, the transmission rate for frame transmissions should be selected by considering the communication quality between the transmitter and receiver. The transmitter needs to change the selected transmission rate in accordance with the change of radio environment. The transmitter can use a higher transmission rate when the channel condition is good. On the other hand, the transmitter should use a lower transmission rate when it detects degradation of channel condition. Therefore, estimation of the channel condition is important.

There are some ways to estimate the channel condition. The simplest way is to estimate the channel condition using the recent transmission results, The basic channel access procedure of the IEEE 802.11 wireless LAN called DCF requires a STA to return an ACK frame after successful reception of a unicast frame. If no ACK frame is received within a specific time period after transmission of a unicast frame, the transmitter understands that the transmission was not successful. Therefore, the transmitter can estimate the channel condition whether it is good enough to send a frame of a specific length using the selected MCS from the results of recent transmissions. This simple method does not require any additional procedures to send a frame and is widely used in the actual system. There are some ideas to consider the Received Signal Strength Indication (RSSI) and/or Signal to Noise Ratio (SNR) of the received frame from the receiver of the transmitting frame. However, it might be sometimes difficult to estimate the channel condition from the received signals because of the asymmetric characteristics of the communication channel.

Another example of estimating channel condition is to collect channel state information. The IEEE 802.11n specified a way to obtain channel information by the channel state information (CSI) feedback mechanism. Although they are originally specified for the purpose of transmit beamforming, it can also be used for the rate switching. The CSI feedback mechanism, however, needs additional frame exchange sequence to obtain latest channel information which introduces additional overhead. The use of CSI feedback is optional in the standard and it is not widely used in the actual products. Therefore, we focus on the conventional history based rate switching algorithm.

2.3.2 Typical rate switching algorithm

An example of the typical rate switching algorithm used in the IEEE 802.11 wireless LAN is to change the transmission rate to a higher rate after continuous success of transmission attempts for specified times and to a lower rate after successive transmission failure.

Every STA keeps counts of continuous success or failure of frame transmissions for each of the receivers. When frame transmissions succeed for U times successively, the transmitting STA chooses higher transmission rate. On the other hand, the transmitting STA chooses lower transmission rate when transmission attempts failed for D times successively. This behavior of transmitter can be observed by analyzing the actual frame sequence between the STAs.

In this study, we call U the Rate-Up threshold which is the threshold to change the transmission rate to the higher rate. Also we call D the Rate-Down threshold. When the Rate-Up threshold is small, the STA chooses a higher data rate more aggressively which could eventually result in higher frame error rate. When the Rate-Up threshold is large, the STA becomes careful to use a higher transmission rate. When the Rate-Down threshold is small, the STA chooses lower transmission rate more easily.

One of the biggest issues of the rate switching algorithm is its impact on the throughput performance. Throughput degradation caused by the rate switching algorithm is observed at every STA in the service area, and the throughput degradation is a serious issue which results in reduced system capacity. For example, let us look at a STA

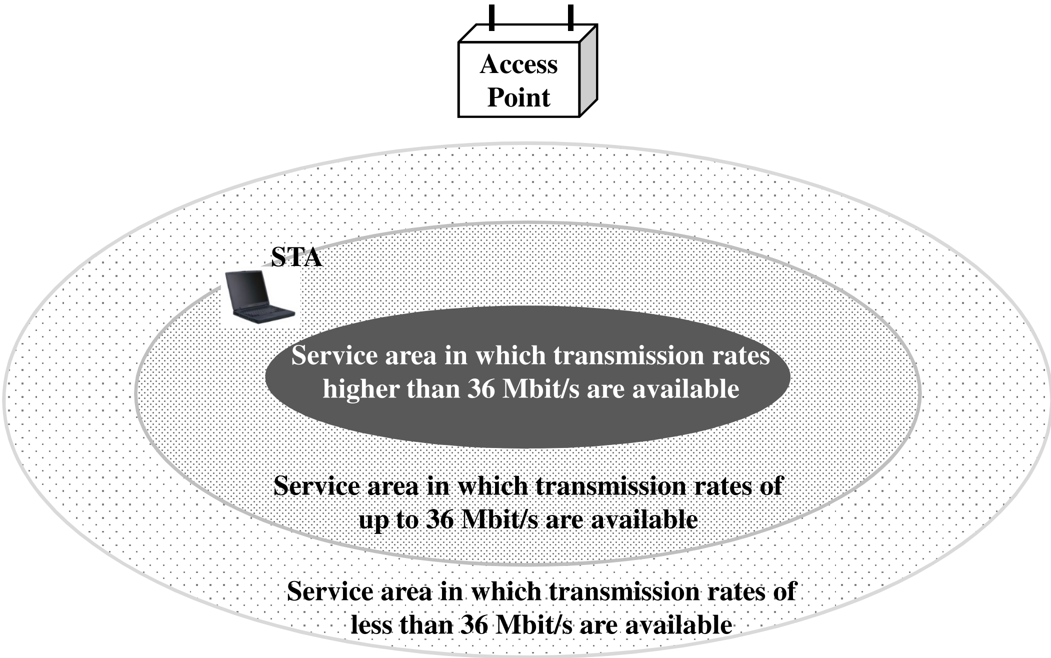

Fig. 2.1 Available transmission rates within a service area.

communicating with the AP of the service area. For simplicity, the communication quality of the channel between the AP and STA only depends on the distance between the AP and STA, i.e., the effect of fading nor shadowing is not considered. Then, the area in which a specific transmission rate is available becomes a circle as shown in Fig. 2.1 where the AP is located in the center of it. Suppose that the maximum transmission rate for the STA is 36 Mbit/s.

Figure 2.2 shows an example of the frame sequence using conventional rate switching algorithm. This figure explains the issues in the conventional rate switching algorithm. When transmissions of data frames succeed for U consecutive times, the STA assumes that the channel condition is good and changes the transmission rate to a higher rate. In the case of Fig. 2.2, the AP changes the transmission rate from 36 Mbit/s to 48 Mbit/s after U successful transmissions. However, required signal quality for the transmission rate of 48 Mbit/s is different from that for the transmission rate of 36 Mbit/s, and the transmissions using 48 Mbit/s rate likely to be unsuccessful unless

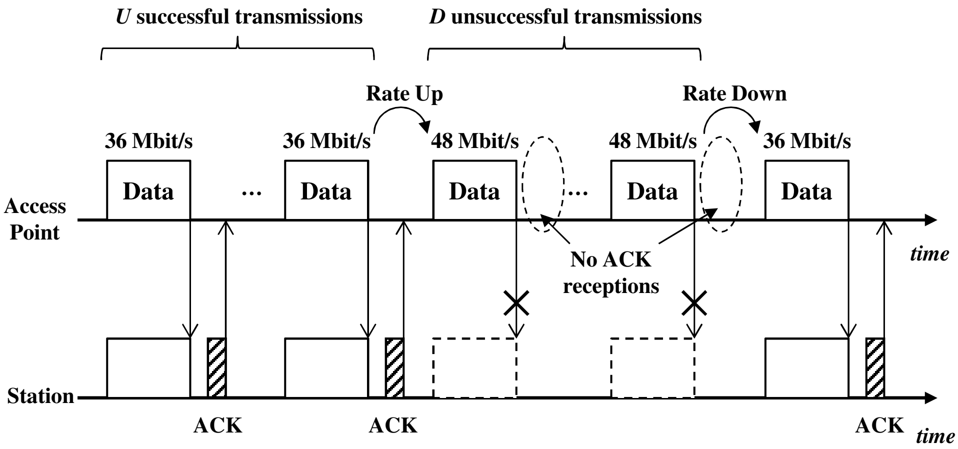

Fig. 2.2 Data transmissions using conventional rate switching algorithm. ©2004 IEEJ

communication quality of the channel between AP and STA is improved. The sender changes the transmission rate to a lower one after D consecutive failure of transmission attempts. In the meantime, the frequency resource is wasted. The ratio of successful transmission becomes $U/(U + D)$ and, in such cases, it is difficult to achieve high throughput. This throughput degradation will be observed in the places where the highest transmission rate is not available. Therefore degradation of throughput, and eventually the degradation of system capacity caused by the rate switching algorithm is a serious issue.

2.4 Proposed Rate Switching Algorithm

The proposed rate switching algorithm is a simple extension of the conventional rate switching algorithm to reduce inappropriate choice of a transmission rate. The point is to switch back to the previous transmission rate when the newly selected rate is too high to cause a higher frame error rate.

As described in the conventional rate switching algorithm, each STA has two counters called Rate Up Counter (RUC) and Rate Down Counter (RDC) to keep the results of frame transmissions. The RUC is incremented by one after every successful transmission of a data frame and is reset to zero after failure of a data frame transmission or change of the transmission rate. On the other hand, the RDU is incremented by one after every transmission failure of a data frame and is reset to zero after a successful data transmission. A STA changes the transmission rate to a higher one when the value of RUC reaches to U. On the other hand, a STA changes the transmission rate to a lower one when the value of RDU reaches to D. Those controls are the same as the conventional rate switching algorithm.

The difference between the proposed scheme and the conventional rate switching algorithm is described below. In the proposed scheme, a STA is assumed to have an additional variable called Rate Fix Counter (RFC) and a threshold to decide whether to continue to use the newly selected transmission rate, which is denoted by F.

The RFC is started when a STA changes the transmission rate to a higher one and is incremented by one after every successful transmission. It is stopped and reset to zero when the value of RFC reaches to F or a transmission failure occurs. In other words, the RFC is stopped and reset to zero when the STA judges that the channel condition between the STA and AP is good enough to continue to use the newly selected transmission rate, or a transmission failure occurs before the STA judges that the channel condition is good enough.

Figure 2.3 shows an example of how the proposed rate switching algorithm works. In this figure, the AP is sending data frames to a STA with a transmission rate of 36 Mbit/s. After U successful transmissions of data frames, the AP changes the transmission rate to 48 Mbit/s. When the AP changed the transmission rate to 48 Mbit/s, the AP starts RFC and observes the results of following data transmissions. In this figure, the data transmission using 48 Mbit/s fails because the channel quality is not good enough to use the transmission rate of 48 Mbit/s. In this case, the AP immediately switches back the transmission rate to 36 Mbit/s and stops RFC. By using RFC and the threshold F, throughput degradation caused by choosing excessively high transmission rate is mitigated.

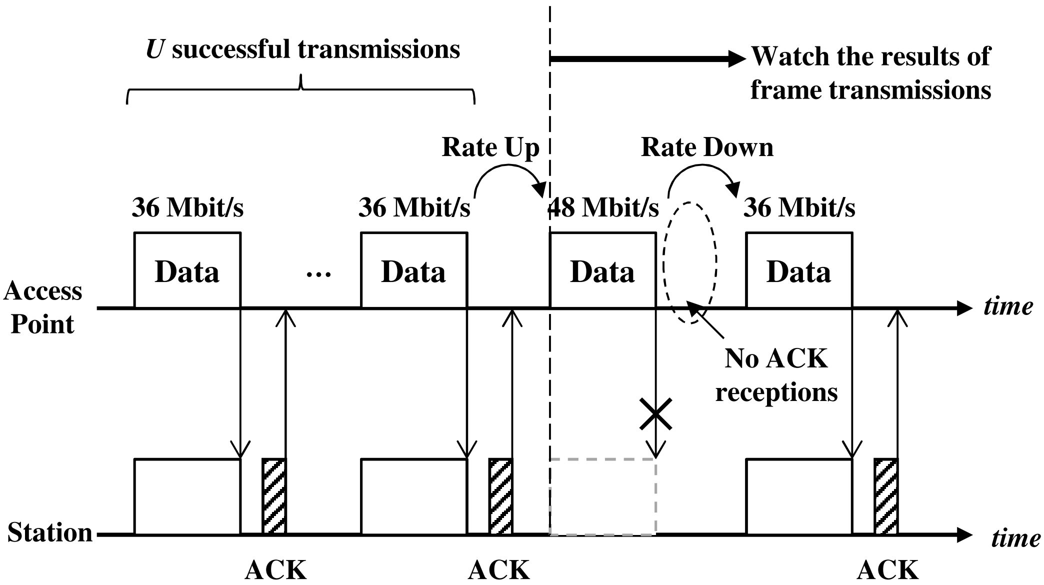

Fig. 2.3 Proposed rate switching algorithm.

2.5 Performance Evaluation

In this section, throughput performances of the IEEE 802.11 wireless LAN using conventional and proposed rate switching algorithms are evaluated. Since the intention here is to investigate the impact of rate switching algorithms, the effect of collisions by the CSMA/CA protocol is not considered. And, for simplicity, the effects of fading and shadowing are not considered. After evaluating the throughput performance, the system capacity is evaluated using the similar technique.

2.5.1 Throughput Analysis

Throughput is calculated from the frame sequence for the basic access procedure of the IEEE 802.11 called DCF and the frame formats for data and ACK frames. For simplicity, the protection mechanism using RTS/CTS exchange is not considered in this study.

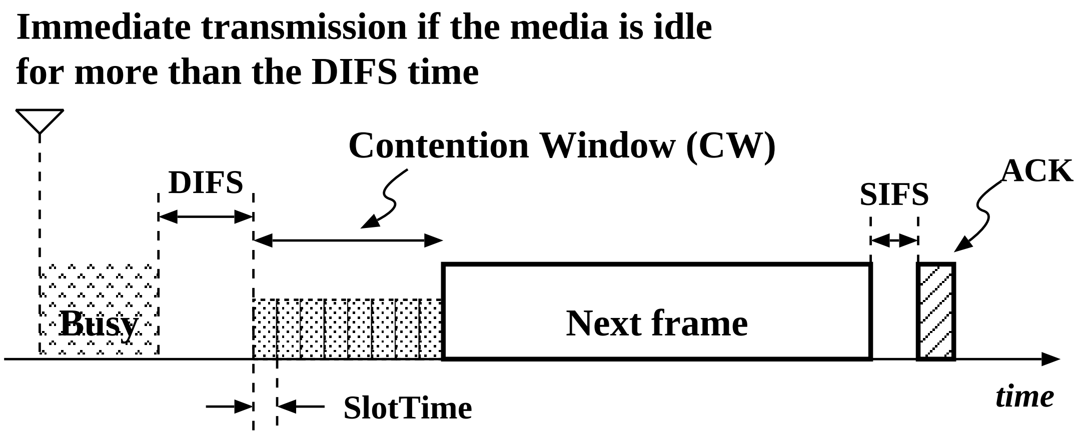

Fig. 2.4 Frame sequence of the IEEE 802.11 DCF procedure.
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The basic frame sequence of the IEEE 802.11 DCF is shown in Fig. 2.4. All STAs including AP continuously sense the channel and regard the channel in the idle state if no signal is detected for more than a time period called DIFS. When a data frame is generated or arrives at the MAC layer of a STA while the channel is idle, the STA immediately starts transmission of the data frame. If the data frame is generated or arrived at the MAC layer while the channel is busy, the STA carries out binary exponential backoff algorithm to randomize the transmission timing between other STAs to avoid collisions. In binary exponential backoff, a transmitting STA generates a random number between 0 and CW, and memorize the number in a counter. The STA decrements the counter value every after a specified time period defined as SlotTime as long as the channel is idle, and starts transmission when the counter value reaches zero. The initial value of CW is $CW_{\min}$ and doubled every time the STA repeats retransmission until it reaches $CW_{\max}$. Therefore, the CW value for a frame in the n_{th} retransmission stage is given by Eq. (2.1).

$$CW(n) = \begin{cases} (CW_{\min}+1)\cdot 2^{n-1}-1 & \text{for } CW(n) \le CW_{\max}, \\ \\ CW_{\max} & \text{otherwise.} \end{cases} \tag{2.1}$$

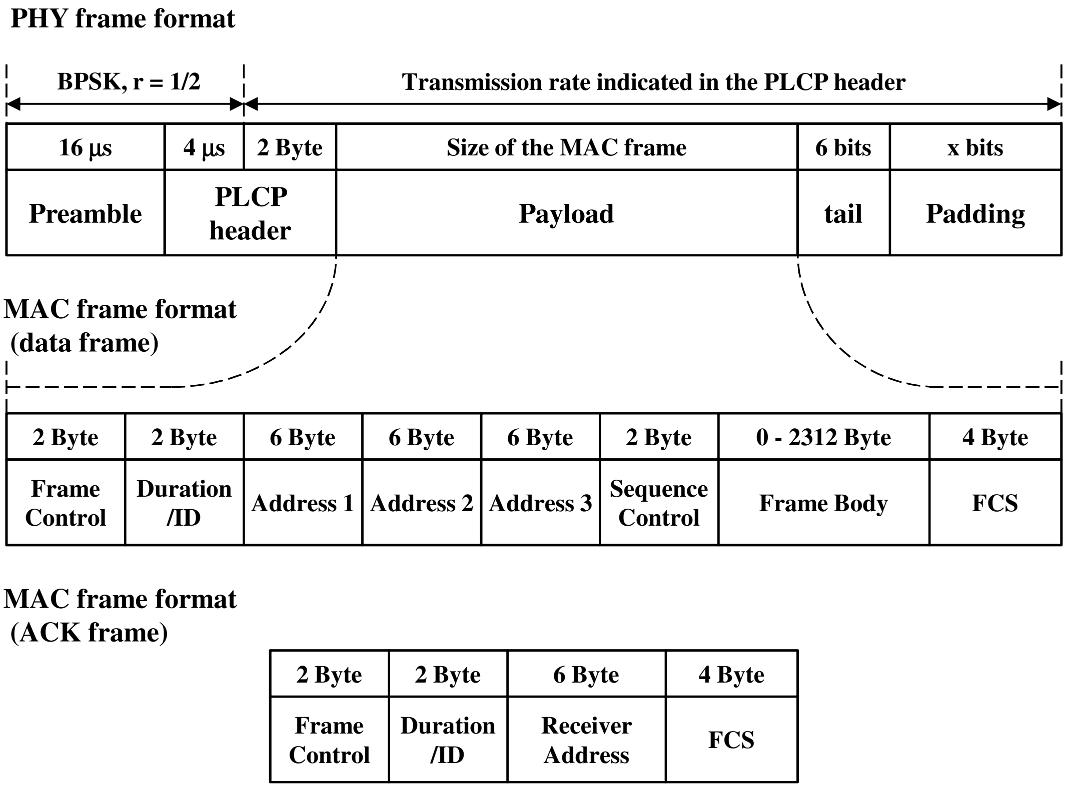

Fig. 2.5 Format of the IEEE 802.11 Data and ACK frames.

The destination of transmitted frame returns an ACK frame SIFS time after completion of reception. The sender of the frame determines that the transmission was successful when an ACK frame is returned, and the CW is reset to $CW_{\min}$. On the other hand, the sender regards the transmission to be failure if no ACK is returned within a specified time period called “ACKTimeout”, and in this case, the transmitting STA starts retransmission process.

In the IEEE 802.11a standard, DIFS, SIFS and SlotTime are determined to be 34 μs, 16 μs and 9 μs, respectively, and $CW_{\min}$ and $CW_{\max}$ are defined to be 15 and 1023, respctively. The ACKTimeout value had not specified in the IEEE 802.11 standard and is assumed to be 60 μs considering enough time for an ACK frame to be returned.

Figure 2.5 shows the frame formats of data and ACK frames of the IEEE 802.11

WLAN which are used for the throughput analysis. According to the IEEE 802.11a standard, the signal starts with the preamble whose length is 16 μs followed by the PLCP header. The MAC frame starts after the PLCP header followed by tail bits and padding bits.

By using those values, average throughput can be theoretically derived. Let us first calculate the time to complete a basic frame sequence. Transmission time for a 1500 byte data frame and for an ACK frame at each transmission rate is shown in Table 2.1. Transmission rate of an ACK frame is not allowed to be higher than the corresponding data frame. We assumed that the transmission rate of an ACK frame is the highest mandatory rate not exceeding the corresponding data frame. Therefore, possible transmission rates of an ACK frame are 6, 12 or 24 Mbit/s as in Table 2.1.

Table. 2.1 Transmisson time for a 1500 byte Data frame and an ACK frame.

Frame Type	Transmission Rate (Mbit/s)	Time (μs)
Data	54	248
	48	276
	36	364
	24	532
	18	704
	12	1044
	9	1384
	6	2064
ACK	24	28
	12	32
	6	44

Throughput with conventional rate switching algorithm is calculated as follows. First of all, throughput for a STA for which the maximum transmission rate is available is calculated dividing the amount of data by the time required for a frame sequence of DIFS - Backoff - Data - SIFS - ACK. We denote the throughput for such a

STA by $S(r_{\max})$ and expressed by the following equation;

$$S(r_{\max}) = \frac{L_{\mathrm{d}}}{T_{\mathrm{D}} + \overline{CW} + t(r_{\mathrm{dm}}) + T_{\mathrm{S}} + t(r_{\mathrm{am}})}. \tag{2.2}$$

L_{d} denotes the data size [bit]. T_{D} and T_{S} are the time duration of DIFS and SIFS, respectively. $t(r_{\mathrm{dm}})$ and $t(r_{\mathrm{am}})$ are the time durations to send a data frame at the maximum speed and the corresponding ACK frame, respectively. $\overline{CW}$ is the average backoff time and can be derived by $(CW_{\max} \cdot t_{\mathrm{SL}})/2$, where t_{SL} is the slot time.

When an AP or a STA is communicating using a transmission rate other than the highest rate, the AP or STA repeats U successful data transmission at a transmission rate $r_{d(i)}$ and D unsuccessful data transmissions at transmission rate of $r_{d(i+1)}$. Therefore, the throughput $S(r_i)$ is expressed by the following equation;

$$S(r_i) = \frac{U \cdot L_{\mathrm{d}}}{U \cdot t_{\mathrm{s}}(r_i) + D \cdot t_{\mathrm{f}}(r_{i+1}) + t_{\mathrm{bo}}}, \tag{2.3}$$

where $t_{\mathrm{s}}(r_i)$ is a time duration of a frame sequence when the data frame is transmitted using data rate r_i.

$$t_{\mathrm{s}}(r_i) = T_{\mathrm{D}} + \overline{CW} + t(r_{\mathrm{d}_i}) + T_{\mathrm{S}} + t(r_{\mathrm{a}_i}). \tag{2.4}$$

In Eq. (2.4), $t(r_{\mathrm{d}_i})$ is the time duration to transmit a data frame using the transmission rate $r_{d(i)}$, and $t(r_{\mathrm{a}_i})$ for an ACK frame using transmission rate corresponfing to the transmission rate of $r_{d(i)}$.

$t_{\mathrm{f}}(r_{i+1})$ denotes the time duration of a frame sequence that failed to send a data frame using a transmission rate $r_{d(i+1)}$ which is the higher transmission rate next to $r_{d(i)}$ without time for binary exponential backoff (BEB), and expressed as following equation;

$$t_{\mathrm{f}}(r_{i+1}) = T_{\mathrm{D}} + t(r_{d_{i+1}}) + T_{\mathrm{AT}}. \tag{2.5}$$

In Eq. (2.5), T_{AT} is the ACKTimeout interval which is the maximum time interval that a transmitter of a data frame waits for an ACK frame after sending a data or a management frame that requires a response. If the reception of the ACK frame does not start within this time interval, the transmitting STA concludes that the previous transmission was unsuccessful and starts retransmission procedure.

t_{bo} is the average deferral time for the binary exponential backoff procedure. After every unsuccessful transmission of the data frame, the transmitter basically doubles the range of contention window unless the contention window reaches the maximum value;

$$t_{\mathrm{bo}} = \frac{1}{2} \sum_{k=1}^{D} \left\{ (CW_{\mathrm{min}} + 1) \cdot 2^k - 1 \right\} \cdot t_{\mathrm{SL}}. \tag{2.6}$$

From the above equations, throughput performance of the proposed scheme is derived as follows. The proposed scheme has the same throughput as the conventional scheme for the highest MCS is available. A transmitter using proposed rate switching scheme changes the transmission rate after U successful transmissions just like the conventional scheme, however, it switches back to the previous transmission rate if a transmission failure occurs before RFC reaches to the threshold F. Therefore $S(r_i)$ with the proposed rate switching scheme is expressed as follows;

$$S(r_i) = \frac{U \cdot L_{\mathrm{d}}}{U \cdot t_{\mathrm{s}}(r_i) + t_{\mathrm{f}}(r_{i+1}) + t'_{\mathrm{bo}}}. \tag{2.7}$$

In Eq. (2.7), $t_{\mathrm{s}}(r_i)$ and $t_{\mathrm{f}}(r_{i+1})$ are the same as in Eqs. (2.4) and (2.5), respectively. The deferral time for the backoff procedure t'_{bo} is obtained by setting $D = 1$ in Eq. (2.6).

$$t'_{\mathrm{bo}} = \frac{2(CW_{\mathrm{min}} + 1) - 1}{2} \cdot t_{\mathrm{SL}}. \tag{2.8}$$

As it can be seen in the above equation, overhead induced by the retransmission procedure can be reduced by the rate fix threshold F.

Figures 2.6 and 2.7 show the results of throughput calculations. Figure 2.6 shows the throughput performance when the rate up threshold U is set to 12 with different values of the rate down threshold D. The payload size of the data frame is assumed to be 1500 Byte. In this figure, the throughput curve with ideal rate switching shows the reference of throughput performance when there is no degradation caused by the rate switching.

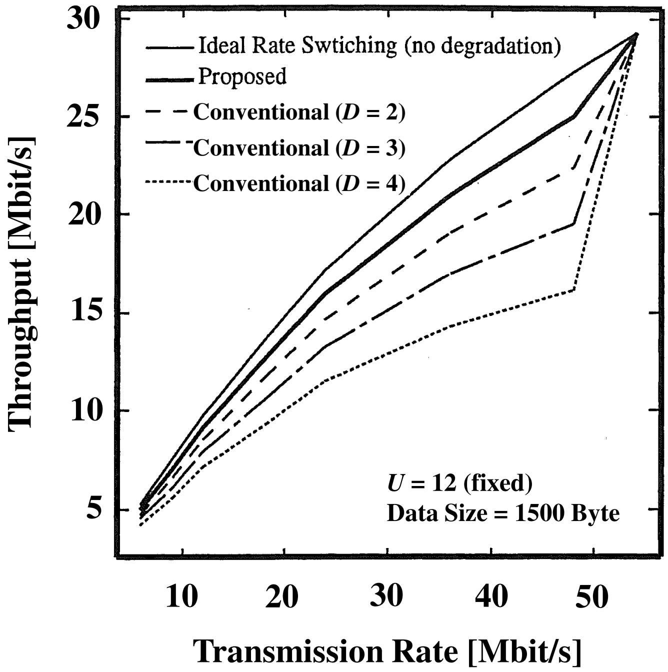

Fig. 2.6 Throughput performance with a fixed Rate Up Threshold.
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The curve with the highest performance is the throughput with ideal rate switching, i.e., no degradation caused by rate switching. This is a reference showing the performance limit.

The thick line marked as "Proposed" shows the throughput performance with the proposed rate switching algorithm and the other curves are throughput using conventional rate switching algorithm with different values of rate down threshold D. When using the proposed rate switching algorithm, a STA changes the transmission rate back to the previous one not depending on the rate down threshold D but depending on the rate fix threshold F if the newly selected transmission rate was too high. Therefore the throughput degradation caused by the rate switching can be minimized.

On the other hand, a STA using the conventional rate switching algorithm chooses lower transmission rate after D continuous transmission failures. Therefore, the impact of the rate down threshold D on the throughput performance becomes bigger as the value of D increases. This is because the time duration to switch to the lower transmission rate increases as the value of D becomes large. Therefore a smaller value of D has less impact on the throughput degradation. However, the system may not be efficient in some environments when the value of D is too small because the choice of the transmission rate will easily be affected by low signal to interference and noise ratio (SINR) and/or collision of the packet.

The proposed scheme is able to keep the high throughput by introducing rate fix threshold F so that the transmitter can quickly switch back to the previous lower transmission rate when the signal quality is not enough after choosing a higher transmission rate. Comparing the throughput of the proposed rate switching algorithm with the one using the conventional algorithm, the proposed scheme can improve the throughput from 11% to 27% when the rate down threshold D is 3.

Figure 2.7 shows the throughput performance with the fixed value of the rate down count $D = 3$ when the values of the rate up threshold U changed. Since the criteria of choosing a higher data rate is the same for the proposed and the conventional rate switching scheme, both of the throughput performances are affected by the rate up threshold U.

Compared with the performances shown in Fig. 2.7, the change of the rate up threshold U has less impact on the throughput performance than the change of the rate down threshold D. As the value of U becomes smaller, a transmitter aggressively chooses the higher transmission rate and the probability of transmission error after selecting a higher transmission rate becomes higher. This is the reason why the throughput with smaller value of U is lower. Comparing the throughput performance when the U is 16, in which case the difference between the proposed and the conventional schemes is the smallest, the proposed scheme offers 8 to 20% higher throughput than the conventional scheme.

Choosing an appropriate value of the rate fix threshold F is important. The condition to continue to use the new transmission rate will become more stringent when the value of F becomes larger. Therefore, a STA easily switches back to the previous

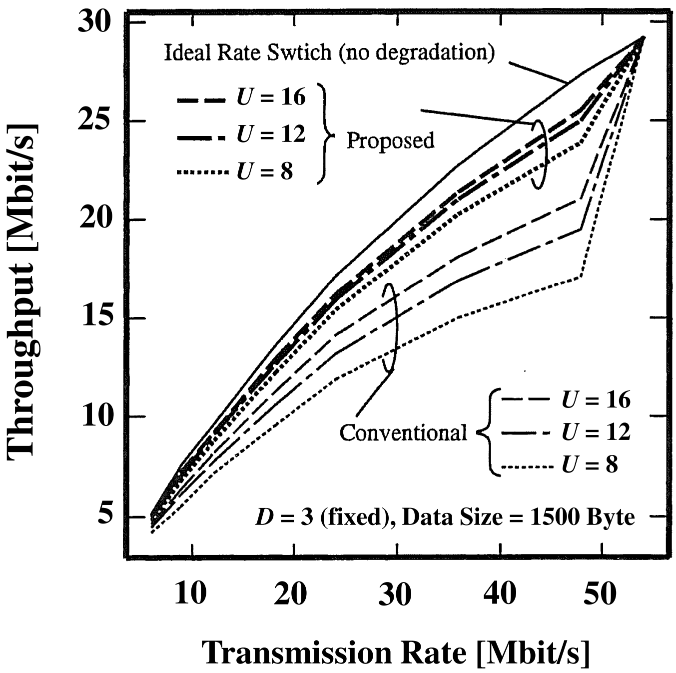

Fig. 2.7 Throughput performance with fixed Rate Down Threshold.
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lower transmission rate due to a packet error caused by an instantaneous degradation of signal quality or a packet collision even though the signal quality between the transmitter and receiver is improved, for example, by the STA's movement. Therefore, the throughput and system capacity becomes lower as the value of F becomes larger. When choosing the smaller value of F, on the other hand, the condition to continue to use the new transmission rate after switching to the higher one is relaxed. However, in this case, it takes more time for the recovery of choosing a too high transmission rate. Therefore, an optimization considering both the rate fix threshold F and the rate down threshold D will be important.

2.5.2 Impact of Rate Switching Algorithm on the System Capacity

The impact of rate switching algorithm on the system capacity is evaluated. For this purpose, we assumed a simplified model of service area as depicted in Fig. 2.8, i.e., the communication areas where different transmission rates are available are formed concentrically around the AP. In this figure, $a_0, a_1, \cdots, a_7$ indicate the size of areas in which maximum transmission rate of 54 Mbit/s, 48 Mbit/s, $\cdots$, 6 Mbit/s are available, respectively. We derive the system capacity by averaging the throughput of each area in the entire service area.

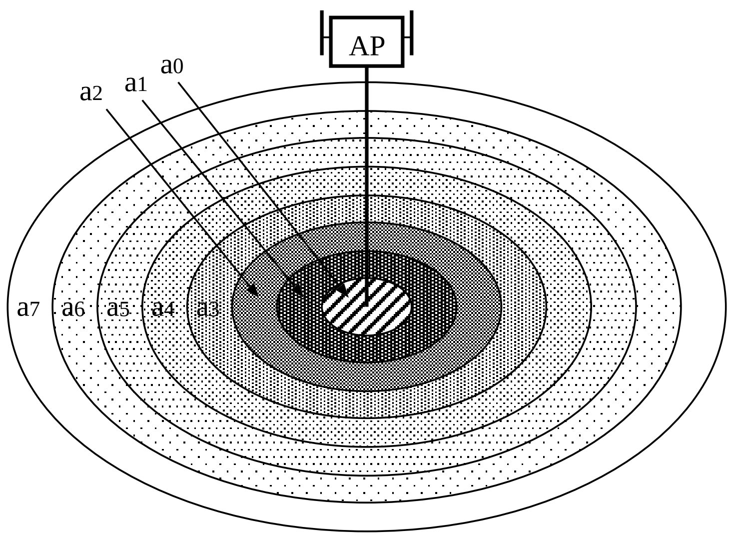

Fig. 2.8 Service area of the IEEE 802.11a WLAN system.

In order to derive the system capacity, we calculate size of each area. To avoid complexity, we used a simplified model to calculate the radius of each area by considering the transmission power, antenna gains and path loss.

The path loss model of the 5 GHz band channel which is used by the IEEE 802.11a wireless LANs has been standardized by the ITU-R recommendation to have the break point at 1 m and to have the path loss exponent of 3.1 [18]. In this case, the path loss at a point from distance x m from the access point is expressed by the following equation;

$$L(x) = 20 \cdot \log(\frac{4\pi}{\lambda}) + 31 \cdot \log(x-1). \tag{2.9}$$

Therefore, the received signal power at a point x m apart from the access point is obtained by the following equation;

$$P_r(x) = P_\mathrm{t} + G_\mathrm{t} + G_\mathrm{r} - L(x), \tag{2.10}$$

where P_t, G_t, and G_r denote transmit power, transmit antenna gain and receive antenna gain, respectively.

Table. 2.2 Reqired received signal level for the IEEE 802.11a PHY.

Data Rate (Mbit/s)	Received Signal Power (dBm)	Range (m)
6	-82	37.0
9	-81	34.4
12	-79	29.8
18	-77	26.0
24	-74	21.0
36	-70	15.8
48	-66	12.0
54	-65	11.2

Table 2.2 shows the relationship between the transmission rate and the required received signal power defined in the IEEE 802.11a standard. Range is calculated from Eq. (2.10) by assuming transmission power of 13 dBm and 0 dBi for both transmit and receive antenna gains.

From the above assumptions, we analytically derive the system capacity. For simplicity, we assumed a uniform distribution of clients in the service area. The number of clients that communicate using each transmission rate is proportional to the size of corresponding area. Therefore, the system capacity C_{c} using the conventional rate switching algorithm is expressed as follows;

$$C_{\mathrm{c}} = \frac{\sum_{i=0}^{n-1} a_i \cdot U \cdot L_{\mathrm{d}}}{\tau_{\mathrm{c}}}, \tag{2.11}$$

where

$$\tau_{\mathrm{c}} = U \cdot \sum_{i=0}^{n-1} a_i \cdot t_{\mathrm{s}}(r_{n-i-1}) + D \cdot \sum_{i=1}^{n-1} a_i \cdot t_{\mathrm{f}}(r_{n-i}) + \sum_{i=1}^{n-1} a_i \cdot t_{\mathrm{bo}}. \tag{2.12}$$

In the above equations, n denotes the number of transmission rates defined in physical layer and r_i is the transmission rate, where $r_0 < \cdots < r_i \cdots < r_{n-1}$.

On the other hand, the system capacity with the proposed rate switching algorithm C_p is expressed by the following equation;

$$C_{\mathrm{p}} = \frac{\sum_{i=0}^{n-1} a_i \cdot U \cdot L_{\mathrm{d}}}{\tau_{\mathrm{p}}}, \tag{2.13}$$

where

$$\tau_{\mathrm{p}} = U \cdot \sum_{i=0}^{n-1} a_i \cdot t_{\mathrm{s}}(r_{n-i-1}) + \sum_{i=1}^{n-1} a_i \left\{ t_{\mathrm{f}}(r_{n-i}) + t'_{\mathrm{bo}} \right\}. \tag{2.14}$$

Figure 2.9 shows the system capacity calculated from Eqs. (2.11) and (2.13). The x-axis is the rate up threshold U and the y-axis is the system capacity. The thick line shows the system capacity with the proposed rate switching algorithm and the dashed line shows the capacity with the conventional rate switching with rate down threshold $D = 3$. The thin line on the top shows the system capacity with an ideal rate switching algorithm where no degradation is caused by the rate switching algorithm.

The results show that the system capacity can be enhanced by using the proposed rate switching algorithm. As the value of U increases, the difference between the proposed and the conventional rate switching becomes smaller. This is because the STAs have less attempts to use a higher transmission rate as U gets larger. To

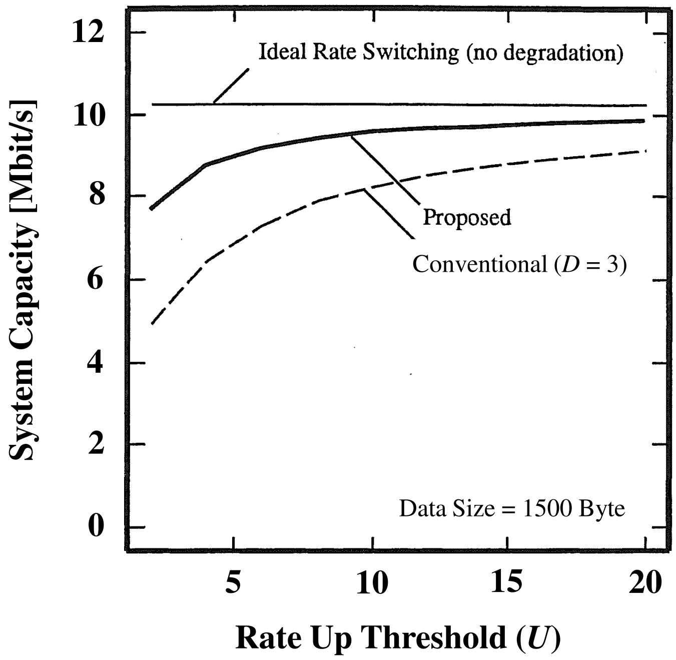

Fig. 2.9 Comparison of the systems capacity.
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achieve the same capacity as the proposed rate switching scheme, the conventional rate switching needs to have large value of U which is not realistic to adapt to the change of radio propagation environment. Taking the close look at the realistic region of $8 \geq U \geq 12$, the proposed rate switching algorithm can enhance the system capacity by 14% to 20% compared to the conventional one.

2.6 Summary

In this chapter, a novel rate switching algorithm for the IEEE 802.11 WLANs is proposed. The conventional rate switching algorithm with which the transmission rate is selected from the history of previous transmission attempts had an issue of performance degradation due to the overhead. With the proposed rate switching algorithm, a STA carefully monitors the results of transmission attempts especially after a higher transmission rate is selected and switches back to the previous transmission rate if the signal quality is not enough for the newly selected rate. Therefore, performance degradation due to the rate switching algorithm can be minimized with the proposed scheme. The effect of parameters used in the conventional and the proposed rate switching algorithms on the throughput performance is theoretically analyzed. We also evaluated the system capacity for the cases of using the proposed and the conventional rate switching algorithms. From the results of the capacity analysis, it was shown that the proposed rate switching algorithm can improve the throughput performance by 20 to 28% and can improve the system capacity by 14 to 20%.

Chapter 3

Communication Quality Control Schemes for Wireless LANs

3.1 Overview

As a way of providing Class of Service (CoS) over the Ethernet, a communication quality control scheme based on user priority (UP) is widely adopted. The UP was originally defined by the IEEE 802.1p standard and now incorporated into the IEEE 802.1D Annex H.2 [19]. It is also desired that the priority control scheme is available in the wireless communication network to support applications such as voice over IP and video which are sensitive for the communication quality. This chapter presents CoS control scheme for the IEEE 802.11 wireless LANs. Two kinds of access mechanisms based on PCF and DCF defined in the IEEE 802.11 standard are presented to provide parameterized QoS and prioritized QoS services, respectively. The parameterized QoS is to ensure specific communication parameters such as bandwidth, delay, and jitter in a managed environment, while prioritized QoS is a general QoS scheme to make traffic differentiation based on the priority of the data. From the results of computer simulations, it is shown that the priority control is possible under contention based access where data transmission services are provided in a best effort basis. Finally, it is shown that the proposed scheme effectively suppresses the bandwidth for a greedy user who occupies the bandwidth by sending and/or receiving great amount of data.

3.2 Introduction

In recent IP networks, communication quality control is one of the important features in order to deal with various requirements on communication quality of various applications such as e-mail, web browsing and transport of voice and video streams. The communication quality control schemes can generally be divided into two categories. One is Quality of Service (QoS), which is to assure the required communication quality parameters such as bandwidth, delay and/or jitter faithfully based on the requirements from the applications. The other one is Class of Service (CoS), in which the data transport service is divided into multiple service classes just like the flight seats of first class, business class and economy class, and the communication quality of corresponding service class is provided. In CoS schemes, data flows belonging to a higher service class have better service quality compared to the flows belonging to a lower service class.

As the ways of supporting QoS and/or CoS, the Resource Reservation Protocol (RSVP) [20, 21] and the Differentiated Services (DiffServ) [22] are used at layer 3 and upper levels. The Multi-Protocol Label Switching (MPLS) [23] is used at layer 2 and 3 levels, and procedures based on the user priority (UP) defined by the IEEE 802.1D Annex H.2 are used at layer 2 level. Many products supporting those protocols have already been available and have been actually installed in the public and private networks.

QoS/CoS should be offered between the end points of a data flow. For the QoS support, therefore, signaling protocols are used to reserve the communication resources between the source and the destination nodes and to do the necessary scheduling at the intermediate nodes. For this reason, the amount of control traffic and processing load at each node increases explosively with the increase of the network scale and the number of data flows. In the CoS, in contrast, the assurance of communication quality parameter(s) and/or the priority control are provided according to the predefined service classes. Because the number of the service classes is limited, the CoS is regarded as a simple and realistic way of assuring communication quality parameters and/or the prioritization of the data flows.

Recently, high speed WLANs such as the IEEE 802.11a/g/n/ac/ad have been standardized and data rates of more than 1 Gbit/s are available also for the wireless communications. As a higher transmission rate becomes available, demands for transmissions of not only the DVD and the Blu-ray but also the 4K and 8K videos which require much higher communication quality are increasing. In such a case, the wireless LAN system is desired to have mechanisms for communication quality control to provide better service quality for those video traffic collaborating with the QoS mechanisms of the external networks.

In order to provide the communication quality control, the wireless LAN system needs to have following functions.

- The classification function of incoming data.
- The queuing and the scheduling functions based on the priority information.
- The access control function to send data frames to the destination, assuring specific communication quality parameters or making the service differentiation.

In the IEEE 802.11 standard, two kinds of channel access functions are specified which are the DCF and the PCF. The DCF is the basic access procedure for the IEEE 802.11 wireless LANs in which each STA makes its own decision to start transmission of a data frame on the channel based on the CSMA/CA protocol. Because of the distributed feature of the random access protocol, the DCF is not suitable for assuring specific communication quality parameters required by a data flow. However, it is able to make priority controls between the data of different service classes if it is enhanced to consider the priority of the data. In such a case, the prioritized QoS is provided. The PCF is a centralized access function in which AP controls the transmission of all STAs associated to it using the polling procedure. Although it is specified as an optional procedure on top of the DCF, the PCF is suitable for assuring specific communication quality parameters, i.e. supporting the prioritized QoS in a managed environment. The details of the scheduling and management functions of the PCF, however, are left to the individual vendors.

This chapter proposes communication quality control schemes to support parameterized QoS based on the PCF and to support prioritized QoS based on the DCF. We assumed that there are two kinds of data, i.e., the data which require the assurance of the communication quality parameters such as the bandwidth guarantee and the bounded delay, and the data which do not. We also assumed that the information about the communication quality based on the IEEE 802.1D Annex H.2 is contained in the header field of the layer 2 frames. Since the IEEE 802.1D is the specification to be applied to the bridge device, the AP plays that role in the IEEE 802.11 wireless LAN network. The communication quality is controlled by applying the PCF and the DCF for transmissions of those data. For the use of PCF based access scheme, a management procedure is proposed to control the period of using the centralized channel access according to the amount of data requiring the communication quality assurance. Moreover an access control mechanism based on the DCF is proposed to make service differentiation between the service classes which do not require assurance of communication quality parameters.

The rest of this chapter is organized as follows. System configuration assumed in this study is described in section 3.3. Relation between the communication quality control schemes and the IEEE 802.11 MAC protocol is discussed in section 3.4. Section 3.5 explains the proposed schemes and shows simulation results of evaluating the proposed protocol. Section 3.6 concludes this chapter.

3.3 System Configuration

As described in the previous section, the priority information of the data based on the user priority defined in the IEEE 802.1D Annex H.2 is contained in the header field of the layer 2 frames. Figure 3.1 shows an example of the Ethernet frame in which the user priority is contained in the priority code point (PCP) field of the IEEE 802.1Q header.

It is assumed that the AP has a classification function of the downlink data packets coming from the external wired network according to the priority information in the

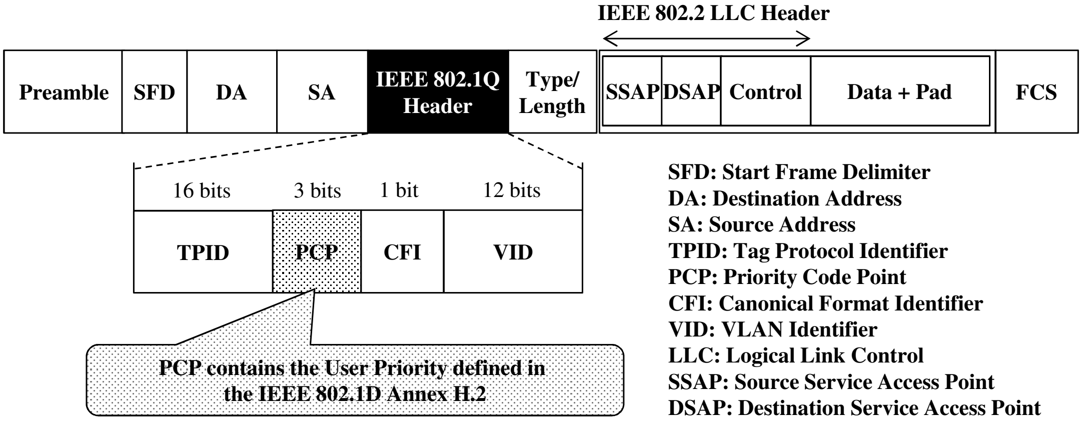

Fig. 3.1 User priority information contained in an Ethernet frame.

header field. The AP is also assumed to be equipped with multiple queues corresponding to the service classes. The AP enqueues the incoming down link data into the corresponding queue to make service differentiation between the traffic flows belonging to the different service classes. Furthermore, the AP maintains the scheduling list for the uplink and downlink transmissions when the PCF is assumed.

Figure 3.2 shows the detailed system architecture. The procedure for the downlink data transmission, i.e. from AP to STA, is described in Step 1 to 3.

Step 1 The AP detects the priority information contained in the header field of the received data frame and enqueues it in the queue of the corresponding service class.

Step 2 A packet is taken out from the queue considering the priority.

Step 3 By using the proposed access control schemes, which are extended access control mechanisms of the IEEE 802.11 MAC protocol, the downlink transmissions are carried out considering the service class of data.

The procedure for the upward data transmission, i.e. from STA to AP, is as shown in Step 4 to 7.

Step 4 STA notifies the priority class of the data to be sent to AP.

Step 5 AP manages the priorities of data notified by a STA and maintains the

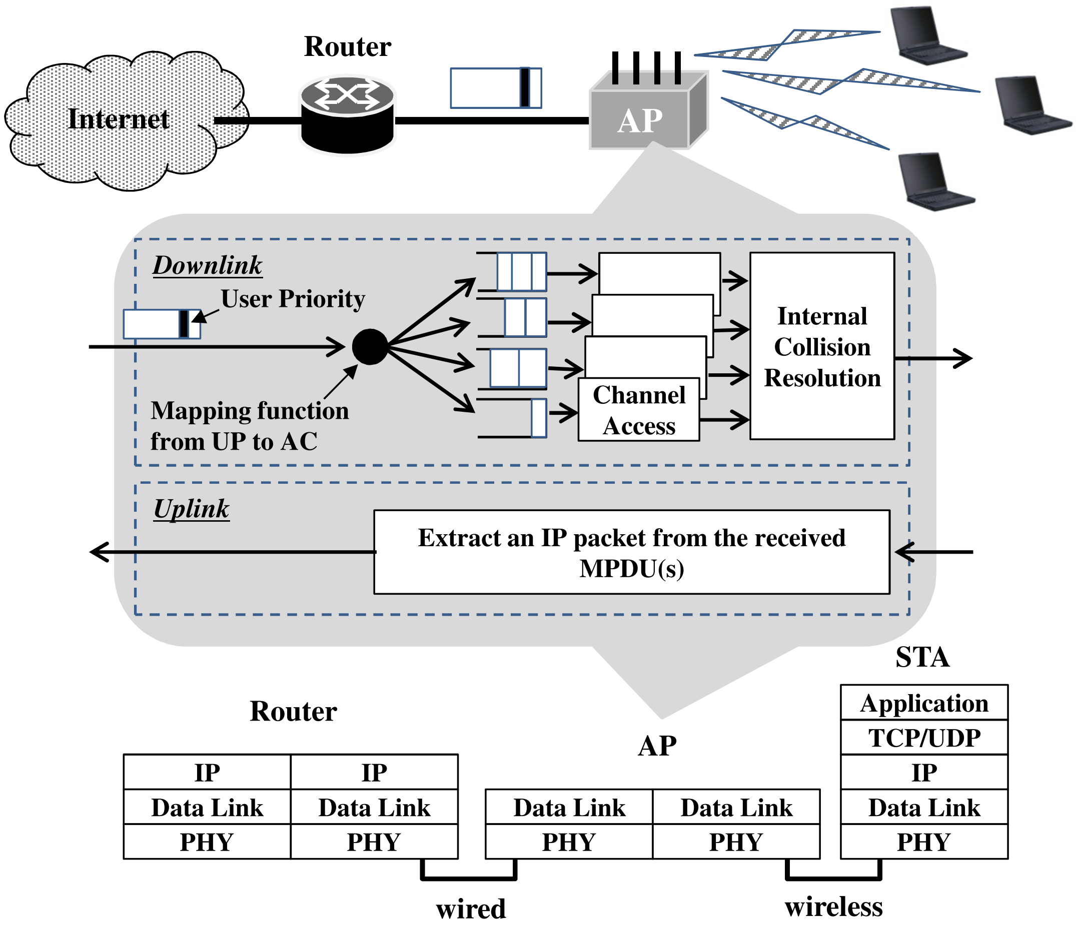

Fig. 3.2 System architecture.

scheduling list.

Step 6 According to the scheduling list, the AP controls the duration of CFP and manages the order of the STAs to permit data transmissions.

Step 7 By using the polling-based access control of PCF, the uplink transmissions are carried out ensuring communication parameters. For the priority control not ensuring any communication parameters, an enhanced random access control scheme is used outside the PCF.

By considering the amount of the downlink and the uplink data, the AP determines the duration to use PCF and frequencies of the uplink and the downlink data transmissions, and makes priority-aware data transmissions for both downlink and uplink.

3.4 IEEE 802.11 Standard and Communication Quality Control Schemes

The IEEE 802.11 wireless LAN standard specifies two kinds of access control procedures. One is the DCF in which STAs including the AP carry out the distributed access control based on the CSMA/CA protocol for a transmission of a data frame. The time period in which the STAs follow the DCF procedure is called contention period (CP).

The other one is the centralized access control method called PCF in which the AP controls all transmissions and a STA is allowed to send its data after receiving a polling frame. The time period in which PCF is used for data transmissions is called contention free period (CFP).

The CFP and the CP alternate with a constant interval called CFP interval as shown in Fig. 3.3. A CFP starts with a transmission of a Beacon containing an information element announcing the start of CFP, and, by definition, a CFP is allowed to expand for more than one Beacon interval. The CFP Interval and the Beacon interval become the same when every Beacon contains the information element.

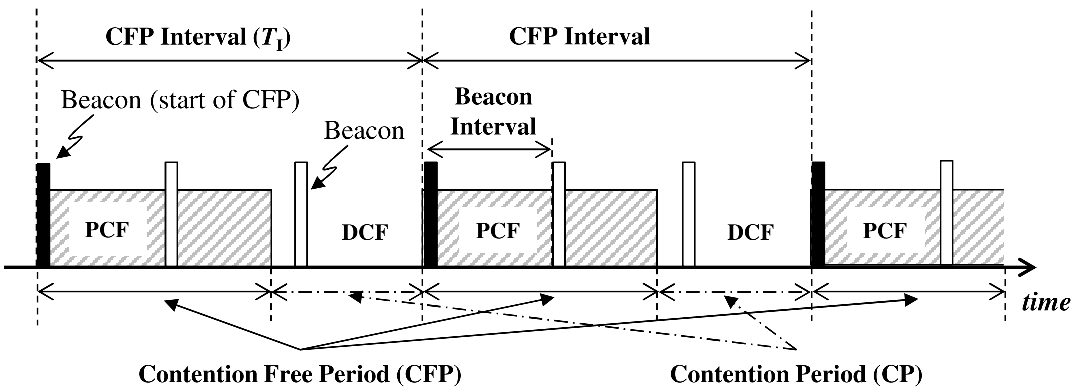

Fig. 3.3 Relation between CP and CFP on the IEEE 802.11 channel.

Characteristics and issues of the communication quality controls by using the PCF and DCF procedures are described as follows.

(1) Challenges of the PCF for communication quality control

PCF is the centralized control, and is suitable for ensuring the communication quality. During the PCF is activated, all transmissions of the STAs are controlled by the AP. Although, in this case, the management and scheduling functions for the PCF becomes very important to ensure specific communication quality parameters, details of such functions are not specified in the IEEE 802.11 standard. In order to use the PCF procedure for data transmissions considering the QoS requirements, additional functions, such as the management function of data traffic and its QoS characteristics, and scheduling functions for both uplink and downlink data considering the QoS requirements, need to be implemented.

There have been some proposals of the communication quality control using PCF [26–29]. [26] and [27] discuss the ways for STAs to transmit the voice or video data by using the PCF procedure. The AP ensures the communication quality of the uplink data from such STAs by managing the polling list and by scheduling the uplink data transmissions. However, the details of the scheduling mechanism for the downlink transmissions are not discussed in those papers. It is assumed that the downlink transmission should follow the algorithm defined by the IEEE 802.11 standard. The specification of the PCF procedure in the IEEE 802.11 standard allows the data to be piggybacked to the polling and/or the ACK frames. Therefore, the downlink data can be sent in conjunction with the polling frame when the destination of the data frame and the STA that the AP is allowing an uplink transmission is the same. Therefore, this scheme is effective for the bi-directional flow such as voice data and video phone. If we think about more general scheduling scheme, it is necessary to consider the communication parameters of not only the uplink but also the downlink data flows to satisfy their QoS requirements.

[28] proposes a way to support the voice and the data communications simultaneously. The voice STAs transmit data frames using the PCF procedure and the data

STAs transmit data frames using the DCF procedure. In this scheme, however, the downlink transmissions of voice packets are not detailed and it is anticipated that the same assumption as [26] and [27] have been made. Therefore, it is also true for this work that the scheduling function is important for the downlink voice packets to satisfy the required QoS parameters.

[29] proposes a mechanism to cope with data flows with different bandwidth requirements. With a scheduling method considering the state of wireless channels between the AP and STAs, the AP is able to guarantee the required bandwidth. This scheme intends to guarantee the minimum bandwidth for the downlink data with a fixed PCF duration. Therefore, it is difficult to flexibly adapt to the changes in the amount of data traffic and to deal with data flows having different types of QoS requirements.

From the discussions above, it can be said that a management and scheduling functions, which is different from the schemes proposed in [26–29], are needed.

(2) Challenges of the DCF for communication quality control

In the DCF procedure, all STAs including the AP follow the CSMA/CA protocol for a data transmission. In the CSMA/CA protocol, the STA contends with other STAs for the opportunity to send a data frame. Therefore, it is difficult to satisfy the requirements on specific communication quality parameters. However, it is possible to make service differentiations between the data flows belonging to different service classes.

The priority control schemes based on the DCF have been proposed in [30–32]. In [30] and [31], STAs having data to transmit start transmissions of pulses with a specified length successively after the channel is confirmed to be idle. The number of the pulse becomes larger as the priority of the data becomes higher. Therefore, the STA having data with the highest priority starts the data transmission. Although this is a way of making priority controls, the problems in [30] and [31] is the increased overhead due to pulse transmissions.

[32] proposes a scheme where the range of the random variable generation is di-

vided according to the priority. This is another way of making the priority control, however, there is a potential problem that the collision probability is increased when the number of contending STAs is increased which degrades efficiency due to the increased retransmissions.

This study intends to provide CoS over the wireless LAN, by using additional functions with the PCF to ensure required communication parameters and by enhancing the DCF procedure to have priority control which are described next.

3.5 Proposed Communication Quality Control Scheme

The communication quality control scheme proposed in this thesis utilizes both DCF and PCF defined in the IEEE 802.11 wireless LAN standard. For the communication quality, the user priority defined in the IEEE 802.1D Annex H.2 is assumed. A management scheme is proposed for the PCF to determine the duration of CFP adaptively to deal with the data flows requiring the assurance of different communication qualities efficiently. Moreover, the proposed scheme considers the balance between the uplink and the downlink traffics to satisfy the requirements on the communication quality for both of the uplink and the downlink. With regard to the DCF, an enhancement for the DCF procedure is proposed where the binary exponential backoff of the CSMA/CA protocol is improved, so that services for the flows belonging to the different service classes can be differentiated.

Seven service classes are defined in IEEE 802.1D Annex H.2, as shown in Table 3.1. They are the "Background" (BK) class which has the lowest priority, the "Best Effort" (BE) class which is the default service class for the best effort traffic, the "Excellent Effort" (EE) class, which is also for the best effort traffic, but with higher priority. The "Controlled Load" (CL) class is for the data to assure the minimum bandwidth, and the "Video" and the "Voice" (VV) classes are used for the data requiring the assurance of the maximum delay and/or jitter. The "Network Control" traffic has the highest priority. Those classes are roughly divided into the two categories. One is the service class requiring assurance of certain communication quality parameters (VV and CL classes). The other one is the service classes which simply require

Table. 3.1 Priority defined in the IEEE 802.1D Annex H.2.

User Priority	Traffic Type	Characteristics
1	Background	Bulk transfer and other activities permitted on the network
2	Spare	—
0 (default)	Best Effort	Best Effort
3	Excellent Effort	Best Effort to the most important customers
4	Controlled Load	Bandwidth Reservation
5	Video	$\leq$ 100 ms latency and jitter
6	Voice	$\leq$ 10 ms latency and jitter
7	Network Control	Maintain and Support the network infrastructure

priority control (BK, BE and EE classes). The proposed scheme is explained for the cases of the communication quality control to assure the required parameters and communication quality control to make the priority control. The access control schemes for each of those schemes are discussed. In the following discussions, data or a packet containing data belonging to a service class is denoted with its service class, e.g. data or a packet containing data belonging to the VV class is called VV data or a VV packet. Similarly, a STA transmitting and/or receiving VV data or VV packets is called a VV STA.

3.5.1 Controls for the Guaranteed Flows

3.5.1.1 Proposed Control Scheme

VV and CL are the service classes that require assurance of a certain communication quality. Consequently, data packets belonging to the service classes of VV or CL are transmitted by using the PCF procedure. As shown in Fig. 3.2, the scheduling for the downlink data is executed using the queues corresponding to the service classes. As for the order of taking out a packet from the queues, a VV packet requiring assurance of the maximum delay and/or jitter is given the highest priority. CL data are taken out from the queue when there are no VV data in the queue.

The schedules for the uplink data transmissions by using polling procedure are managed by the polling list of the AP considering the priority of the data that the associated STAs are trying to transmit. As for the order of polling, the VV STAs have the highest priority. The AP polls the VV STAs considering the interval to satisfy the requirements on the maximum delay and/or jitter. For the CL STAs, the AP controls the interval to poll to satisfy the requirements of the minimum bandwidth.

In the proposed scheme, furthermore, the balance between the uplink and the downlink traffics of VV and CL data is examined for each CFP interval, and the ratio of the transmission frequency of the uplink data M_{up} and transmission frequency of the downlink data M_{down} is determined. Assuming $(M_{\mathrm{up}} + M_{\mathrm{down}})$ as a unit, M_{up} times of uplink and M_{down} times of downlink transmissions of the VV and the CL data are executed.

3.5.1.2 Configuration of CFP

The duration of a CFP is determined adaptively according to the amount of data requiring assurance of communication quality parameters. In the proposed scheme, the AP checks the service class and the number of the packets coming from the wired network. The AP also checks the service class of data packets that the associated STAs are trying to transmit for each CFP Interval. Based on the result of those examinations, the length of CFP in the next CFP Interval is determined. Figure 3.4 shows the procedure to determine the duration of CFP.

Since the PCF and the DCF alternate as shown in Fig. 3.3, the assurance of the communication quality may become impossible when the duration of a CP is longer than the maximum delay allowed for the VV data. Consequently, the existence of VV data in the AP's queue and existence of VV STAs on the polling list are used as the highest priority condition to determine the CFP duration. In other words, if VV data exist in the queue or a VV STA exists in the polling list, CFP is set so that the

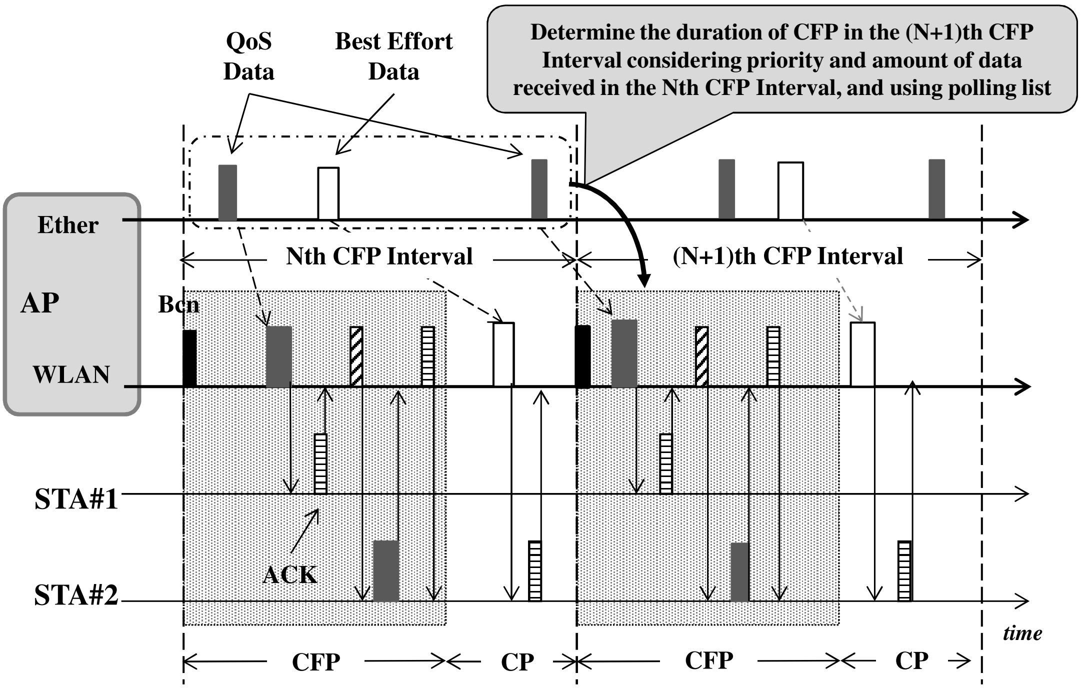

Fig. 3.4 Determination of CFP duration.

maximum delay is satisfied. The control schemes for the following cases are discussed.

Case 1) Controls for the VV data and the VV STAs,

Case 2) Controls for the CL data and the CL STAs when no VV data exist in the AP's queue and no VV STA exits on the polling list, and

Case 3) Control for the case where neither VV nor CL data exist in the queue, and neither VV nor CL STA exists on the polling list.

T_{I} is the interval between the start time of two successive CFP, which is identical to the beacon interval when the CFP starts with every beacon, and is assumed to be fixed. Let us take a close look at the relationship between T_{I} and the maximum delay allowed for the VV data which is denoted T_{VV}. When $T_{\mathrm{I}} < T_{\mathrm{VV}}$, the requirement on the maximum delay can be satisfied for the downlink VV data arriving at AP in N-th CFP Interval, even if it is transmitted in $(N+1)$-th CFP Interval. When

$T_\mathrm{I} > T_\mathrm{VV}$, however, the requirement on the maximum delay may not be satisfied if CP exceeds T_VV. In other words, for the VV data arriving at the AP during the N-th CFP Interval, the requirement on the maximum delay cannot be satisfied, unless the VV data are transmitted within the CFP of the N-th CFP Interval. For the VV data arrivng at the AP during CP of the N-th CFP Interval, the requirement on the maximum delay cannot be realized, unless it is transmitted in the CFP of the $(N+1)$-th CFP Interval. Therefore, in the case of $T_\mathrm{I} > T_\mathrm{VV}$, a severe condition is posed for the packet scheduling compared to the case of $T_\mathrm{I} < T_\mathrm{VV}$. When we think about the actual use of the wireless LAN, the order of CFP Interval is 100 ms since a typical value for the beacon interval T_I is 100 ms while the order of the maximum delay and/or jitter allowed is 10 ms for the voice data and 100 ms for the video data. It is more general to assume that $T_\mathrm{I} > T_\mathrm{VV}$ to cope with the case of severe condition.

Case 1) Controls for the VV data and the VV STAs

Consider the case of Fig. 3.4, where VV packet arrives in N-th CFP Interval that can provide the bounded delay. Then, $(N+1)$-th CP is set so that it does not exceed T_VV, independently of the traffic. In other words, since the CFP Interval is fixed, the duration of CFP is also set to a constant value whenever VV data arrive at the AP during the previous CFP Interval.

When there is a VV STA on the polling list, the uplink transmissions of VV data for M_up times and the downlink transmissions of VV data for M_down times are accommodated within a unit period of $(M_\mathrm{up} + M_\mathrm{down})$. By this scheme, the VV STA transmits VV data at least once in a $(M_\mathrm{up} + M_\mathrm{down})$ transmissions.

The time to complete transmission of a downlink data packet (T_DOWN) is the time to transmit a data packet and the time to receive ACK for the packet. It is given by Eq. (3.1).

$$T_\mathrm{DOWN} = 2 \cdot T_\mathrm{SIFS} + T_\mathrm{data} + T_\mathrm{ACK}. \tag{3.1}$$

Note that T_SIFS is Short Inter-Frame Space (SIFS) time defined in the IEEE 802.11 standard, and T_data is the time to transmit a data packet. T_ACK is the transmission time for the ACK.

The transmission of the uplink data is allowed only for the STA which is polled by the AP. Consequently, the time to transmit an uplink data packet (T_{UP}) is the time obtained by adding the time to transmit a polling frame ($T_{\mathrm{POLL}} + T_{\mathrm{SIFS}}$) to ($T_{\mathrm{DOWN}}$). It is given by Eq. (3.2);

$$T_{\mathrm{UP}} = T_{\mathrm{POLL}} + T_{\mathrm{data}} + 3 \cdot T_{\mathrm{SIFS}} + T_{\mathrm{ACK}}. \tag{3.2}$$

Thus, the setting condition for CFP length (T_{CFP}) is given by;

$$\begin{aligned} T_{\mathrm{I}} - T_{\mathrm{CFP}} \quad < \quad & T_{\mathrm{VV}} - M_{\mathrm{up}} \cdot (T_{\mathrm{POLL}} + T_{\mathrm{data}} + 3 \cdot T_{\mathrm{SIFS}} + T_{\mathrm{ACK}}) \\ & - M_{\mathrm{down}} \times (2 \cdot T_{\mathrm{SIFS}} + T_{\mathrm{data}} + T_{\mathrm{ACK}}). \end{aligned} \tag{3.3}$$

In setting of CP and CFP, it may happen that CP may be extended for some reasons such as lack of time to complete a PCF data exchange sequence and a start of a transmission of BE data just before the start of the next CFP. Therefore, CFP duration is determined considering the case where CP is extended most.

When the remaining time of the CFP is shorter than the time to complete a downlink or an uplink data exchange sequence, and the time to send the CF-End frame to indicate the end of CFP, AP terminates CFP before a predetermined time. In this case, the CFP is shorten, or equivalently the CP is extended, at maximum by T_{S} shown by;

$$T_{\mathrm{S}} = 2 \cdot T_{\mathrm{SIFS}} + T_{\mathrm{data}} + T_{\mathrm{ACK}}. \tag{3.4}$$

As shown in Fig. 3.5, CP can also be extended when transmission of a data packet is started just before the end of CP. In this case, CP is extended by the time to send one packet at the maximum. Thus, DCF period can be extended by the period indicated as T_{L} in Fig. 3.5 at the maximum.

T_{L} is given by

$$T_{\mathrm{L}} = T_{\mathrm{SIFS}} + T_{\mathrm{data}} + T_{\mathrm{ACK}}. \tag{3.5}$$

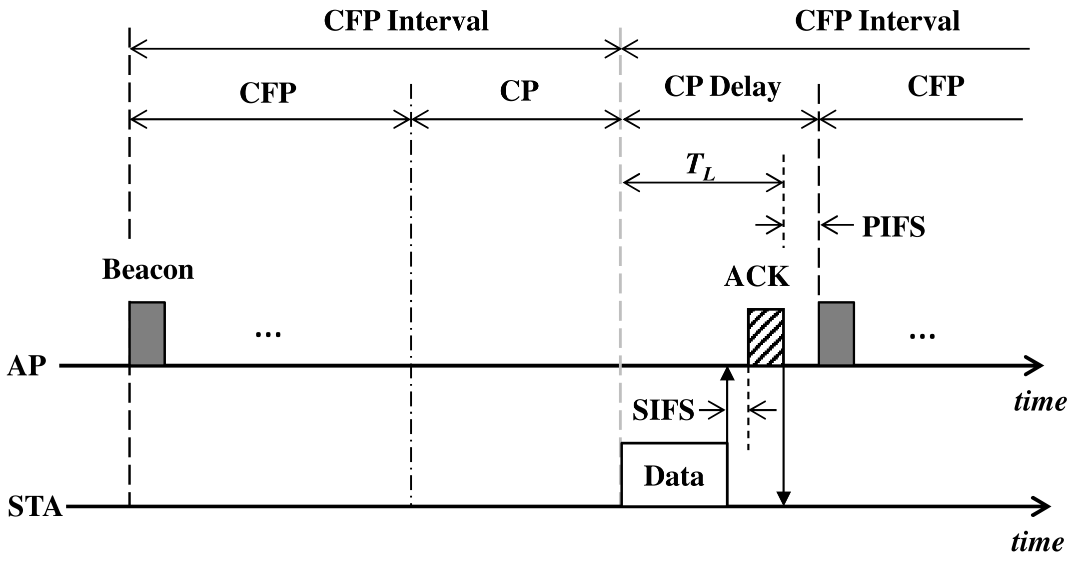

Fig. 3.5 Extension of CP.

As shown in Eqs. (3.4) and (3.5), CP can be extended in two cases. One is the case where the current CFP is terminated before the period that was originally planned. The other is the case where data are transmitted just before the end of CP. The maximum extension of CP, therefore, is $T_{\mathrm{S}} + T_{\mathrm{L}}$. Then, the maximum extension of CP is $(3 \cdot T_{\mathrm{SIFS}} + 2 \cdot T_{\mathrm{data}} + 2 \cdot T_{\mathrm{ACK}})$. It is also specified in IEEE 802.11 that the minimum duration of a CP shall be long enough to send a shortest data frame and to receive the ACK for the data.

Consequently, the setting condition for T_{CFP} is given by Eq. (3.6), which is obtained by combining the conditions shown in Eq. (3.3).

$$\begin{aligned}
& 2 \cdot T_{\mathrm{SIFS}} + T_{\mathrm{data}} + T_{\mathrm{ACK}} \\
< \; & T_{\mathrm{I}} - T_{\mathrm{CFP}} \\
< \; & T_{\mathrm{VV}} - M_{\mathrm{up}} \cdot (T_{\mathrm{POLL}} + T_{\mathrm{data}} + 3 \cdot T_{\mathrm{SIFS}} + T_{\mathrm{ACK}}) \\
& - M_{\mathrm{down}} \cdot (2 \cdot T_{\mathrm{SIFS}} + T_{\mathrm{data}} + T_{\mathrm{ACK}}) \\
& - (3 \cdot T_{\mathrm{SIFS}} + 2 \cdot T_{\mathrm{data}} + 2 \cdot T_{\mathrm{ACK}}).
\end{aligned} \tag{3.6}$$

Case 2) Controls for the CL data and the CL STAs when no VV data exist in the AP's queue and no VV STA exits on the polling list

When a VV data do not arrive in the N-th CFP Interval and there is no entry of VV STAs in the polling list, the CFP duration in the $(N+1)$-th CFP Interval is defined as the time for sending downlink CL data arrived in the N-th CFP Interval and for sending uplink CL data notified by CL STA in the polling list.

Case 3) Controls for the case where neither VV nor CL data exist in the queue, and neither VV nor CL STA exists on the polling list

When neither VV data nor CL data arrived in the N-th CFP Interval and neither VV nor CL STAs are on the polling list, the CFP duration in the $(N+1)$-th T_{I} is set to the minimum value.

3.5.1.3 Evaluation

The effect of the proposed communication quality assurance scheme using PCF was evaluated by computer simulations. Since the communication quality assurance scheme has to work effectively under high traffic load conditions, best effort traffic of 14 Mbit/s was posed to evaluate the performance of the proposed scheme under high traffic load. Firstly, throughput of the downlink CL data is examined by changing the load of the downlink CL data. In the evaluation, the requirement of the minimum bandwidth of CL data is assumed to be 5 Mbit/s. Secondly, maximum delay performance of the VV data was evaluated changing the traffic load of the VV data while assuming the same traffic load for both uplink and downlink transmissions. The requirement for maximum delay of VV data is set to be 100 ms. There are assumed 5 STAs transmitting and receiving the VV data. Table 3.2 shows the parameters used for the simulations. T_{CFP} is set as the minimum integer satisfying Eq. (3.6).

Figures 3.6 to 3.9 show the results of the computer simulations. The "Conventional-A" in the figures is the communication quality control scheme proposed in [29] which deals with multiple downlink data flows having different requirements on the communication quality. The "Conventional-B" is the communication quality control scheme

Table. 3.2 Simulation Parameters.

Transmission Rate	24 Mbit/s
Data Size	1000 Byte
ACK Size	14 Byte
MAC Header	28 Byte
PLCP Header	4 μs
Preamble	20 μs
SIFS	16 μs
PIFS	25 μs
DIFS	34 μs
Slot Time (for EE class)	9 μs
Slot Time (for BE class)	18 μs
CFP Interval	300 ms
Max. Retry Number	4
CW_{min}	15
Number of BE STAs	40

proposed in [27] for uplink video streams. Figures 3.6 and 3.7 show the throughputs of CL and BE data, respectively, with the load of CL data changed. Figure 3.6 compares the effects of the proposed and the conventional communication quality control schemes to guarantee required bandwidth. Figure 3.7 shows the impact of the proposed and the conventional bandwidth guarantee schemes on the throughput of BE data.

The Conventional-A guarantees the required minimum bandwidth for the downlink data by scheduling with a fixed duration of the CFP. From Fig. 3.6, it is verified that the bandwidth is guaranteed for CL data by the scheme of the Conventional-A. In the Conventional-B, the requirement on the maximum delay is assured for the uplink data flow, by scheduling the uplink transmissions based on the polling list. There is no specification of the scheduling method for the downlink data transmissions by the AP. In the IEEE 802.11 standard, it is specified that the data from AP to STA can

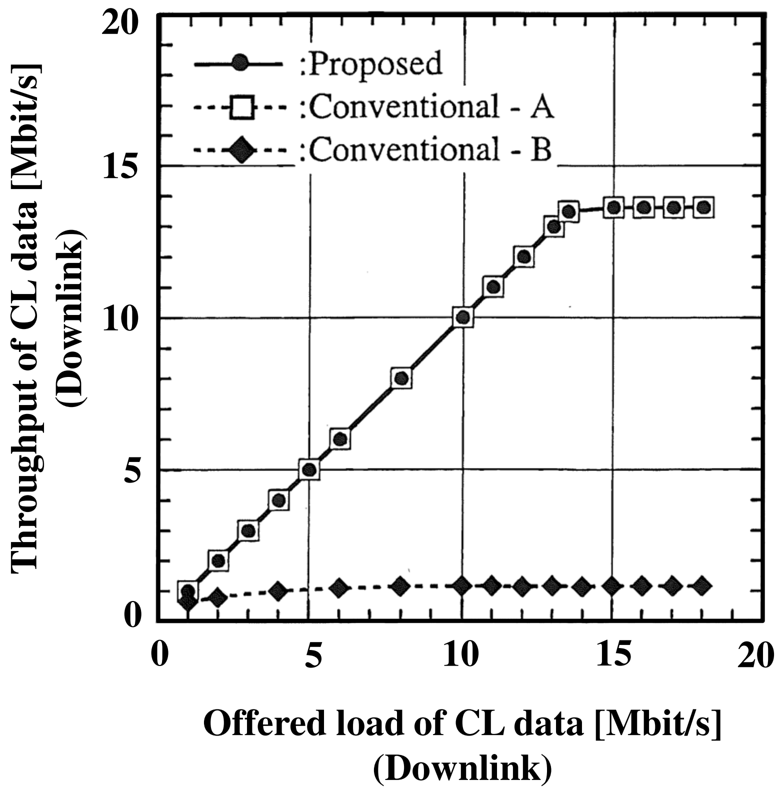

Fig. 3.6 Throughput of the CL data.

be piggybacked to the polling frame when the destination STA of the polling frame is also the destination of the downlink data. When we assume this procedure, the opportunity of a downlink data transmission depends on the scheduling for the uplink data. As a result, it becomes impossible for Conventional-B scheme to satisfy the required bandwidth of 5 Mbit/s. The proposed scheme, in contrast, can achieve almost the same performance as the conventional scheme which guarantees the minimum bandwidth for the downlink data and can outperform the Conventional-B. From Fig. 3.7, it is observed that the throughput of BE data in the proposed scheme is higher than those of conventional A and conventional B when the load of CL data is relatively low. Therefore, it is verified that the proposed scheme is the most efficient.

Figure 3.8 shows the maximum delay of the downlink VV data when the traffic

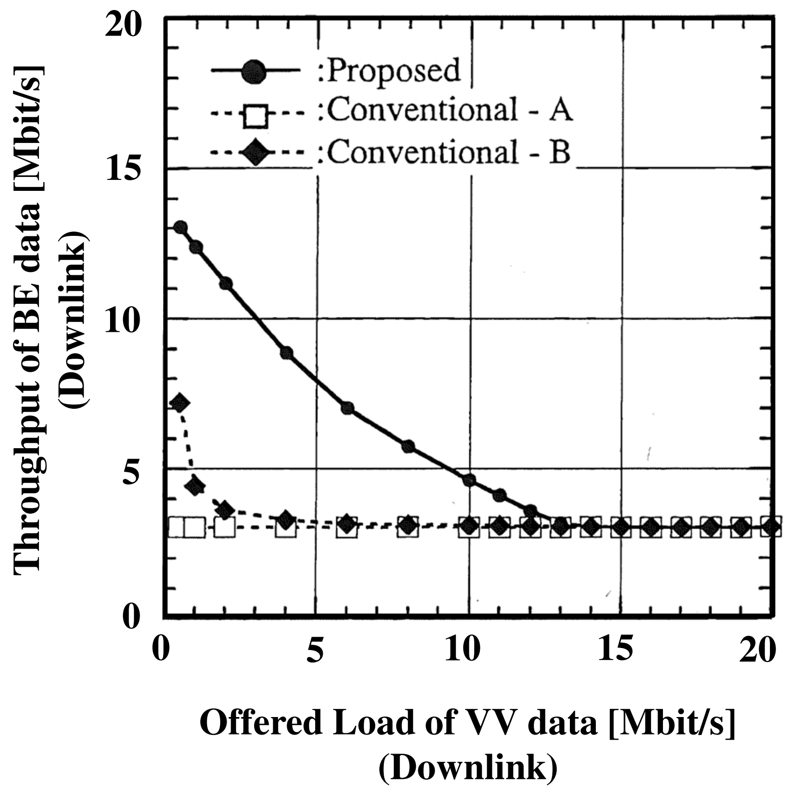

Fig. 3.7 Throughput of the BE data with the load of CL traffic changed.

load of VV data is changed while the background traffic of the BE data is fixed to 14 Mbit/s. The ratio of the uplink and the downlink loads is kept constant. It is an evaluation to compare the maximum delay between the proposed and the conventional schemes. It is seen that the maximum delay requirement of 100 ms is satisfied for the VV data until the channel is saturated. Compared to the Conventional-A where the downlink data are scheduled, the amount of the downlink data that can be transmitted by the proposed scheme, in which both uplink and downlink data are scheduled, is small. The maximum delay in the proposed scheme, however, is almost the same as in the Conventional-A scheme until the channel is saturated, with a large improvement observed compared to the Conventional-B.

Figure 3.9 shows the maximum delay for the uplink VV data with the load of VV

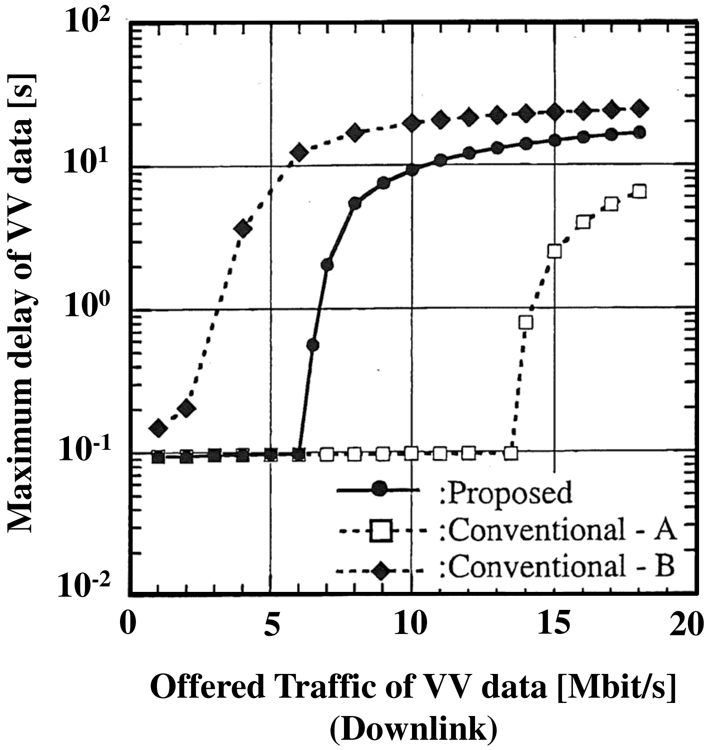

Fig. 3.8 Maximum delay for the downlink VV data.

data changed. Again, the amount of the uplink and downlink VV data are kept the same. This is an evaluation to compare the maximum delay between the proposed and the conventional schemes. Since the Conventional-A is for the transmissions of downlink data, the Conventional-B is used as the object of comparison. The saturation points of the channel differ between the Conventional-B scheme where the uplink data are scheduled and the proposed scheme in which both uplink and downlink data are scheduled. Compared to the Conventional-B, however, it is verified that the proposed scheme can offer the same performance on the maximum delay of 100 ms until the channel is saturated.

From the above results in Figs. 3.6 to 3.9, it is verified that the proposed scheme can achieve almost the same performances of maximum delay and the minimum bandwidth until the channel is saturated. Compared to the Conventional-B, a great im-

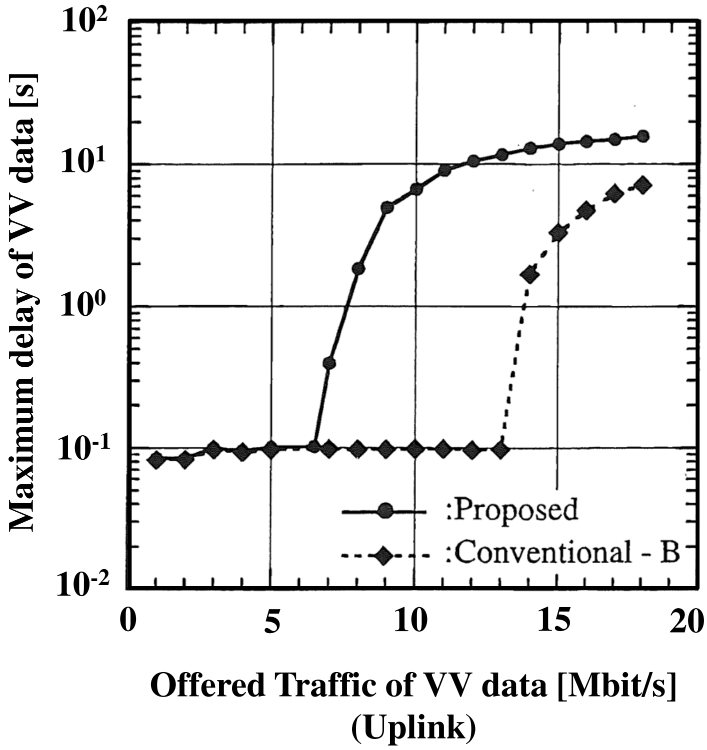

Fig. 3.9 Maximum delay for the uplink VV data.

provement is verified for the proposed scheme with regard to the maximum delay and the minimum bandwidth. It is also observed that the same quality on the maximum delay is provided for the uplink data until the channel is saturated. Consequently, it is expected that the communication quality control is achieved for both uplink and downlink data.

3.5.1.4 Discussions

The requirements on the communication quality have to be satisfied for VV and CL data on an end-to-end basis as in Fig. 3.10 where the policy management is applied to the traffic in each network domain, and the servers managing the policy of the individual domain negotiate. By combining the proposed access control scheme with the negotiation scheme at the upper layer such as DiffServ, it is expected that the

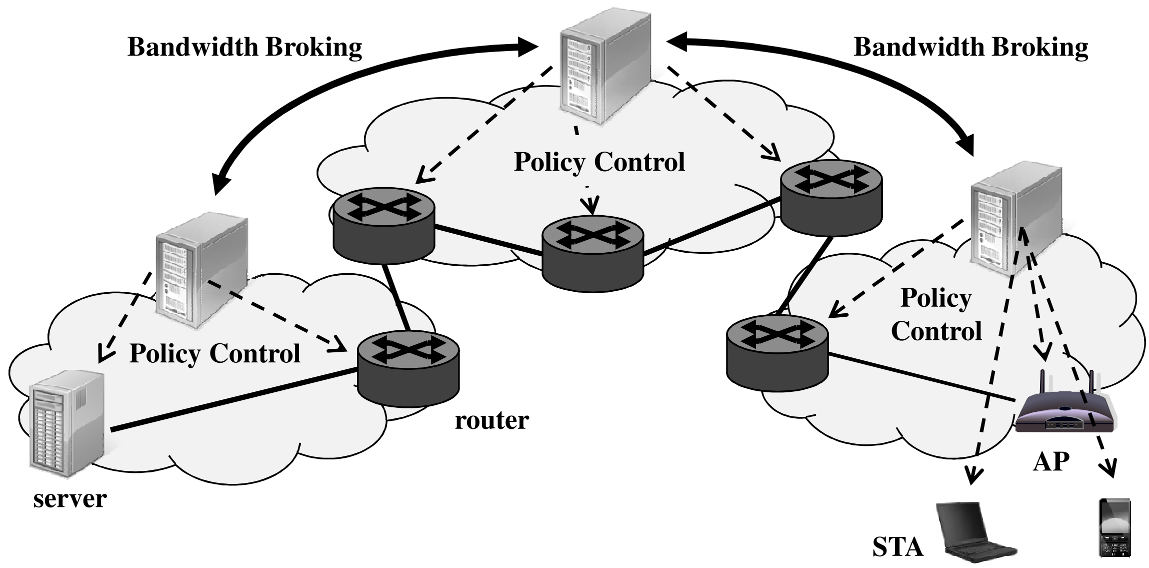

Fig. 3.10 Example of policy control including wireless LAN.

communication quality can be assured for both of the VV and the CL data.

3.5.2 Controls for the Best Effort Flows

3.5.2.1 Proposed Method

The BK, BE and the EE classes in IEEE 802.1D Annex H.2 are the service classes which do not require the assurance of the communication quality, but the order of priority is defined. It is anticipated that a kind of priority control among BK, BE and EE data is used for the purpose of service differentiation. For simplicity, we focus on the priority control between the BE and EE classes in the rest of this section. The proposed scheme is easily extended to consider the BK class as well.

In order to make the priority control using DCF, it is necessary to enhance the CSMA/CA protocol, as discussed in Section 3.4. There will be some ways to make priority control over the CSMA/CA, the method proposed here is to modify the backoff algorithm to randomize the transmission timing of the STAs. The transmission deferral time in the backoff algorithm is determined by a random variable generated at each STA and the slot time defined by the IEEE 802.11 standard. Consequently, there can be two ways of making the priority control during the backoff process. One is to divide the range of the random variable generation according to the service class of the data, and the other one is to use the different slot time values according to the service class. The priority control can be achieved by giving more transmission opportunities for the data belonging to a service class which as higher priority.

The idea of dividing the range of random variable generation has been already presented in the previous work [32]. In this scheme, however, the range of the random variable is narrowed, and a problem arises that the collision probability between the data packets belonging to the same service class will be increased with the increase of the traffic load. In the proposed method, in contrast, the priority control is achieved by varying the slot time according to the service class of the data. The order of taking out the data from queues from BE and EE queues in the AP is determined by the backoff control. In other words, the backoff algorithm is carried out independently for each of the BE and the EE data, the data of associated backoff counter reached 0 first are transmitted. Figure 3.11 shows an example of the transmission procedure.

After the transmission of the EE packet by the STA1 to the AP, AP with having data in EE and BE queues, as well as the STA2 with BE data to transmit, generate random numbers. Each data transmission is deferred for the time, which is obtained by multiplying the generated random number with the slot time. In the proposed scheme, the AP executes the access controls for service classes in parallel, therefore, two random numbers are generated for BE and EE service classes. For the case of the Fig. 3.11, the AP gets 7 for the EE data and 5 for the BE data. The STA2 gets 6 for the BE data.

The slot time is pre-determined for each service class according to the priority. In Fig. 3.11, the EE data use the slot time $T_{\rm A}$ and the BE data use the slot time $T_{\rm B}$ for the backoff procedure. By setting the slot time $T_{\rm A}$ to be smaller than the $T_{\rm B}$, the EE data have more opportunities to be transmitted compared to the BE data. Thus, for the case of Fig. 3.11, the relation between the backoff periods of the EE and the BE data of the AP and the BE data of the STA2 becomes as follows. Comparing the random numbers obtained, the number for the BE data at the AP is the smallest. By the relation of the two slot time values which is $T_{\rm A} < T_{\rm B}$, the order of the transmission

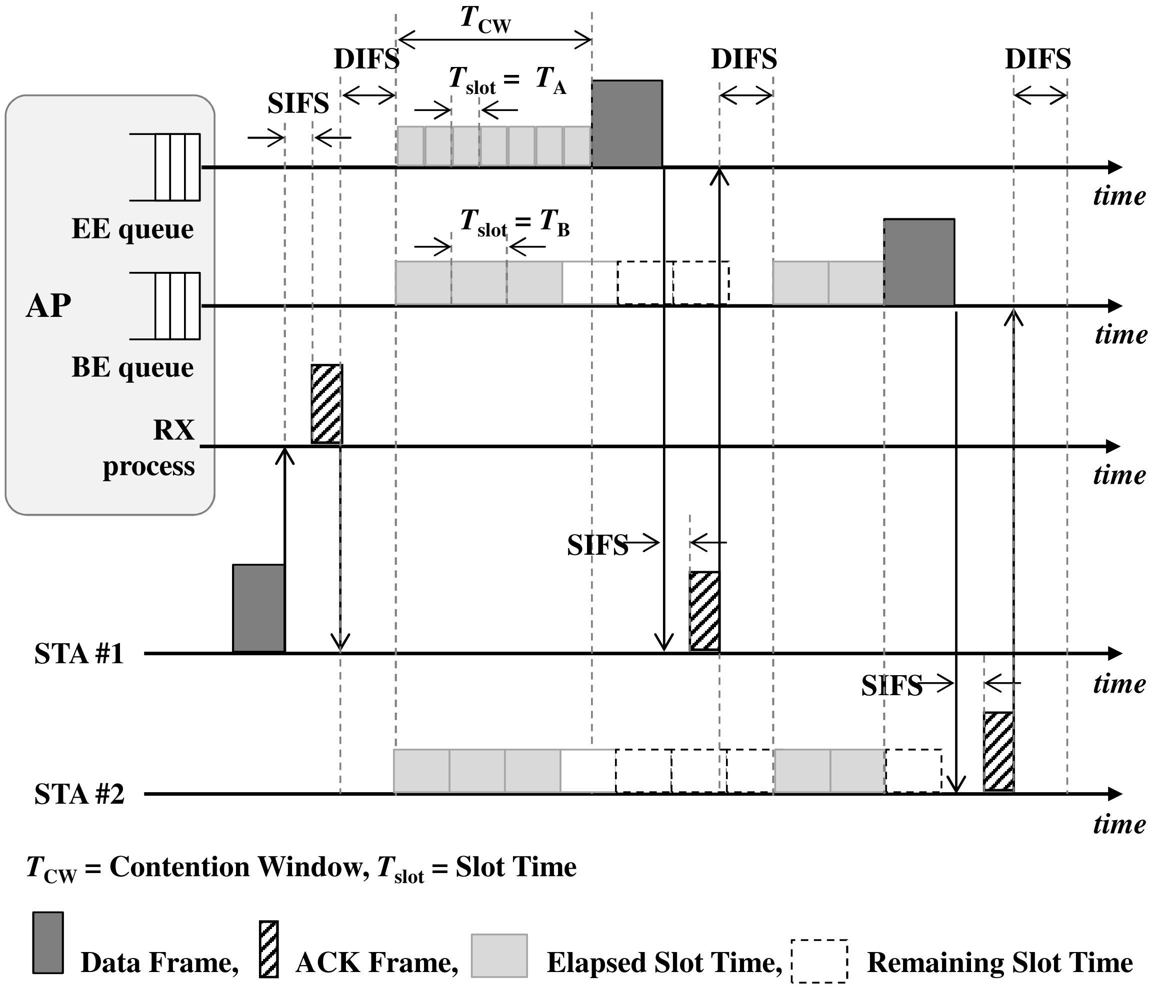

Fig. 3.11 Access control mechanism for EE and BE data.

will be,

1. the EE data at the AP,
2. the BE data at the AP, and
3. the BE data at the STA2.

By the above controls, the proposed scheme provides a prioritized transmissions for the EE over BE data.

3.5.2.2 Evaluation

The effect of the priority control for the EE and BE data is evaluated by the computer simulations. The throughput and the average delay of the EE and BE data are evaluated. Figures 3.12 and 3.13 show the results.

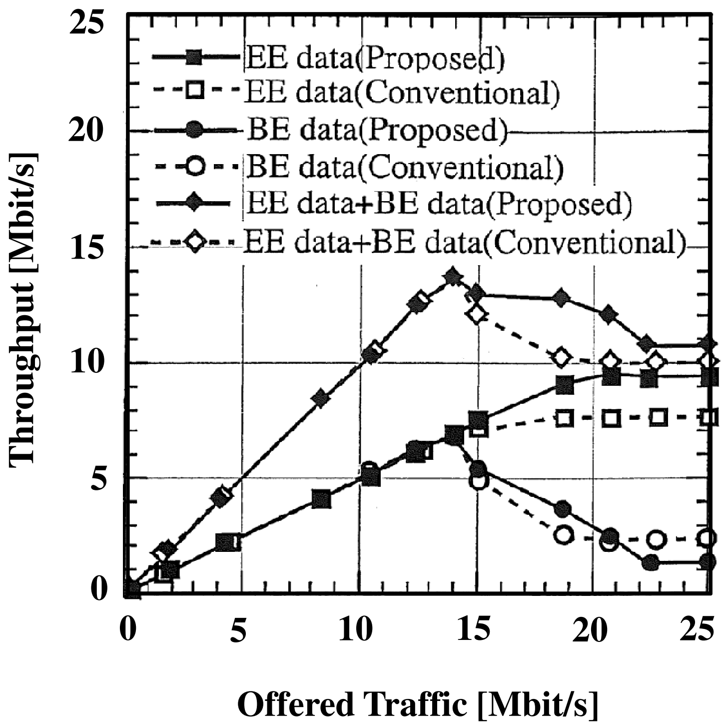

Fig. 3.12 Throughput comparison of EE and BE data.
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The numbers of STAs trying to transmit EE and BE data are set to 25, and all STAs are trying to send the same amount of data. The same simulation parameters as in Table 3.2 are used. The conventional scheme in Figs. 3.12 and 3.13 is the priority control scheme proposed in [32], where the different range of the random variable for the backoff algorithm is assigned according to the priority.

It is seen from Fig. 3.12 that the throughput curves of the EE and the BE data are almost the same for the conventional and the proposed schemes until the channel is

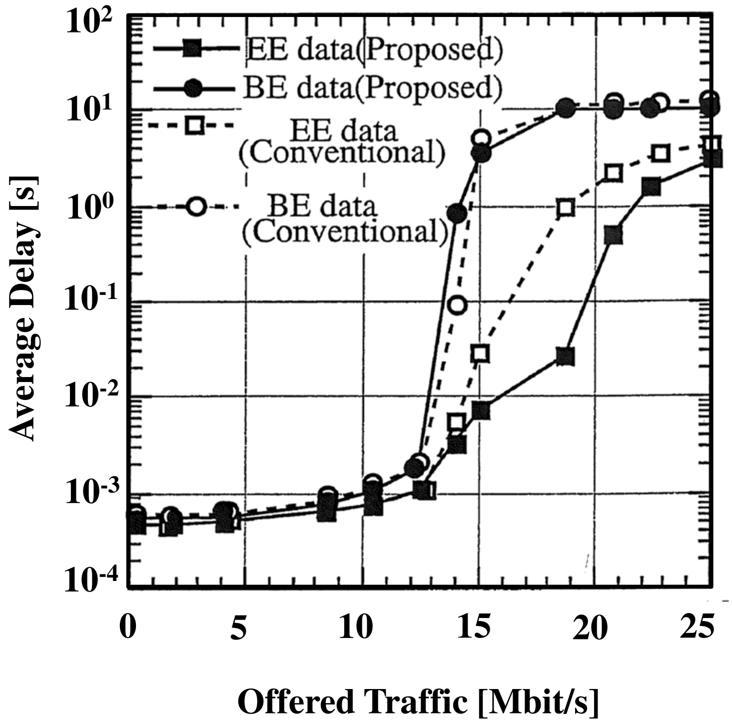

Fig. 3.13 Comparison of average delay for EE and BE data.
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saturated. After the channel is saturated, it is observed that the throughput of EE data monotonically increases with the increase of the traffic load while the throughput of BE data monotonically decreases, for both of the conventional and the proposed schemes. It is also observed that the total throughput and the throughput of EE data in the proposed scheme are higher than those in the conventional scheme, respectively.

The effect of the priority control becomes significant when the channel is highly loaded. From Fig. 3.12, it can be said that both of the proposed and conventional priority control schemes are effective. From the effect of the priority control scheme under the high channel load condition and from the point of view of the total throughput, the proposed priority control scheme can be regarded as better.

3.5.2.3 Discussions

The difference between the proposed and the conventional schemes after the saturation of the wireless channel is discussed for the following three cases:

(1) Throughput of the EE data,

(2) Throughput of the BE data, and

(3) Total throughput.

The range of the random variable distributions in those cases are set in the conventional scheme as 0 to 15 for the EE data, and 16 to 31 for the BE data. In the proposed scheme, the distribution ranges are set as 0 to 15 for both of the BE and the EE data. The slot time of the conventional scheme is set to 1 for both of the BE and the EE data. In the proposed scheme, on the other hand, the slot times for the EE and BE data are set to 1 and 2, respectively.

(1) Throughput of the EE data

In the conventional scheme, the range of the random variable generated for the backoff algorithm is divided into two regions, i.e. one for the BE data and one for the EE data. The ranges of the random variable are as shown in Fig. 3.14 (a). In the proposed scheme, the duration of the slot time for the BE data is set to be twice of the slot time duration for the EE data. Therefore, the distribution of random variable for BE data in the proposed scheme is equivalent to generating an even value at random from 0 to 31. The distributions of random variables for BE and EE data in the proposed scheme are shown in Fig. 3.14 (b).

The throughput of the EE data is affected by the contention between the EE data and the contention between the EE and the BE data. Since the range of random number generated for the EE data is the same for (a) and (b), the collision probability between the EE data is also the same for (a) and (b). Therefore, the difference of the throughput performances of the EE data between the proposed and the conventional schemes comes from the difference of the collision probabilities between the EE and the BE data. It can be explained based on the difference of the distribution of random

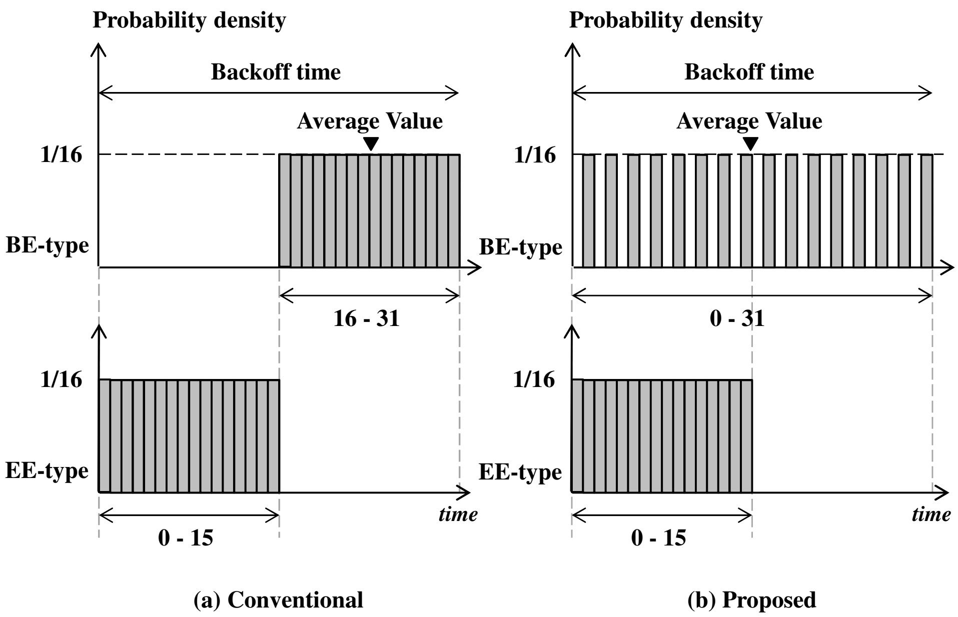

Fig. 3.14 Range of the generated random number for backoff algorithm.
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number generated for the BE data as shown in Fig. 3.14.

As shown in Fig. 3.14, the generation probability of a random number for the BE data is 1/16 for both of the conventional and the proposed schemes. In (a), however, the range for the distribution of random number for the BE data is biased to be from 16 to 31, while in (b) there is no bias. From the view point of the EE data where the generated random number is decreased by one for each time slot during the backoff procedure, the distribution of the random number for the BE data is equivalently regarded as the uniform distribution with the probability density of 1/32, as shown in Fig. 3.15 (b). Consequently, the collision probability between BE and EE data in the proposed scheme is lower than that in the conventional scheme.

(2) Throughput of the BE data

The throughput of the BE data is affected by three factors, which are the average of the random number generated for the BE data, the collision between the BE data, and the collision between the BE and the EE data. The collision between the BE and the EE data has already been discussed above, and, therefore, the other two factors are discussed next.

Figure 3.15 shows the ranges for the distribution of random numbers for the BE data in the conventional scheme (a) and in the proposed scheme (b), respectively. The average of the random number generated for the BE data in (b) is smaller than the average in (a). This implies that the BE data in the proposed scheme are transmitted with a shorter deferral time than in the conventional scheme. Consequently, the BE data in the proposed scheme have more opportunities than in the conventional scheme, to be transmitted.

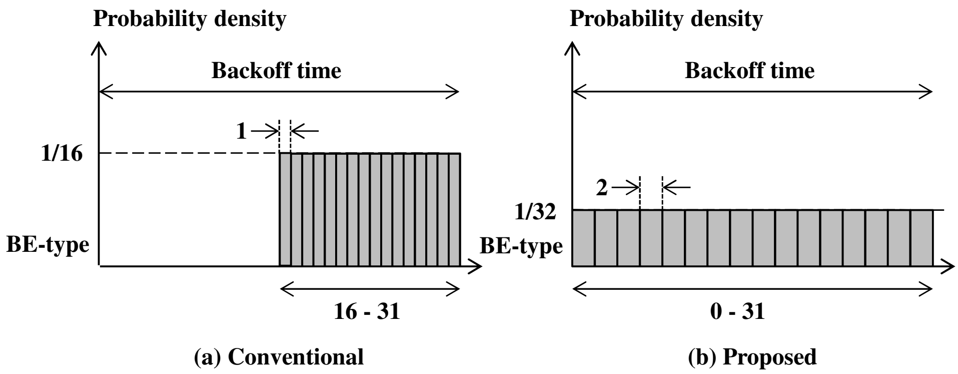

Fig. 3.15 Range of the random number for the backoff control of BE data.
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As the next step, the collision between the BE data is discussed using Fig. 3.16. Figure 3.16 shows the random number at the STA X waiting for transmission of a BE data and the random variable at the STA Y just got a new BE data to transmit, in the conventional scheme (a) and the proposed scheme (b). In (b), a collision of transmitted data arises between the STA X, where the random variable value is decreased, and the STA Y. In (a), the collision between the STA X and the STA Y can be avoided since the random number of the STA X is decreased to 15 or less and the STA Y does not generate a random number between 0 to 15. Because of this situation, the collision probability between BE data is lower in the conventional scheme than in the proposed scheme.

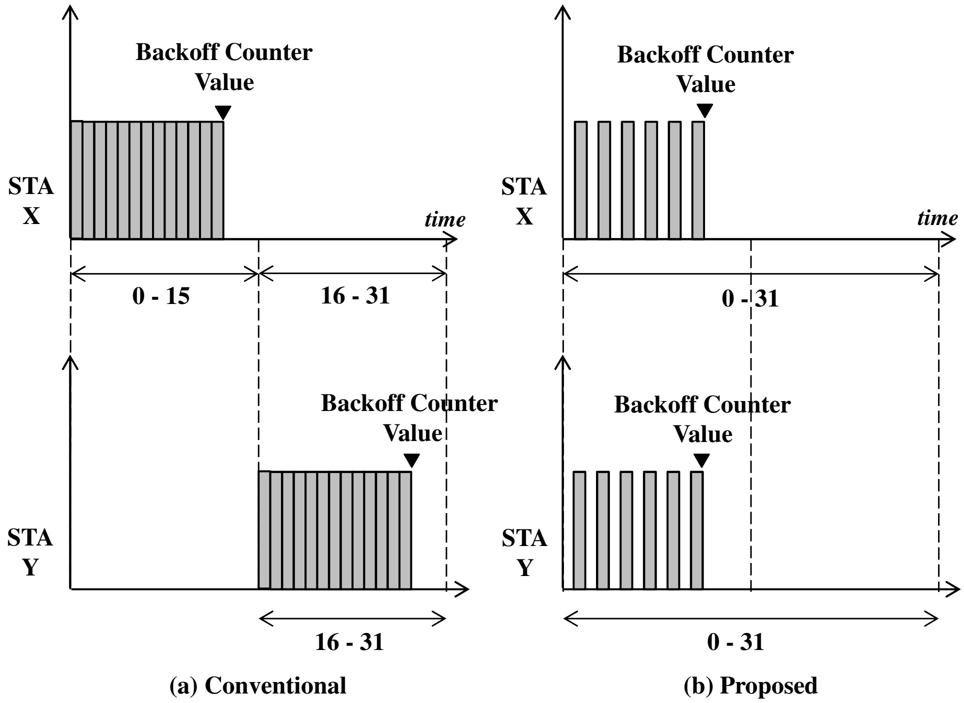

Fig. 3.16 Comparison of backoff counters for the BE data.

As for the factors affecting the throughput of the BE data, the collision between the BE and the EE data, and the difference between the data transmission opportunities seem to be dominant, compared to the contention between the BE data. Consequently, the throughput of the BE data in the proposed scheme is higher than in the conventional scheme. However, when an extremely high traffic load compared to the channel capacity is imposed, collision between the EE data was increased, and the throughput of EE data is saturated, the collision probability between BE-type data becomes dominant as the factor of affecting the throughput of BE data, which makes the throughput of BE data in the conventional scheme higher than in the proposed scheme.

(3) Total throughput

The total throughput in figure 3.12 is the sum of the throughput of the BE and the EE data. The reason why the total throughput of the proposed scheme is higher than that of the conventional scheme can be explained for by the above discussions on the throughput of the BE and the EE data.

As the first step, it is observed from Fig. 3.12 that the total throughputs of the conventional scheme and the proposed scheme decrease after the channel is saturated. This is due to the fact that the number of STAs deferring transmission rapidly increases when the channel is saturated, and the collisions between the data packets are increased rapidly. This decrease of total throughput can be suppressed by increasing the value of the CWmin, which is the upper limit of the random number generated for the backoff of the initial transmission, according to the number of associated STAs to decrease the collision probability. Based on the above results and discussions, it is concluded that the proposed scheme is a more efficient way of providing priority control than the conventional scheme. It is also seen from Fig. 3.13 that the average delay of EE data in the proposed scheme is shorter than in the conventional scheme, which also supports the conclusion that the proposed scheme is a more efficient priority control procedure.

3.5.2.4 An example of applications for the proposed scheme

When the service class of BE is defined as the default class following IEEE 802.1D Annex H.2 specification, the EE class will have a higher priority over the best effort class. In this case, the EE class can be used as the service class for the users requiring a higher priority over the general best effort service while the communication quality assurance, such as constant bandwidth or delay, is not requested. By this approach, the service items can be diversified, such that EE service is used for business, and BE service is used for home applications.

3.6 Summary

In this chapter, we considered the communication quality control schemes for the wireless LAN, and proposed such control schemes from the following two points of view:

- Communication quality assurance scheme to provide parameterized QoS/CoS which satisfies required communication parameters such as bandwidth and maximum delay in a managed environment using a contention-free access scheme, and
- Priority control scheme to provide prioritized QoS/CoS by giving more opportunities for the data belonging to a service class with higher priority to be transmitted.

For the former control scheme, the polling procedure of the PCF is utilized, and additional control schemes to adjust the duration of the CFP and to schedule the uplink and downlink transmissions according to the priority of the data are discussed. For the latter control, the backoff procedure of the CSMA/CA protocol was modified to make the priority control between the data belonging to different service classes. The effect of the two schemes is demonstrated by the computer simulations.

The communication quality assurance scheme is a kind of future-looking control scheme which can be considered an approach for making the policy-based CoS combined with other CoS mechanism such as DiffServ. The priority control scheme, on the other hand, is a more realistic approach to make service differentiation for the wireless systems using the unlicensed spectrum. The concept of making prioritized transmissions by using different deferral time during the backoff procedure was adopted in the HCF contention based access, or Enhanced Distributed Channel Access (EDCA) equivalently, of IEEE 802.11e standard.

Chapter 4

Reliable Multicast with Representative Acknowledgments

4.1 Overview

To improve the reliability and efficiency of multicast transmissions in wireless systems, a novel retransmission procedure is desired. In this chapter, the representative acknowledgment scheme for reliable wireless multicast communication is proposed that offers quiet a low packet loss rate. The proposed protocol carries out retransmissions in the datalink layer within wireless region, and retransmissions do not affect the traffic in the wired region. The representative acknowledgment scheme employs both positive acknowledgment (ACK) and negative acknowledgment (NACK) to achieve reliable multicast transmissions. To reduce the number of the responses to be returned for a multicast data frame, the proposed scheme assumes formation of receivers groups in an area , and one of the members in a group, called a representative station, returns a response for a received multicast data frame while other stations returns a NACK if and only if necessary. Reliable and efficient multicast transmissions are enabled with the proposed representative acknowledgment scheme without spending much time as in conventional reliable multicast protocols. Performance of the proposed protocol is evaluated by numerical analysis and by computer simulations. The results show that transmission time for a multicast data frame is reduced by 30% or more in a typical wireless environment.

4.2 Introduction

IP Multicast [36] is the mechanism which enables delivery of copies of the same datagram to multiple destinations at the network layer. In many applications such as audio and video conferencing, contents delivery, e-educations, collaborative computing, and others, the multicast transmission technique plays a very important role in delivering those datagram.

Multicast datagrams transmitted by using IP Multicast procedure are delivered in a best effort manner. Therefore, there is no way to compensate packet errors. Some reliable multicast transport protocols (RMTP) have been proposed which ensure the reliability in the transport layer by retransmissions [37–40]. In those studies, the source host of the multicast data needs to receive acknowledgments from all destinations, and the source retransmits a packet when a NACK is returned as the response. Because of the end-to-end acknowledgment, the RMTP offers high reliability for multicast transmissions.

Problems for the IP multicast technique with mobile hosts had been pointed out and discussed by Acharya and Baker [41–43] where they focused on the routing problem of the multicast datagram to/from the mobile hosts. These studies pertain to the problem at layer 3 and retransmissions due to the packet errors are not considered.

There are some works on the multicast protocols for wireless systems and they are divided in two categories, i.e. the positive acknowledgment (ACK) based schemes [44] and the negative acknowledgment (NACK) based schemes [45]. Generally, the ACK based protocols offer high reliability because each receiver returns an ACK frame to the sender for a successful data reception. Some problems, however, in the ACK based multicast protocols have been pointed out such as the ACK implosion problem and throughput degradations in bad channel conditions. On the other hand, the NACK based protocols offer high efficiency because the destination STAs which received the data successfully need not to return the response to the sender. In this case, however, reliability is not guaranteed. Compared with wired channels the packet error rate of the wireless channels is much higher and the impact of packet errors on the system performance becomes more serious. Therefore, an ACK based protocol is desired to offer high reliability and a NACK based protocol is also effective from the view point of channel efficiency for the wireless multicast communications.

In this chapter, a reliable multicast transmission protocol for wireless systems is discussed. To obtain reliability of multicast datagrams, the proposed protocol employs both ACK and NACK for responses and a retransmission is carried out when packet error is detected. Generally, the time for acknowledgment tends to become longer [40] as the number of destinations increases in the reliable multicast protocols. For this problem, the proposed protocol reduces the number of responses for each multicast datagram by making groups of receivers, called STA groups, which are subsets of all multicast receivers in a service area. In principle, each group needs to return only one or two responses to indicate the results of data reception in that group. Performance of the proposed protocol is evaluated by numerical analysis and computer simulations. The results are compared with that of the conventional reliable multicast protocol. The effect of grouping is discussed in terms of reliability and transfer time for each multicast datagram.

The rest of this chapter is organized as follows. In section 4.3, the system model and functions in each layer are described. Section 4.4 presents the transmission and retransmission procedure of multicast datagrams. Section 4.5 performance of the proposed protocol is presented from the results of numerical analysis and computer simulations. The performance of the proposed protocol is compared with the conventional reliable multicast protocol and the effect of grouping is discussed. Section 4.6 concludes this chapter.

4.3 System Model and Configurations

When multicast receiving hosts are mobile STAs, a receiving host may locate in a different service area or even in a wired network segment at a different time. Therefore, in a network that comprises both wired and wireless network segments connecting to a backbone network is considered.

4.3.1 Network Model

An example of a network assumed in this study is illustrated in Fig. 4.1. In this figure, a multicast source host connected to a wired network segment is sending multicast datagrams to the destination hosts (STAs) in the wired and wireless network segments.

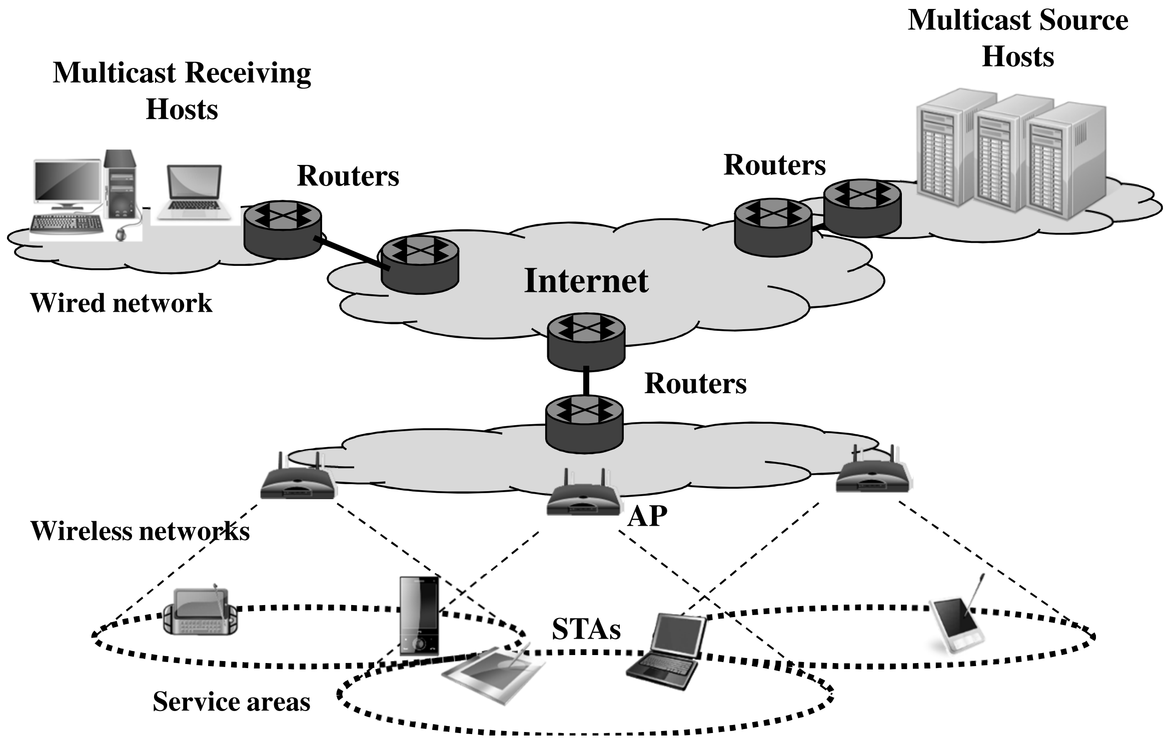

Fig. 4.1 An example of the network configurations.

The network components shown in Fig. 4.1 are summarized below with their functions.

Multicast Source Hosts The source of the multicast data that has multicast capability. As in the definition of the IP multicast protocol, the source does not need to be a member of the host group.

Routers When the multicast source host is in its network segment, the router forwards the multicast datagram to the local and to the backbone network. When a multicast datagram is coming from the backbone network, the router forwards it to the local segment if there are clients receiving it.

AP The AP is responsible for the data delivery of a wireless region. It is desired to have a multicast filtering capability in order not to forward the multicast data to the service area when no client is present.

STAs There are two kinds of STAs. One is called a representative station (RS) which returns a response for the received multicast datagram when it is polled by the AP. The other is a non-representative station (NRS) which returns only NACK frames when necessary. Details are described in the next section.

The above is just one example and there are some other cases in which an AP is co-located in a router, or the source host is a mobile host.

4.3.2 Protocol Configuration

The proposed multicast protocol is intended to compensate for the frame errors in the wireless region. Therefore, the proposed procedure should be implemented in layer 2. The protocol configuration we assumed is shown in Fig. 4.2. Although the multicast protocol is implemented in the MAC (Medium Access Control) sublayer in this figure, it can be in the Logical Link Control (LLC) sublayer.

- Layer 2 protocols

  - MAC sublayer

    In the proposed scheme, the MAC sublayer is supposed to have carrier sense and backoff function similar to the CSMA/CD of the Ethernet protocol because the proposed protocol uses both a centralized and decentralized access method. A STA must perform carrier sense to avoid a collision when it spontaneously sends a frame.
  - LLC sublayer

    The LLC sublayer only handles the datagram coming from the IP layer to the MAC sublayer. Error recovery within this sublayer is not necessary for

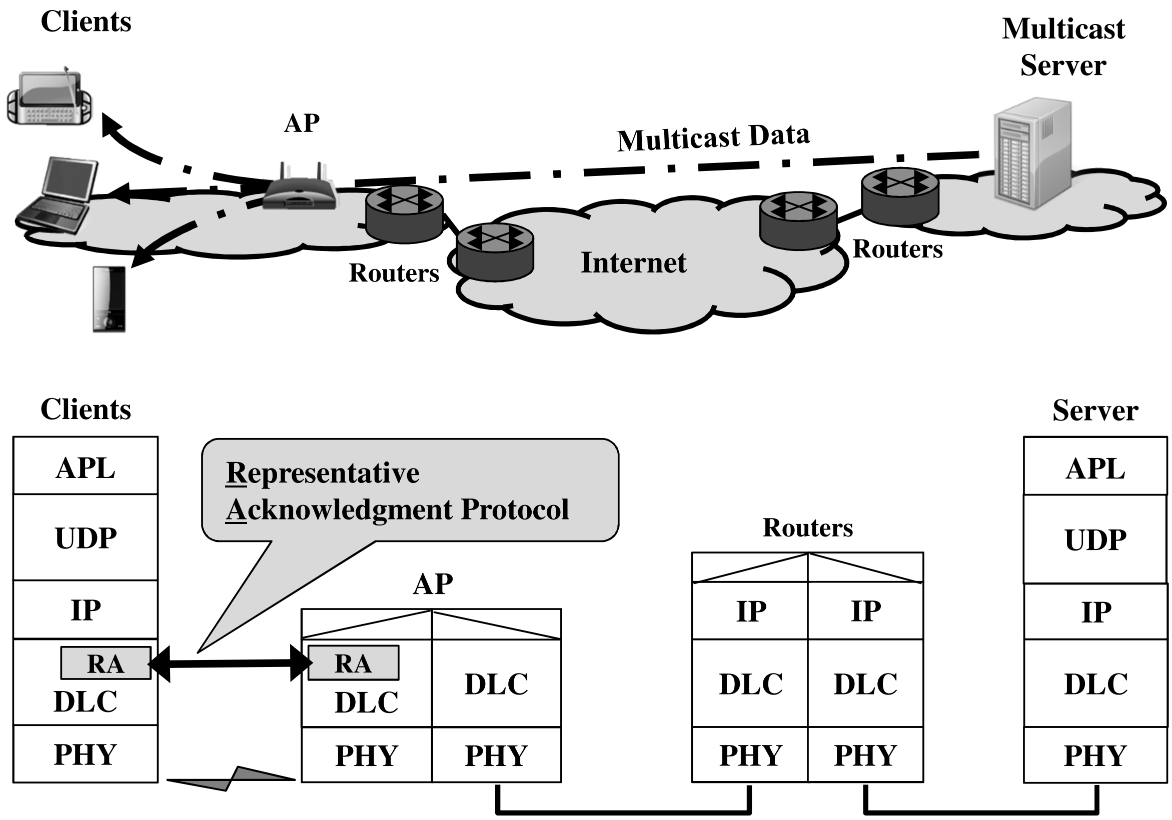

Fig. 4.2 An example of protocol configuration for multicast services.

the proposed multicast method.

- Layer 3 protocol

  The IP multicast protocol is assumed for the network layer. The routing logic of the IP is changed to deliver multicast datagrams to host group. IGMP (Internet Group Management Protocol) must be implemented within the layer 3 module to configure the host groups.

- Layer 4 protocol

  UDP will be used for the layer 4 protocol which does not need acknowledgments from the destination. A multicast protocol, such as RMTP, can be used with UDP. The multicast transmissions will then be significantly more reliable because packet errors occurring in the wired region can be recovered.

4.4 Proposed Multicast Protocol

In this section, the proposed reliable multicast protocol is described. The procedures to form STA groups and to transmit multicast data frames are presented.

4.4.1 The Grouping Procedure

In the proposed protocol, a STA must be a member of a STA group. The procedure to join or form a STA group is described.

4.4.1.1 Configuration of STA Groups

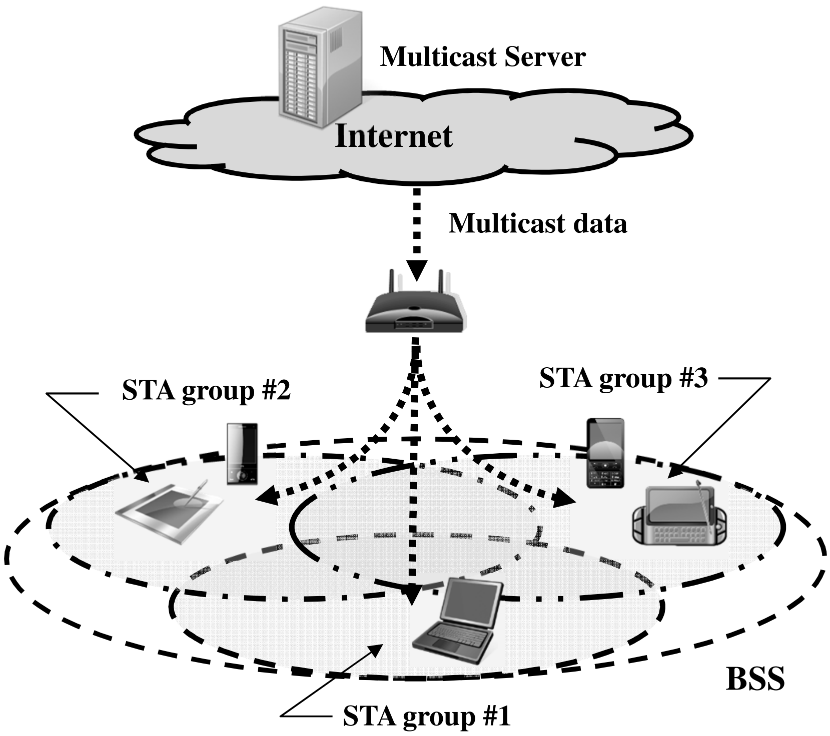

Fig. 4.3 Example of STA groups in a service area.

Figure 4.3 shows an example of three STA groups in a service area. At first, a STA which is ready to receive multicast data listens to the current multicast session and captures the response frames returned from groups. A multicast session is a sequence of frames from the initial transmission of a data frame to an ACK response returned from the final group. The STA then sends a request to the AP to become a member of a STA group which seems to be close to the STA. Information about a STA which returned a response in the previous multicast session is included in the request. The AP then regards the STA as a member of the STA group which is indicated in the request.

If there is no other STA within the transmission range of that STA, the STA sends a request to form a new STA group because that STA could find no STA groups to join. Then the STA is considered as a STA group which comprises one STA (see STA group 1 in Fig. 4.3). In the proposed protocol, responses for a multicast datagram are not returned from all STAs but from each STA group. In the example of Fig. 4.3, at least three responses are returned for a data transmission. The total number of responses is decreased as the number of STAs in a group increases.

4.4.1.2 Selection of the Representative STA

Members of a group are divided in two types.

- Representative STA (RS)
- Non-Representative Station (NRS)

The RS is selected from the members of a group. The role of the RS is to return an ACK or a NACK frame for the received multicast datagram when it is polled by the AP. The NRSs in the group return only a NACK frame if necessary.

At first, the RS is assumed to be the STA that formed the STA group. When there are some STAs in a group, the AP can determine the RS randomly, or select the STA which seems to have the worst performance.

An opportunity to change the RS can be the time when the current RS leaves the service area or the AP receives a NACK frame from an NRS or a request to join the group from a new comer. The AP knows the group of each STA.

4.4.1.3 Re-configuration of STA Groups

Considering mobility of the STAs receiving multicast data, the members of a STA group should be changed dynamically. The AP updates information on the STA groups in the BSS when it receives a control frame from a STA which tells a change of the STA group.

When a STA has moved to some other place in the same service area and the STA could find another STA group, the STA can continue to receive the multicast data by joining that group. If the STA found no STA group, it will make a new STA group. In the above cases, the STA sends a control frame to the AP to join or to make a new STA group. The AP returns an ACK frame to the STA when it received a control frame, and updates information on the STA groups. The STA, then, continues to receive multicast data as a member of the new STA group.

These frame exchanges are carried out between the multicast sessions in accordance with the access control protocol used in the system. Performance of the multicast data transmission is not affected by them because transmissions of a control frame and a following ACK frame are finished in a very short time. Moreover, the change of STA group does not occur so frequently because one multicast session finishes in a short time for a STA to change its location greatly. Therefore, reconfiguration of the STA group within a service area will not affect performance of the multicast data transmission very much.

Even if a STA has moved to another service area, the STA may continue multicast data reception if the AP of the new service area is connected to the same network. At first, in this case, a hand-over procedure is carried out between the STA and the AP. And then, the STA starts to find the STA groups in the service area as described in section. The STA will require delivery of desired multicast data if the STA could not find it. In this case, the STA may fail to receive some multicast data frames. Therefore, a retransmission control mechanism such as RMTP will be required to increase the reliability of data delivery.

4.4.2 The Data Transmission Procedure

The proposed data transmission procedure among the AP and mobile STAs is illustrated in Fig.4.4 and is summarized as follows.

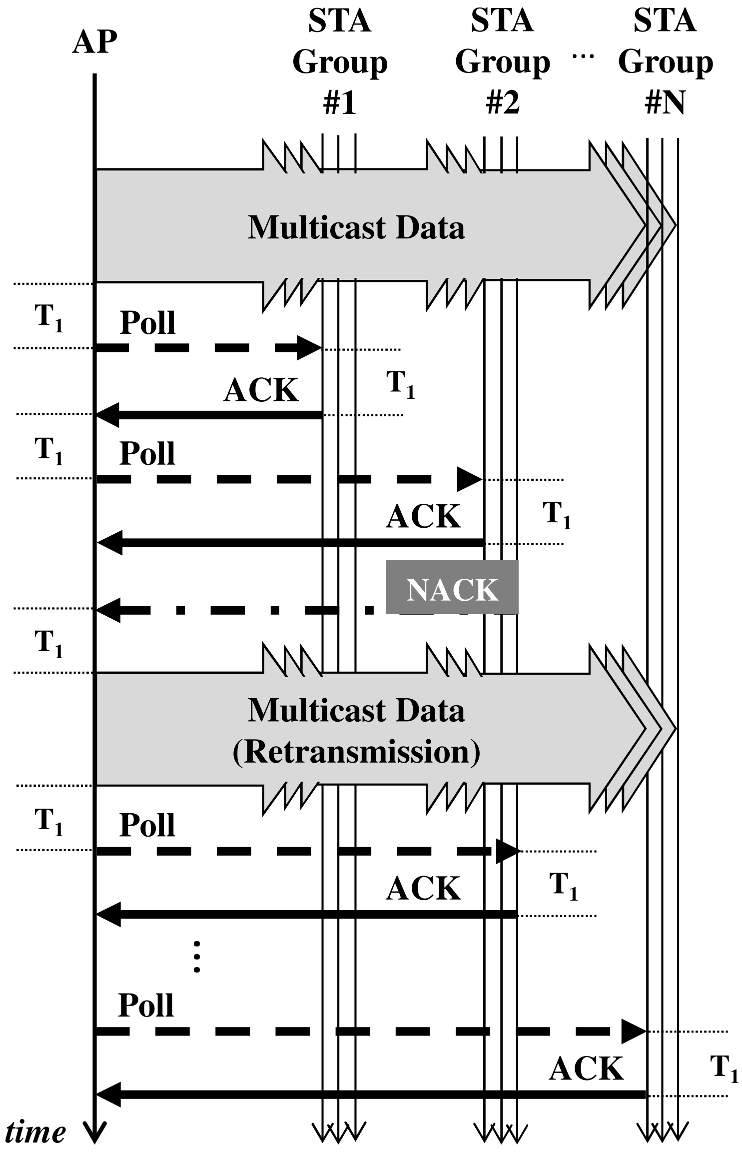

Fig. 4.4 Data transmission sequence of proposed multicast protocol.

1. The AP transmits a multicast data frame destined to all members of the mul-

ticast receiver group in the service area.

2. All STAs of the multicast receiver group in the service area receive the transmitted data frame.
3. The AP sends a polling signal to the RS of a STA group in the service area. The AP shall send a polling frame again when no repose frame is return within a certain time period.
4. The polled RS returns an ACK or a NACK frame depending on its result of reception.
5. The other STAs (NRSs) in the same STA group listen to the response that the RS returned to the AP. If an NRS failed to receive the multicast data frame correctly and if the response that the RS returned does not require a retransmission of the data frame, the NRS returns a NACK frame to the AP.
6. When the AP received a NACK frame, it retransmits the multicast data frame again. When the frame that the AP received was an ACK frame and no NACK frame followed, the AP polls the RS of another STA group (goes back to the 3) after a certain time(T1 in Fig. 4.4).
7. The AP repeats the operations of No. 3 to No. 6 until it receives an ACK frame alone from the last STA group in the service area.

The response frames returned from the RSs play very important role in this multicast protocol and they should be returned with high reliability. To protect those response frames from collisions, the polling procedure is used. The transmission timing of the NRS is randomized by the backoff procedure to avoid collisions. A STA sending a NACK frame must listen to the channel and must quit transmission if another STA sends a NACK frame. However, collisions of NACK frames cannot be eliminated perfectly because of the hidden terminal problem. A collision of NACK frames can be recognized by the senders by detecting another NACK frame which requests a retransmission of the same data after sending their NACK frames. In this case, the STA does not need to retransmit the NACK frame because the data frame will be retransmitted. When the required data frame is not retransmitted within a certain time, the STA which sent a NACK frame can judge that collision has occurred and may retransmit a NACK frame.

The number of retransmissions for data and NACK frames can be limited to a certain number to decrease the total transmission time. However, reliability of data transmissions depends on the limit of retransmissions. The effect of limiting the retransmission is discussed in the next section.

4.4.3 Priority of Frame Transmissions

To avoid collision of the transmitted frames, it is useful to give priorities to frame transmission. One idea is to use Inter-Frame Spaces (IFSs) of different lengths as in the IEEE 802.11 wireless LAN system. In the proposed protocol, two kinds of IFSs are used. We call them IFS_1 and IFS_2 ($\mathrm{IFS}_1 < \mathrm{IFS}_2$). As shown in Fig. 4.5, a polling frame is transmitted after the IFS_1 from the end of a data frame, and an ACK or a NACK frame is returned also after the IFS_1 from the end of a polling frame. The NACK frame from an NRS is returned after the IFS_2 plus random delay time of the backoff procedure and retransmission of the data frame is carried out after the IFS_1 from the end of the NACK frame. This prevents NACK transmissions of the NRSs that failed to receive the NACK frame from the RS.

Another idea is to use the same IFS assuming that collisions will occur only between the NACK frames. In this case, the AP retransmits the multicast data if signal is detected on the channel within a defined period after receiving an ACK frame. This procedure will be effective when no other BSS is detected on the same channel.

4.5 Performance Evaluations

In this section, performance of the proposed protocol is evaluated. At first, we discuss reliability of frame transmission in terms of the limit on data transmission. Next, the results of numerical analyses and computer simulations are presented.

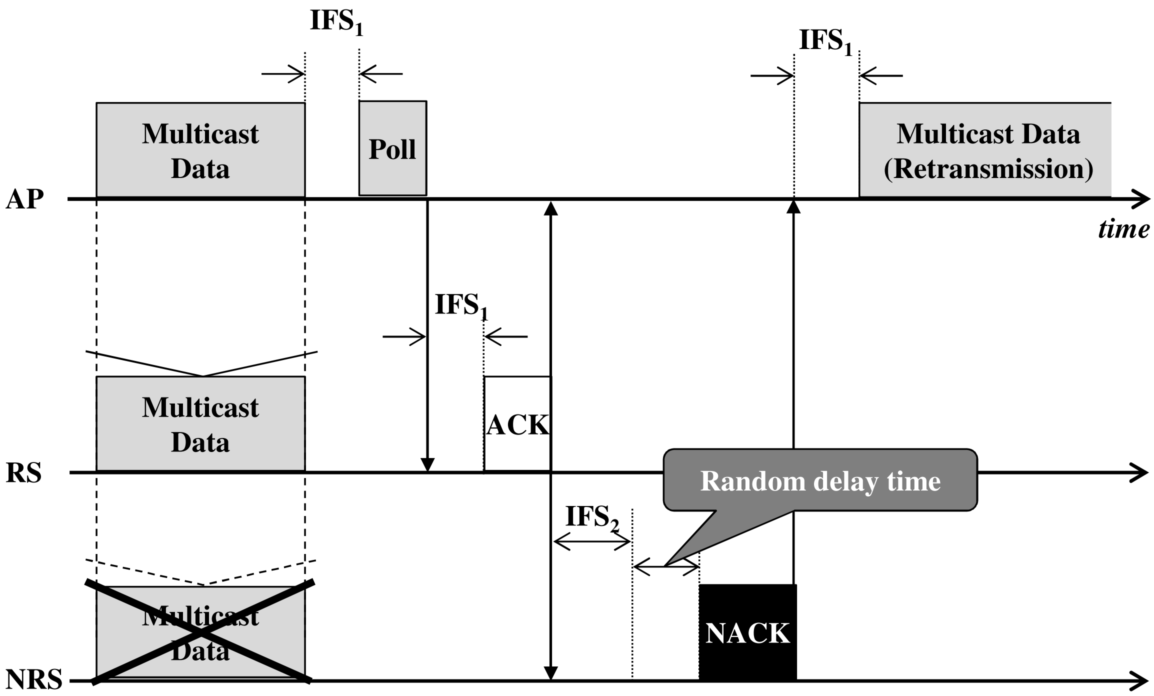

Fig. 4.5 Frame exchange sequence with IFS.

4.5.1 The Effect of Limiting Retransmissions

Reliability of the data transmission has a relationship with the number of retransmissions for a frame. If the infinite retransmissions are assumed, the data transmission will be sufficiently reliable. However, the time to complete a frame transmission will be very long. The trade-off between reliability and the number of transmissions must be taken into account to evaluate the performance of the proposed protocol.

To simplify the analysis, AWGN channel is assumed. When a bit error rate (BER) of the channel ε is given, the frame error rate (FER) ε_{f} is obtained as;

$$\varepsilon_{\mathrm{f}} = 1 - (1 - \varepsilon)^{L_{\mathrm{d}}}, \tag{4.1}$$

where L_{d} is the length of the multicast data frame and $(1-\varepsilon)^{L_{\mathrm{d}}}$ is the frame success rate for one STA. We denote the frame success rate as s_{f}.

When a data frame is transmitted to N STAs, the probability that all N STAs receive the multicast data frame in the initial transmission, $P_N(1)$, is as follow;

$$P_N(1) = (1 - \varepsilon_{\mathrm{f}})^N = {s_{\mathrm{f}}}^N. \tag{4.2}$$

When the frame errors are assumed to occur randomly, the rate of STAs at which frame error occurred is ε_{f} . Therefore, the number of the destination STAs in the second transmission (first retransmission) is $N\varepsilon_{\mathrm{f}}$. Then the probability that all STAs received the multicast data frame successfully at the second frame transmission, $P_N(2)$, is obtained as follow;

$$P_N(2) = {s_{\mathrm{f}}}^{N\varepsilon_{\mathrm{f}}} \cdot (1 - {s_{\mathrm{f}}}^N). \tag{4.3}$$

The probability that the frame transmission completes at the kth transmission $P_{\mathrm{s}}(k)$ can be expressed as follows;

$$P_N(k) = \begin{cases} s_{\mathrm{f}}^N & \text{for} \quad k = 1, \\ \\ {s_{\mathrm{f}}}^{N\varepsilon_{\mathrm{f}}^{k-1}} \displaystyle\prod_{i=1}^{k-1} (1 - {s_{\mathrm{f}}}^{N\varepsilon_{\mathrm{f}}^{i-1}}) & \text{for} \quad k \geq 2. \end{cases} \tag{4.4}$$

In this study, reliability of multicast data is defined as the probability that all STAs receive the data correctly. The probability is increased as the number of transmissions increases and is calculated from the Eq. (4.4). Reliability, $R_N(k)$, is sum of Eq. (4.4) in terms of k. If a certain reliability is given, the necessary number of transmissions is calculated.

The relationship between the number of transmissions for a data frame, k, and the probability that cannot achieve certain level of reliability, $1 - R_N(k)$, is shown in Fig. 4.6 and Fig. 4.7. In these figures, the size of the multicast data is assumed to be 1500 Byte which is the maximum length of the Ethernet frames. 20 STAs and fixed BER of 1.0×10^{-4} are assumed in Fig. 4.6 and Fig. 4.7, respectively.

The necessary number of transmissions for certain N and ε_{f} pairs can be found from these figures. Figure 4.6 shows that nine transmissions of a data frame are required when BER is 5×10^{-5} and 16 transmissions for BER of 10^{-4} to obtain the reliability of 99.99% for 20 STAs. From the results of the above figures, 21 transmissions for a

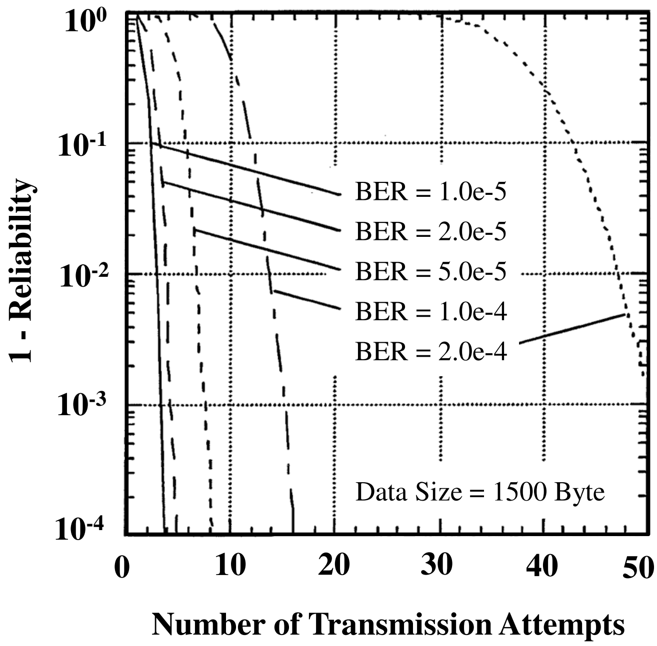

Fig. 4.6 Reliability of 1500 Byte frames for 20 STAs.

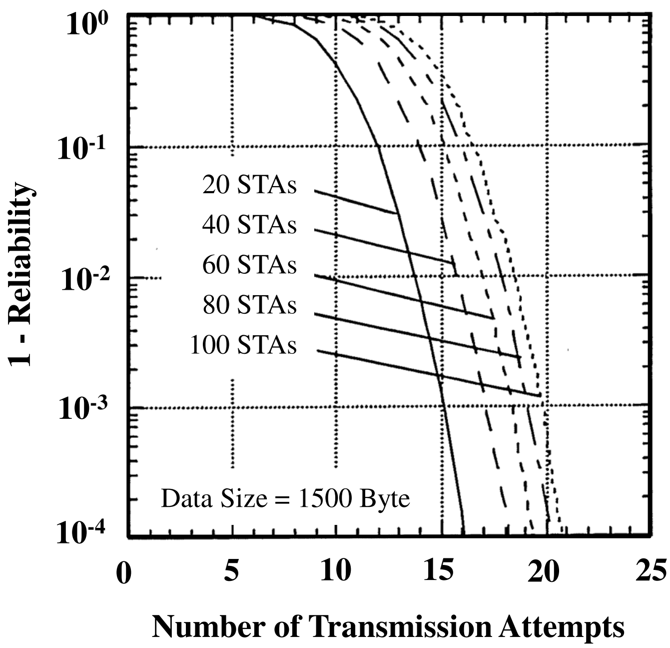

Fig. 4.7 Reliability of 1500 Byte frames for BER of 10^{-4}.

data frame will provide more than 99.99% of reliability for the 1500 Byte data frames when we consider BER of up to 1.0×10^{-4}.

Figures 4.8 and 4.9 show the reliability of a multicast data when the frame size is 64 Byte. This frame size corresponds to the minimum length of the Ethernet frames. From Fig. 4.8, BER of 2.0×10^{-3} is close to be the limit to provide reliable and effective multicast services. On the other hand, Fig. 4.9 shows that 30 data transmissions will provide the reliability of 99.99% even when the number of STAs in the service area is 100.

From the above results, from 20 to 30 transmissions can be considered for a data frame to offer reliable and effective multicast data delivery. Therefore, we assumed the limit of transmission attempts to be 30 for the performance evaluation of the proposed protocol.

4.5.2 Performance of The Proposed Protocol

The performance of the proposed protocol is evaluated by numerical analysis and computer simulations.

4.5.2.1 Numerical Analysis

Performance of the proposed protocol is numerically analyzed. The time to complete a transmission of the multicast data frame at the AP, which is the time from the initial transmission of a multicast data frame to the reception of an ACK frame from the last STA group without NACK frames following the ACK frame is derived. When the AP receives no response frame within a certain time after sending a polling frame to a STA group, the polling frame is retransmitted to the same STA group. A retransmission of the data frame is carried out when the AP receives a NACK frame.

First of all, we defined time units for some events and then calculated the probabilities for each event and the transmission time for a multicast data frame. The transmission time for a multicast data frame is obtained from the time for a events and corresponding probability.

The time units and variables we defined for the calculation are listed below.

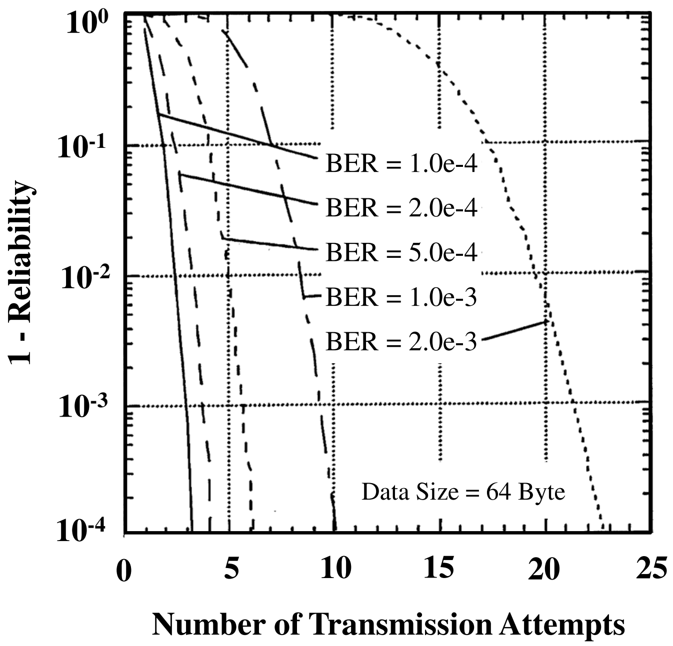

Fig. 4.8 Reliability of 64 Byte frames for 20 STAs.

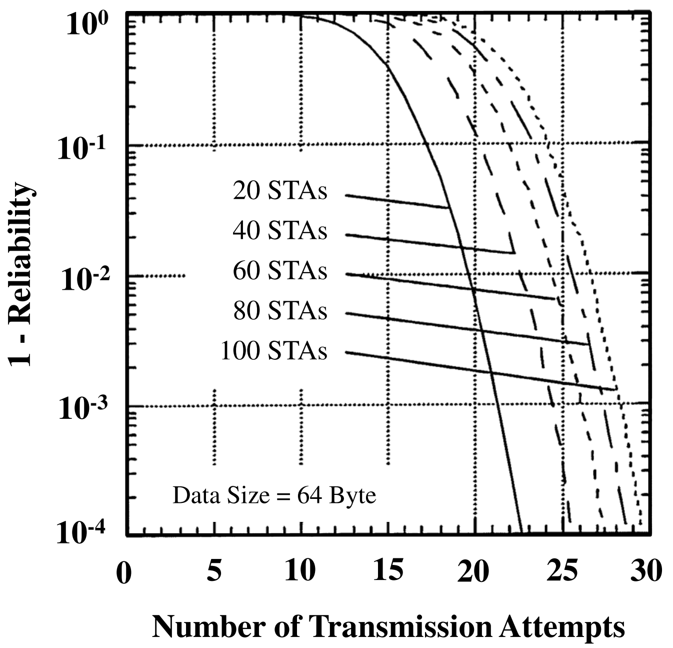

Fig. 4.9 Reliability of 64 Byte frames for BER of 2.0×10^{-3}.

T_0	:	$T_\mathrm{d} + \mathrm{IFS}_1$
T_1	:	$T_\mathrm{c} + \mathrm{IFS}_1 + T_\mathrm{c} + T_\mathrm{out2}$
T_2	:	$T_\mathrm{c} + T_\mathrm{out1}$
T_3	:	$T_\mathrm{c} + \mathrm{IFS}_1 + T_\mathrm{c}$
T_4	:	$T_\mathrm{c} + \mathrm{IFS}_1 + T_\mathrm{c} + \mathrm{IFS}_2 + \overline{D} + T_\mathrm{c}$
N	:	the number of destination STAs
M	:	the number of the STA groups in a service area
L_d	:	the size of a data frame
L_c	:	the size of control frames
T_d	:	time to send a data frame
T_c	:	time to send a control frame
T_out1	:	timeout value for waiting a response frame for a polling frame
T_out2	:	timeout value for waiting a NACK frame following an ACK frame
$\overline{D}$	:	average delay to return an NACK

In the above lists, the control frames are the polling, ACK and NACK frames.

When BER of channel ε is given, FER of control frames ε_c is expressed just like Eq. (4.1).

$$\varepsilon_\mathrm{c} = 1 - (1-\varepsilon)^{L_c} . \tag{4.5}$$

When a data frame is sent successfully, the AP receives the ACK frame from a RS with probability $(1-\varepsilon_\mathrm{c})^2$. This is for the reason that two control frames, i.e., both polling and ACK frames, must be transmitted successfully. We denote this probability α.

$$\alpha = (1-\varepsilon_\mathrm{c})^2 . \tag{4.6}$$

If either one of those control frames failed, the AP sends another polling frame and waits for the response. The interval to retransmit the polling frame after a previous transmission is T_out1, and this event occurs with probability $(1-\alpha)$. When the data are successfully transmitted, and the time that the AP is acknowledged and ready to send the polling frame to the next STA group, T_{ack}, is expressed as follow;

$$T_{\mathrm{ack}} = \sum_{i=1}^{\infty}(1-\alpha)^{i-1}\alpha\left\{(i-1)T_2 + T_1\right\}. \tag{4.7}$$

Just as in the above method, the probability that the AP receiving the NACK frame is derived. The time until the AP becomes ready to retransmit the multicast data frame after a previous transmission, T_{nack} is obtained as follow;

$$T_{\mathrm{nack}} = \sum_{i=1}^{\infty}(1-\alpha)^{i-1}\alpha\left\{(i-1)T_2 + \frac{M}{N}T_3 + \frac{(N-M)}{N}T_4(1-\varepsilon_{\mathrm{c}})\right\}. \tag{4.8}$$

The NACK frame can be returned by the RS when it is polled by the AP and by (at least) one NRS in the STA group when the RS returned an ACK frame for the polling frame from the AP. If the frame errors are assumed to occur randomly, the probability of NACK frames to be returned from RS is proportional to the number of STAs in the service area when BER of the channel is low. Therefore, Eq. (4.8) is an approximation which is effective while BER is low.

FER of data frame ε_{f} is derived as in Eq. (4.1) and the probability that the frame transmissions of a multicast data are repeated up to k times is shown in Eq. (4.4). Then, the time to complete a multicast data transmission is calculated with these equations and the time necessary for each operation. The time for the k th transmission $T(k)$ is obtained as follow;

$$T(k) = P_N(k)\{k\cdot T_0 + (k-1)T_{\mathrm{nack}} + M\cdot T_{\mathrm{ack}}\}. \tag{4.9}$$

Therefore the time to complete a data transmission, T_{all}, is obtained as follow;

$$T_{\mathrm{all}} = \sum_{k=1}^{\infty} T(k). \tag{4.10}$$

The results are shown in Fig. 4.10 and Fig. 4.11 together with the simulation results.

4.5.2.2 Simulation Results

Performance of the proposed protocol is evaluated by numerical analyses and by computer simulations. The conditions are listed in the table 4.1.

Table. 4.1 Simulation Conditions.

Conditions	Values
Channel model	AWGN
Bit rate (wireless region)	20 Mbit/s
IFS1	20 μs
IFS2	60 μs
Header + FCS length	34 Byte
Data field length	1500 and 64 Byte.
Retransmission limit	30
Number of STAs	20
Number of STA group	2, 5, 10 and 20

Figures 4.10 and 4.11 show the performance of the proposed protocol for the data size of 1500 Byte and 64 Byte, respectively. The horizontal axis is BER and the vertical axis is the transfer time for a multicast data frame. The cases of 2, 5, 10 and 20 STA groups are considered. The results of simulations and analyses show good agreement and this means that our analysis is valid.

The time to complete a transmission of multicast data frame depends on the number of STA groups in the service area because the time for acknowledgments becomes longer as the number of STA group increases. However, the probability that the NACK frames collide on the channel will be increased when there are only a few number of STA groups.

The number of STA groups will be a maximum of 6 in the ideal condition just like six triangles in a hexagon. Therefore, the case of 5 STA groups seems to be close to the actual cases.

4.5.3 Comparison with Other Protocols

The performance of the proposed protocol is compared with the conventional reliable multicast protocol where the AP receives ACK or NACK frames from all destinations

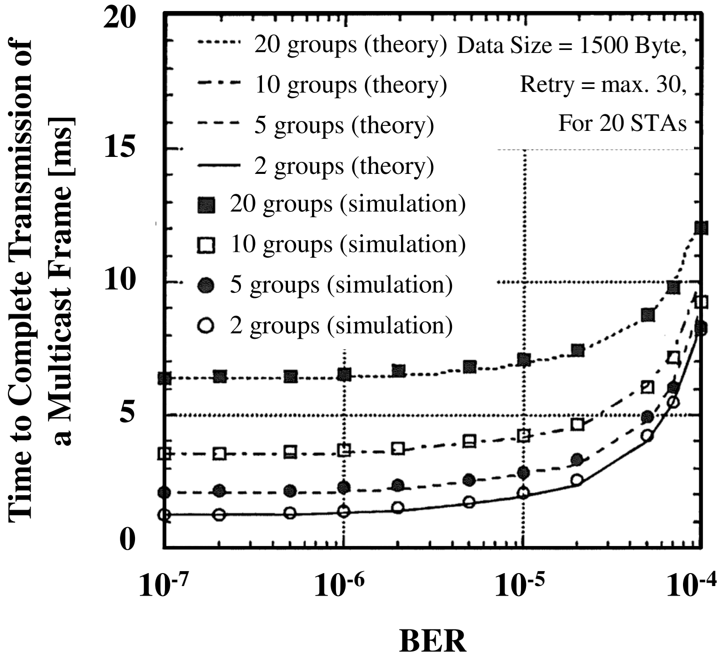

Fig. 4.10 Performance of the proposed protocol for $L_{\mathrm{d}} = 1500$ Byte.

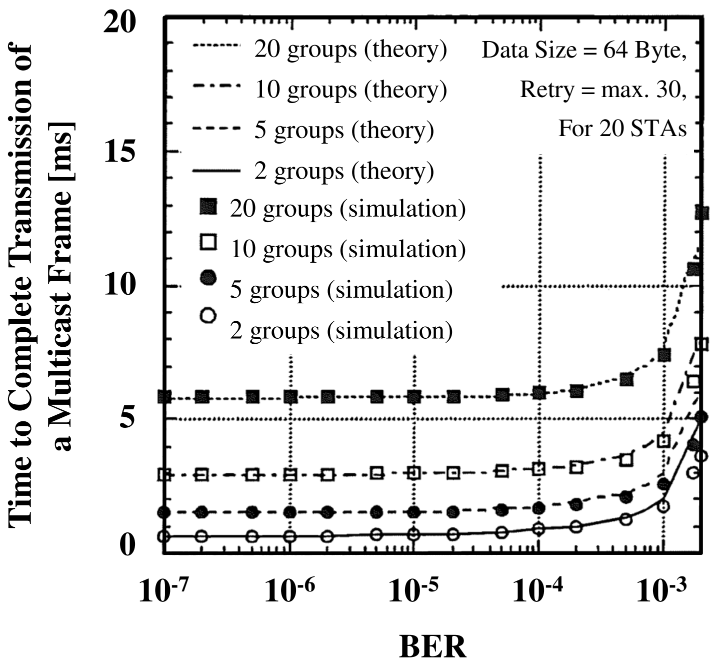

Fig. 4.11 Performance of the proposed protocol for $L_{\mathrm{d}} = 64$ Byte.

for a data transmission.

In conventional method 1, the reverse channel (from a STA to the AP) is assumed to be a random access channel and the STAs must perform the backoff procedure to avoid collision of the responses. In conventional method 2, the reverse channel is assumed to be perfectly scheduled. In the above methods, a polling procedure is carried out when a response frame is not delivered correctly.

Figure 4.12 shows the results of analyses for when the data frame size is 1500 Byte and 20 STAs are in the service area. The effect of the grouping appears when there are 10 STA groups, i.e., two STAs comprise a STA group on the average when BER is 2×10^{-5}.

In the low BER region, conventional methods can exhibit better performance than the proposed one. This is because the proposed method needs two transmissions, i.e. a polling and a response, for a group to be acknowledged while the conventional needs only one transmission for a STA if the channel error is not considered. Basically, the case of 10 STA groups of the proposed protocol needs the same number of control and response frames for a STA. The difference in the above cases in the low BER region is mainly for timeout value of the proposed scheme. The timeout value must be determined considering the time range that the NACK frames to be returned to the AP.

In the multicast communications, the channel condition for a STA differs from each other and the performance of the protocol is dominated by the STAs which have relatively poor channel quality. As an example of the channel quality, packet error rate (PER) of 10% is assumed for system evaluation in IEEE 802.11 Wireless LAN standardization body. This PER corresponds to the BER of about 10^{-5} and 10^{-4} when the data size is 1500 Byte and 64 Byte, respectively. Therefore, the performance at the high BER region has important meaning. The proposed protocol has better performance in a typical BER region of the wireless environment.

Figure 4.13 shows the results when the data frame length is 64 Byte. Basically, Fig. 4.13 shows the same tendency as Fig. 4.12. In this case, however, the timeout value affects the performance more seriously because the frame transmission time is shorter than in the previous analyses. The system designer should determine the timeout

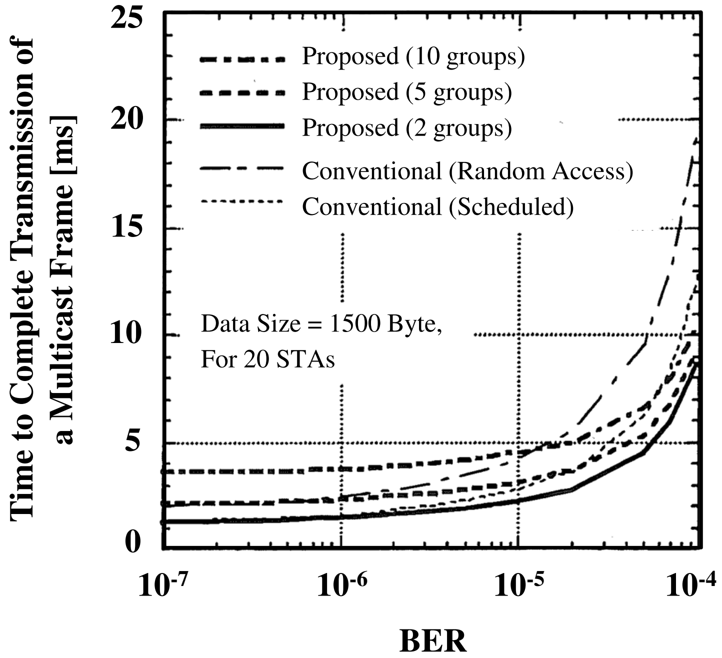

Fig. 4.12 Performance comparison of multicast protocols for $L_{\mathrm{d}} = 1500$ Byte. ©2001 IEICE

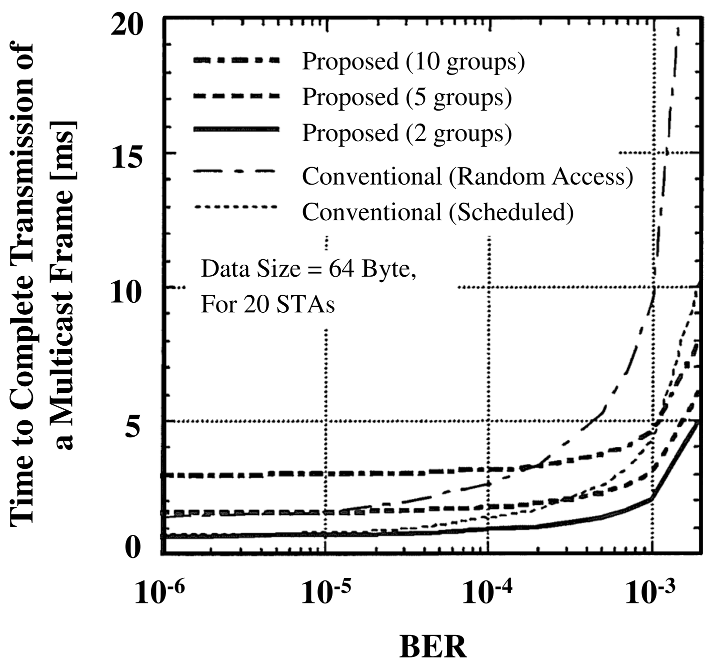

Fig. 4.13 Performance comparison of multicast protocols for $L_{\mathrm{d}} = 64$ Byte. ©2001 IEICE

value carefully considering the number of STAs in each STA group.

Compared with the conventional method 1, the proposed protocol can reduced the transmission time of a multicast data frame about 48% for the case of 2 STA groups and about 28% for the case of 5 STA groups at BER of 10^{-5} when $L_{\mathrm{d}} = 1500$ Byte. When $L_{\mathrm{d}} = 64$ Byte., the improvement at BER of 10^{-4} is about 67% and 35% for the cases of 2 STA groups and 5 STA groups, respectively. The above results mean that the services using the multicast technique can be provided at a much lower cost with the proposed protocol.

4.6 Summary

In this chapter, a reliable multicast protocol for wireless systems is proposed which reduces the transmission time for a multicast frame using the grouping procedure and representative acknowledgment scheme.

The performance of the proposed protocol is evaluated in a random channel error environment and compared with the conventional reliable multicast protocol. In the performance evaluations, it was found that the number of retransmissions can be limited within a certain number which can offer sufficient reliability. Thirty transmissions for a data frame can provide the reliability of 99.99% for a typical wireless environment. The effect of the proposed protocol appears when a STA group comprises 4 or more STAs on an average. For example, the transmission time for a multicast data frame can be reduced by 48% and 28% for the cases of 2 and 5 STA groups, respectively, when the frame length is 1500 Byte and BER equals 10^{-5}. The services using the proposed multicast protocol can be provided at a significantly lower cost than is now possible.

Although the procedures to form STA groups in the service area and to select a representative STA are briefly introduced in this chapter, there will be some methods to accomplish them.

The proposed protocol can be used with other reliable multicast protocols, such as RMTP, which ensure the reliability of the multicast data at the transport layer. In this case, perfect reliability may not be required for the layer 2 protocol and other problems, e.g., optimization of the timeout value and the number of retransmissions must be solved.

Chapter 5

MAC Protocol for Inter-Vehicle Communication Networks

5.1 Overview

In order to improve the safety and efficiency of the road traffic, the short range communications such as vehicle-to-vehicle communication and vehicle-to-road side infrastructure communication will play an important role. The IEEE 802.11p standard defines the "Wireless Access in the Vehicular Environment (WAVE)" based on the OFDM PHY of the IEEE 802.11a/j wireless LANs. Thinking about the communication and ranging capabilities, however, spread spectrum communications can be an alternative for such communication systems. Considering the mobility of the terminals, the vehicle-to-vehicle communication network need to be modeled as a dynamic network where terminals come in and go out from the communication range of one terminal. Therefore, its operation and management have to be done in a decentralized manner. In this chapter, a MAC protocol for the inter-vehicle communication network using spread spectrum technique is proposed. In such a network, a vehicle is assumed to exchange various data regularly with other vehicles around it. Computer simulations have been carried out to evaluate the basic performance of the proposed protocol in a high way scenario. It is shown that inter-vehicle communications can be smoothly carried out among the vehicles in such a environment.

5.2 Introduction

Recently, demands for the driving support systems and autonomous car driving systems are increasing and many car manufacturer implements sensors and communication devices in the cars. In such systems, communications among vehicles in proximity and communications between a vehicle and a road side infrastructure play an important role to exchange or to collect various information around the vehicle. There have been many researches for short range vehicle-to-vehicle (V2V) and vehicle-to-infrastructure (V2I) communications systems and networks [51–54]. By providing information of other vehicles such as speed, acceleration, steering, the driver or the vehicle itself can make right decision to avoid traffic accidents.

There had been a standardization activity in the IEEE 802.11 WG to use the wireless LAN system as a basis for the V2V and V2I communication systems, and the IEEE 802.11p standard was published in 2010. The IEEE 802.11p standard defines the OFDM PHY that the IEEE 802.11a and 802.11j have developed and network architecture similar to the IBSS, i.e. each vehicle communicates with the other vehicle directly in a peer-to-peer manner based on the CSMA/CA protocol.

In order to recognize the changes of situation, the differential information of speed, acceleration and/or distance will be important. For this purpose, periodic information exchange will be one of the requirements. For the PHY technology, the spread spectrum (SS) communication is known to have anti-jamming, anti-interference and multiple access capabilities [55]. Especially for the direct sequence spread spectrum (DSSS) communications, ranging capability is also expected during the communication. Thus it is advantageous to apply the DSSS techniques to the vehicle-to-vehicle and vehicle-to-infrastructure communications.

In order to use the spread spectrum communication technique, assignment of the spreading code will be significant for code division multiple access (CDMA) capability. The "SS Boomerang Communications" technique has been proposed [56–59] which enables communications with other terminals without knowing the spreading code assigned to them. However, its networking and management methods have not yet been discussed.

In this chapter, a MAC protocol for the V2V and V2I communication networks using the "SS Boomerang Communication" technique is proposed. It is a modified method based on Reservation-ALOHA protocol, and its slot reservation and channel access schemes are discussed. The basic performance of the protocol is evaluated by computer simulations under the highway situation. From the result, it is shown that inter-vehicle communication can be smoothly carried out between one and the surrounding vehicles.

Rest of this chapter is organized as follows. In section 5.3, the network model and basic mechanism of the "SS Boomerang Communication" are presented. In section 5.4, MAC protocol for the inter-vehicle communication network is explained. In section 5.5, performance of the proposed protocol is evaluated by computer simulations. Section 5.6 summarizes this work.

5.3 System Model Description

5.3.1 Network Model

Figure 5.1 shows an example of the inter-vehicle network. For simplicity, we assumed a highway scenario in which moving direction of vehicles is fixed and the distribution of the vehicles obeys Poisson point process.

From the above assumption, the probability density function of k vehicles existing within a certain region of x can be expressed by following formula.

$$f_{k,x}(k,x) = \frac{(\lambda x)^k}{k!}\exp(-\lambda x). \tag{5.1}$$

Then the distribution of the headway between two successive vehicles becomes exponential distribution and the probability density function is expressed as following equation.

$$f_{h_s}(r) = \lambda \exp(-\lambda r), \tag{5.2}$$

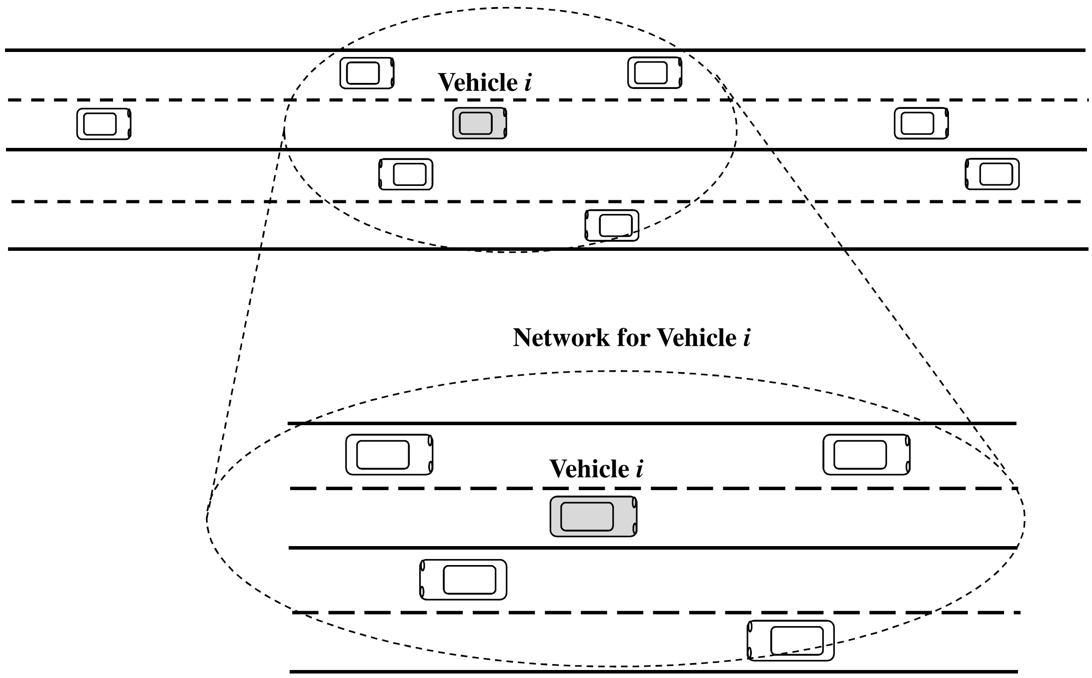

Fig. 5.1 An example of Inter-Vehicle Communication Network.
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where $1/\lambda$ is average head spacing of the vehicles.

From the results of statistical investigations, it is also known that the speed of a vehicle follows the normal distribution in such an environment that the vehicle can travel freely like a highway. And it is also known that, in such a situation, vehicles organize groups.

We assumed that a network is organized by the vehicles in proximity that should mutually pay attention to. In general, the region that a vehicle needs to take care differs from vehicle to vehicle. Thus, it is appropriate to assume that every vehicle has its own network within a certain area.

5.3.2 Communication Mechanism

In the V2V and V2I communication network, a vehicle communicates with other vehicles in proximity in a point-to-multipoint manner. The membership of the network may change from time to time and a network for a specific vehicle continues to move. In order to improve the safety and efficiency of the road traffic adapting to the environment that continues to change, a vehicle needs to communicate with other vehicles frequently and periodically to exchange information. The protocol for the V2V and V2I communications needs to be flexible against the mobility of the vehicles and the network. In such a situation, the network management has to be done in a decentralized manner (i.e. base-station-less). For the frequent and periodic data exchanges, a TDMA like protocol is desirable. However, in a decentralized network, there is no base-station which assigns the slot for data transmission to the vehicles. The Reservation-ALOHA (R-ALOHA) is suitable for such a network.

The other points in designing the V2V and V2I communication protocol is the use of DSSS communication technique and assignment of spreading code. Before describing the MAC protocol, SS Boomerang Communication technique is reviewed.

5.3.3 SS Boomerang Communication

Figure 5.2 shows the principle of SS Boomerang Communication technique. The SS Boomerang Communication is a method that enables a vehicle to communicate with other vehicles without knowing the spreading code of other vehicles.

In this figure, vehicle A provides its data to vehicle B. The vehicle A and vehicle B use PN_A and PN_B, respectively. When vehicle B needs the data of vehicle A, vehicle B sends its spreading sequence of period L (L is the length of the packet defined by the system) to vehicle A. We call this transmission "Request Transmission" and the transmitted sequence "Request Packet."

Vehicle A receives the Request Packet and it multiplies the packet by its data. After that, vehicle A sends back the packet to vehicle B. We call this transmission "Reply Transmission" and the packet "Reply Packet."

Vehicle B receives the Reply Packet. The Reply Packet is modulated by the spreading sequence of vehicle B, vehicle B can demodulate it.

By assigning different PN sequence to each vehicle, the Request Transmission can

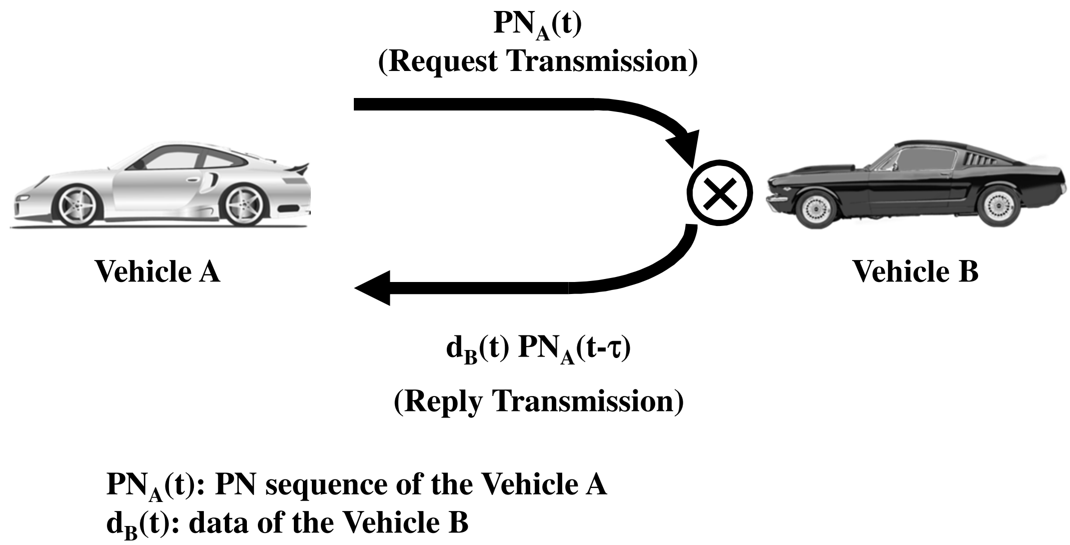

Fig. 5.2 Principle of Spread Spectrum Boomerang Communication.
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be multiplexed in the code domain. And the vehicle A can provide its information to multiple vehicles in a point-to-multipoint manner.

5.4 Proposed MAC Protocol

In this section, the MAC protocol proposed for the V2V and V2I communication networks is described. The protocol proposed here is modified version of the Reservation-ALOHA protocol to accommodate the SS Boomerang Communication technique.. The channel configuration and slot reservation algorithm, data exchange algorithm are described below.

5.4.1 Channel Configuration

We assume that the channel is divided into frames which contains N time slots as shown in Fig. 5.3. A slot is comprised of the data part and acknowledgement part. The data part has double length of a packet plus processing delay including the propagation time and the acknowledgment part has ACK and NACK slots as the feedback for the previous transmission.

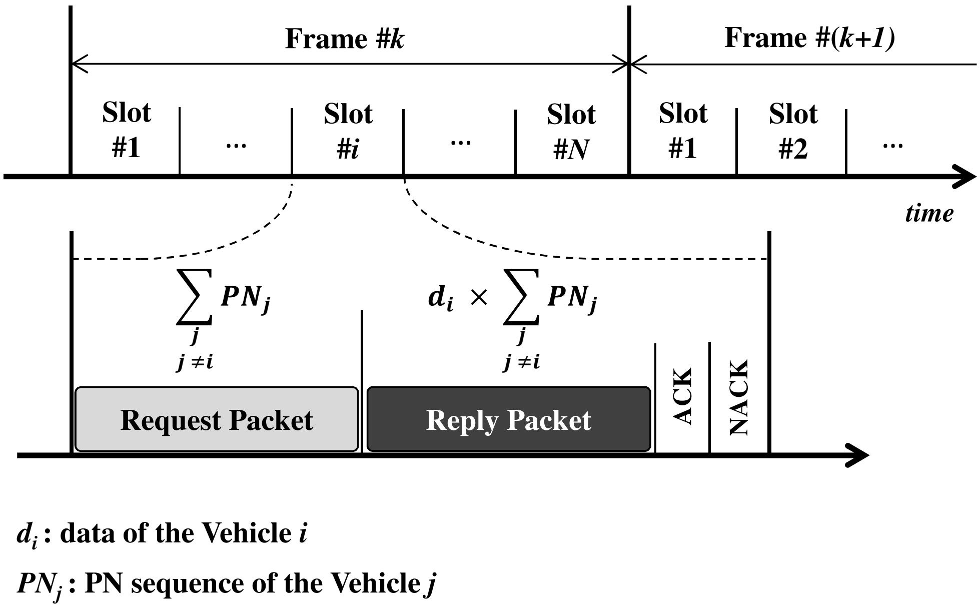

Fig. 5.3 Channel Structure.

Each vehicle reserves one slot from the empty slots in a frame and is either in the “Request Mode” or the “Reply Mode”. A vehicle is in the Request Mode in a slot other than the one it has reserved and transmits its PN sequence for L periods which corresponds to the length of a data to retrieve information of another vehicle. This transmission is called “Request Transmission” and this PN sequence is called “Request Packet”. In the slot that a vehicle has reserved, the vehicle waits for the Request Packets from the other vehicles. After receiving the PN sequences, the vehicle in the Reply Mode multiplies the Request Packets and its data, which is called “Reply Packet,” and then sends it back to the other vehicles.

A vehicle in the Request Mode demodulates and decodes the received Reply Packet and sends an acknowledgment in the ACK slot if the Reply Packet was decoded successfully. Otherwise, the vehicle sends returns NACK by sending a signal in the NACK slot. By analyzing the acknowledgment part of the reserved slot, a vehicle in the Reply Mode is able to make a decision of whether it should change the slot to provide its information.

5.4.2 Slot Reservation Algorithm

Before starting the data exchange, a vehicle must reserve one from the non-used slots in the frame to provide its information. A vehicle starts slot reservation when it joined to a network or when it found the reserved slot overbooked.

The slot reservation is conducted be the following steps.

1. Transmit a Request Packet in each slot of the frame. If a Reply Packet is received, the slot is marked as used. If not, the slot is marked as empty. This step is only required for the vehicles that do not have the status of the slots.
2. Choose one slot from the empty slots randomly.
3. Go into the Reply Mode in the reserved slot and send the Reply Packet.
4. Analyze the acknowledgment part of the reserved slot. If the transmission is successful, the reservation process is completed. If NACK is detected, the slot is marked as used, and go back to Step 2.

After completion of the slot reservation process, a vehicle needs to track the result of Reply Transmission and decides to change its slot if NACK signal is detected successively.

5.4.3 Data Transmissions and Receptions

After completing the slot reservation, a vehicle starts data exchange with vehicles in proximity. Fig. 5.4 shows the data exchange algorithm by describing the behavior of the vehicles in each slot.

A vehicle goes into the "Request Mode" in the slots other than the reserved slot and goes into "Reply Mode" in the reserved slot. A vehicle in the "Request Mode" starts the "Request Transmission," i.e., a transmission of its PN sequence at the beginning of the slot. The PN sequence corresponding to the data length of L is transmitted

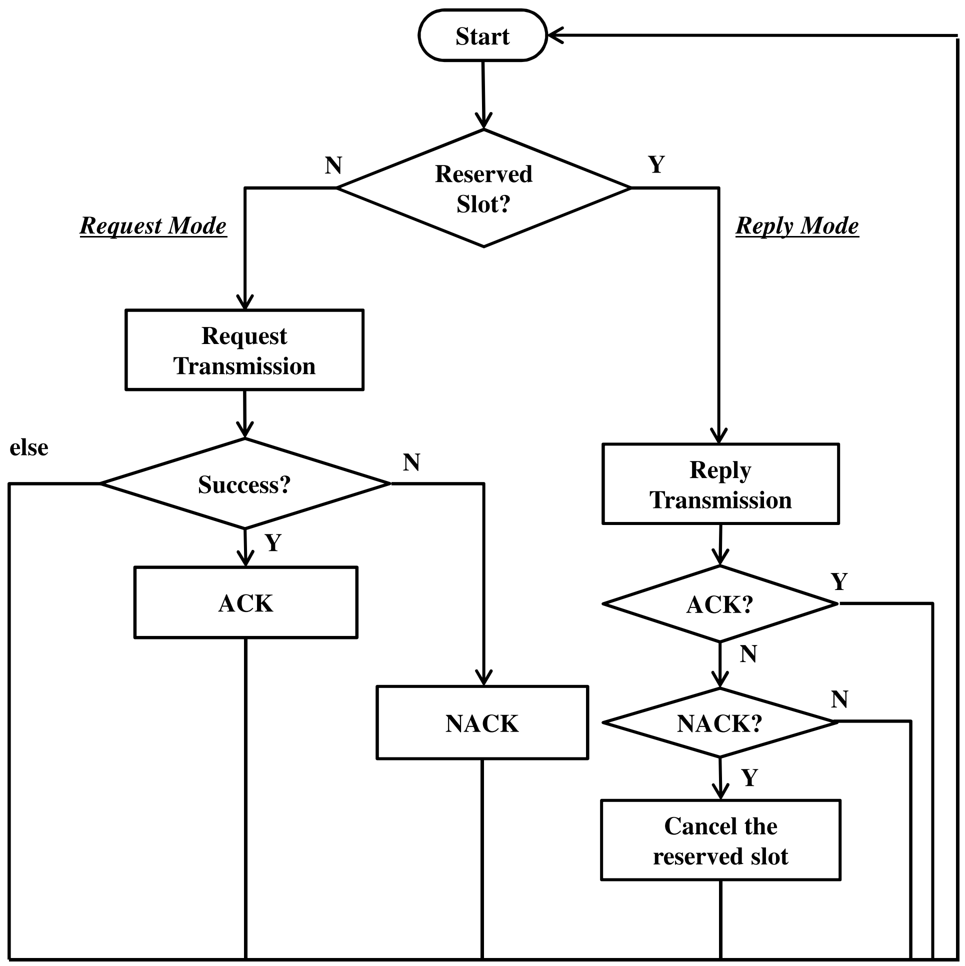

Fig. 5.4 Data exchange algorithm.

without data modulation. After sending the Request Packet, the vehicle waits for the Reply Packet.

The PN sequence of vehicles other than the one that reserved the slot is multiplexed and received by the vehicle which reserved the slot. The Request Packets received at the vehicle i, denoted $Rq_i(t)$ is expressed as follow.

$$Rq_i(t) = \sum_{\substack{j \\ j \neq i}} PN_j\left(t - \tau_j\right), \tag{5.3}$$

where τ_j is the propagation delay between vehicle j and vehicle i.

Aftre receiving the Request Packet, the vehicle i starts transmission of the Reply Packet which is denoted $Rp_i(t)$;

$$Rp_i(t) = d_i(t) \cdot \sum_{\substack{j \\ j \neq i}} PN_j\left(t - \tau_j - \tau_0\right), \tag{5.4}$$

where τ_0 is the processing time. When the transmission of the Reply Packet is completed, the owner of the slot observes the acknowledgement part of the slot.

The vehicles in the Request Mode receive the "Reply Packet" and demodulate and decode the packet to retrieve the information.

$$Rp_i(t) \cdot PN_j(t - \tau_0 - 2\tau_j) = d_i(t - \tau_j) + \sum_{\substack{k \\ k \neq i,j}} C_k, \tag{5.5}$$

where C_k is cross-correlation of the spreading codes.

The vehicle demodulate the "Reply Packet" correctly sends a tone signal to ACK slot, and if failed NACK slot.

If tone signal is detected in the ACK slot, the slot reservation is continued for the next frame, i.e., the vehicle goes "Reply Mode" in the same slot of the next frame. However, when the tone signal is detected in the NACK slot, or no signal was detected in the acknowledge slots, the vehicle cancel the current reservation and starts slot reservation process again. The flowchart of the data exchange algorithm is shown in Fig. 5.4.

5.5 Evaluation of the Proposed MAC Protocol

The performance of the proposed MAC protocol was evaluated by computer simulations. Assuming that the distribution of the vehicles running on the highway obeys the Poisson distribution, computer simulations are carried out by changing the density of the vehicles. If the average head spacing is smaller, the road is crowded.

5.5.1 Simulation Conditions

The simulation conditions are shown in table 5.1. We assumed that synchronization to the slot and frame is done ideally and considered only the interference from other networks.

Table. 5.1 Simulation Conditions for Evaluation of the Proposed MAC Protocol.

Modulation Method	DSSS BPSK
Spreading Sequence	Gold Code
Spreading Code Length	127 [chips]
Propagation Path Loss	$\propto r^{-2}$
Transmission Range	160 [m]
Distribution of Vehicles	Poisson distribution
Traffic Generation	Poisson Process

Simulations were conducted changing the average head spacing $\overline{hs}$ of the vehicles to be 80, 40, 20, and 10 m. Transmit power of each vehicle is assumed to be a fixed value and the path loss exponent is assumed to be 2.0.

5.5.2 Convergence of the Reservation Algorithm

Figure 5.5 shows the convergence rate of the slot reservation algorithm with different values of average head spacing $\bar{hs}$. The x-axis is the number of the slots in one frame and the y-axis is the rate of the trials that the slot reservation was completed within 10 frames.

Figure 5.6 shows the average number of the frame to complete the slot reservation. From the result of this simulation, the number of frames needed for slot reservation was less than two frames. It is reported that frequent information exchanges, e.g. several times per second, is desired for the V2V communications and the slot reservation algorithm needs to converge quickly. The proposed slot reservation algorithm

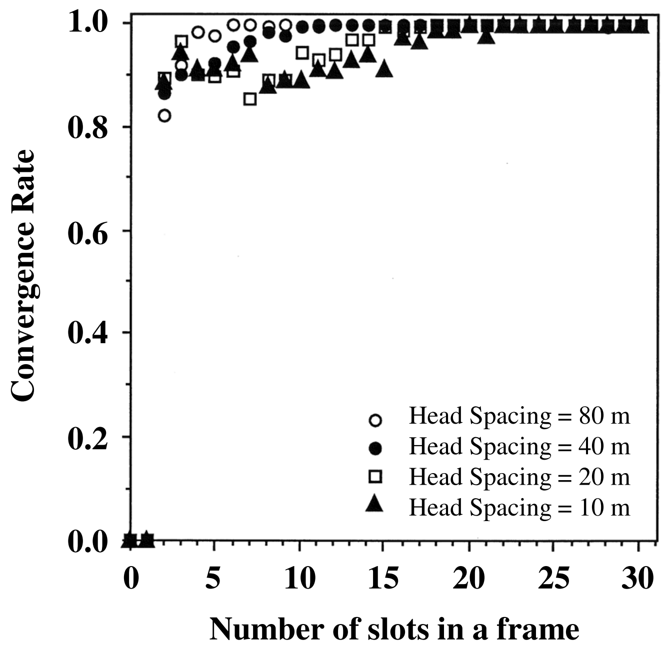

Fig. 5.5 Convergence of the slot reservation algorithm.

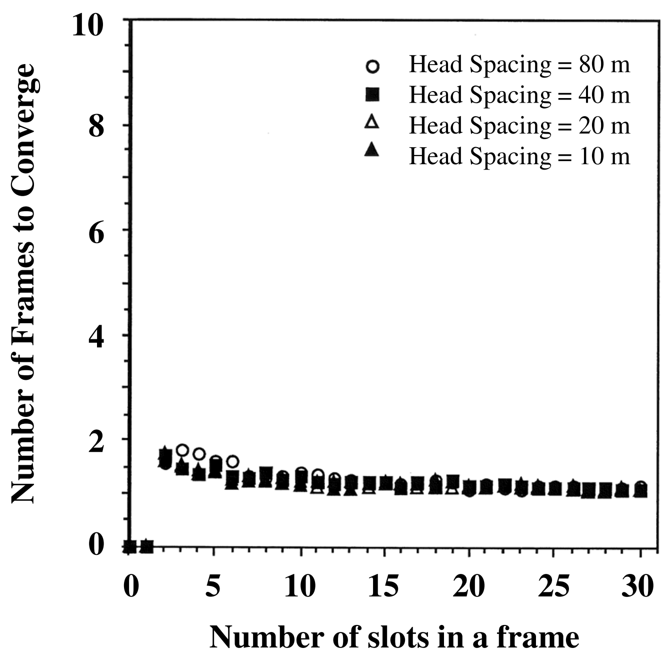

Fig. 5.6 Number of frames to complete slot reservation.

converges within two frames and it will be acceptable for the actual environment if there are more than 15 slots in a frame.

As shown in Figs. 5.5 and 5.6, the slot reservation algorithm may not converge when the number of the slots in one frame is less than 15. The smaller average head spacing means that the density of the vehicle is high. In such situation, the slot reservation algorithm will not be converged since there will be more number of cars that can be accommodated.

However, the convergence rate becomes almost 1 for all cases of the average head spaces if there are more than 20 slots in a frame. This number relates to the density and the communication range of the vehicle. For the scenario considered in this study, 20 slots/frame seems to be acceptable value to accommodate the vehicles.

There will be other ways to improve the convergence rate of the slot reservation algorithm. An example is the use of transmission power control considering the density of the vehicle. It might be difficult for a vehicle to estimate the density of the vehicle in a given situation. In such a case, it will be good if road side infrastructure could collect environmental information and provide it to the vehicle using the V2I communication.

5.5.3 Performance Evaluations

Figures 5.7 shows the success probability of Reply Packet. The x-axis is the number of slots in one frame and the y-axis is the success probability of a Reply Packet. It is assumed that a Reply Packet is successful if more than one vehicle(s) in proximity received it correctly. If there are more than 15 slots in a frame, a success probability of a Reply Packet becomes more than 95% and reliable communications can be made.

Figure 5.8 shows the number of vehicles in a network. It is observed that the number of the vehicle in a network increases as the number of the slot and the density of vehicles increase. However, the increase rate gradually moderate when the number of slots in a frame exceeds 15 or 20. Therefore, it can be said that a vehicle can communicate with other vehicles in proximity if there are 15 or 20 slots in a frame.

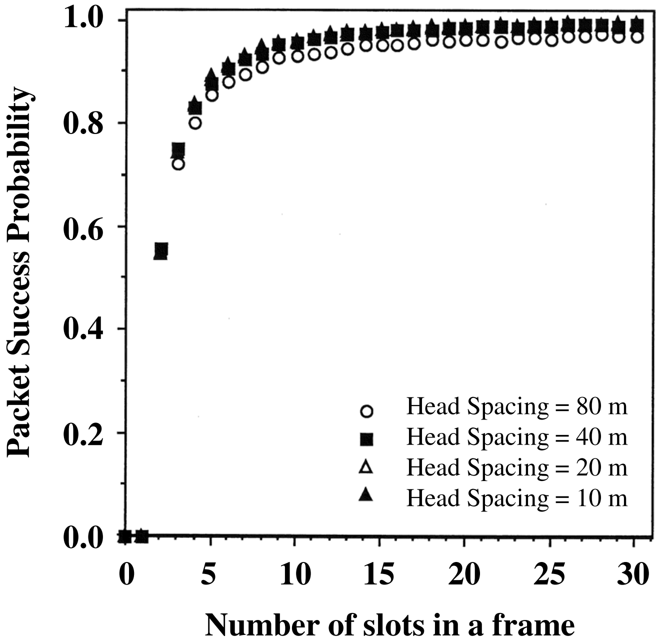

Fig. 5.7 Success probability of a reply packet.

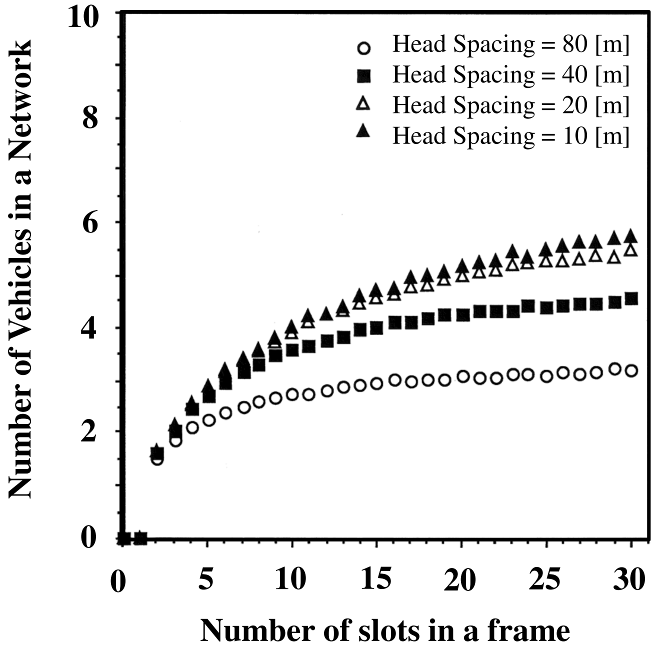

Fig. 5.8 Number of vehicles in a network.

5.6 Summary

In order to improve road traffic safety and car infotainment, inter-vehicle communication network plays an important role. In this chapter, MAC protocol for short range inter-vehicle communication network was proposed. The R-ALOHA protocol was modified to adopt the SS Boomerang communication technique as the PHY layer scheme. The channel structure was proposed accommodate the transmissions of PN sequences to retrieve the information of the slot owner, transmission of the data by the slot owner and the acknowledgments. Autonomous slot reservation algorithm was also presented. The basic performance of the proposed MAC protocol was evaluated by the computer simulations. It was observed that the slot reservation algorithm converges when there are 15 or 20 slots in a frame. In such a case, a vehicle can choose the slot within 2 or 3 frames and reliable data exchange with the vehicles in proximity can be achieved by regular and frequent information exchanges.

Chapter 6

Conclusions

In this thesis, techniques to improve the efficiency and functionalities of the distributed access control for the wireless systems and networks have been studied. Originally the distributed access control mechanism was designed to coordinate the access control in a distributed manner and was suitable for transmissions of the best effort data. However it is expected to have better efficiency and more functions to support various kinds of services and applications and to improve the user experience. For those purposes, following techniques have been studied.

Improvement of the Management Function

For the better spectrum utilization, management functions of the wireless system need to be improved. For this purpose, the impact of rate switching algorithm on the system capacity was analyzed based on the IEEE 802.11a system, and a simple modification to the history based rate switching algorithm was proposed in chapter 2. This is an example of the cross-layer optimization techniques to achieve high spectrum utilization.

The benefit of the history based rate switching is that no feedback mechanism is required for the purpose of the rate selection which means no additional overhead is necessary. A typical rate switching algorithm makes decision to use a higher or a lower transmission rate after continuous success or failure of transmissions, respectively. By introducing the new parameter, the proposed rate switching algorithm improves the system capacity of the IEEE 802.11a network from 14 to 20%.

Enhancement of the Functionalities

In chapter 3, communication quality control schemes were discussed based on the IEEE 802.11 WLAN. By extending the polling based data transmission procedure called PCF, the parameterized QoS can be provided in a managed, i.e. interference free, environment. Moreover, a technique to provide prioritized QoS was discussed based on the basic access procedure of the WLAN called DCF which is more suitable to be used in the unlicensed frequency bands where coexistence with other WLAN network and/or even with other systems is required.

The point of providing the parameterized QoS is queue management and reordering of the data considering the QoS parameter associated to it. The prioritized QoS is achieved by assuming multiple queues in a STA and by doing contention control by using access parameters associated to each priority. Simulation results were presented for each of the parameterized and prioritized QoS schemes and communication quality control can be achieved by the proposed procedures.

Note that the concept of using different contention window sizes depending on the priority class was adopted in the Enhanced Distributed Channel Access (EDCA) mechanism defined in the IEEE 802.11e standard.

Chapter 4 presented a reliable multicast protocol for the WLANs called the representative acknowledgment scheme. To improve the reliability of the multicast data in the wireless region, acknowledgments are returned for each multicast frame.

In order to reduce the number of acknowledgments, the proposed protocol assumes the STA groups to be created in a service area of an AP and the acknowledgment is returned not from each STA but from each of the STA group. By limiting the number of acknowledgments only from the representative STA of each STA group, both reliability and efficiency is obtained. With the proposed multicast protocol, the transmission time for a multicast frame can be reduced from 48% to 28% while keeping the reliability of a multicast frame of 99.99%.

Support for the Emerging Applications

Finally, chapter 5 discussed an vehicle-to-vehicle communication network and its MAC protocol as an emerging application of the distributed wireless network. In order to improve the road traffic safety, a vehicle is assumed to do cyclic information exchange with other vehicles in proximity. In the vehicle-to-vehicle communication network, data transmissions and network management have to be done in a distributed manner. Considering those requirements, an R-ALOHA based communication protocol and slot reservation algorithm were studied.

Future Works

In this thesis, MAC layer functions and layer management functions of the distributed wireless systems and networks have been studied to support various kinds of applications efficiently. Each function discussed is a building block of the distributed wireless systems and networks.

There will be some directions that we can think about for further improvement of the distributed wireless systems.

Layer Management and Cross-Layer Optimization

As the advanced PHY techniques such as multi-user MIMO (MU-MIMO) and Orthogonal Frequency Division Multiple Access (OFDMA) are introduced, the importance of management functions and cross-layer optimization techniques will be significant. For example, the multi-user MIMO technique introduced by the IEEE 802.11ac standard is an effective way to improve the system capacity by transmitting data for different users simultaneously. When thinking about the actual environment, however, the traffic flows will have different QoS requirements and the destination STAs will have different characteristics in terms of radio propagation. In order to take advantage of the MU-MIMO technique, management functions such as pairing and scheduling considering the actual environment and traffic characteristics will be important.

Upper Layer Functions

In order for the better collaboration of different wireless systems such as cellular and wireless LAN, without degrading user experience, upper layer functions will be important.

For security related items, authentication, authorization and accounting (AAA) and key management for encryption will play important roles for interworking of different systems. From the user's point of view, another important feature will be the session continuity while handover between the different systems.

All those features discussed above will be important to provide various services over the distributed wireless systems and networks in an efficient and cost effective way to satisfy the user experience. Therefore, further researches and studies are expected to improve the distributed wireless systems and networks.

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Awards

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