Quality-of-Service Control Scheme for Wireless Local Area Networks

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Abstract

Progress in the Internet services and the development of high-performance communication devices have promoted rapid adoption of broadband wired and wireless communications. Especially with regard to wireless communications, the number of wireless local area network (WLAN) users has exploded in recent years. Therefore, this thesis deals with techniques related WLANs and wireless sensor networks, which represent a sophistication extension of WLANs.

Wireless communication suffers from several inherent problems that make its throughput lower than that of wired communication and its bit error rate higher than that of wired communication. Effective solutions include Multiple-Input Multiple-Output (MIMO), multi-level modulation schemes, and powerful error correction codes. Although these advances can satisfy user demands, which are stimulated by the advanced features of wireless terminals, the advances are not enough. In order to use the limited radio resources efficiently and to provide services with the highest possible quality, it is so important to achieve fair use of all radio resources. Quality-of-service (QoS) control technology in the medium access control (MAC) layer can use the limited radio resources to guarantee the desired service quality of each user. In this thesis, the problems of conventional technologies in achieving of services that distribute throughput equally and guarantee transmission delay, etc. are clarified and solutions to these problems are described.

In order to realize QoS control in wireless communications, radio resource control in the MAC layer is indispensable because the MAC layer protocol achieves important role to guarantee QoS. There are two kinds of resource control schemes for wireless communication system: centralized access control and distributed access control. The former are mainly used to realize asynchronous transfer mode (ATM) transport, while the latter are used to realize of Ethernet mode transport in wireless communication. These have developed as transmission modes for specific technical domains.

In Chapter 3, we clarify the problem of the drastic increase in transmission delay when, under the centralized access approach, only one kind of resource request method is employed. A new resource management scheme is proposed that dynamically selects one polling scheme or one random access scheme as a radio resource request method so as to guarantee the minimum transmission bandwidth for each user while fairly sharing the surplus bandwidth. In this chapter, we also reveal the effectiveness of the proposed scheme in providing smaller transmission delay than the conventional schemes. Chapter 4 introduces a new transmission control scheme that offers stable transmission rate control and guarantees fair share of throughput even when the round trip time (RTT) of the radio link is large for ATM mode under the centralized access approach. The proposed scheme predicts the transmission rate of each wireless terminal and controls it by applying the

virtual destination virtual source (VD/VS) technique. This thesis also shows the effectiveness of the proposed scheme. Chapter 5 introduces a throughput control scheme that guarantees throughputs for each user under the distributed access approach; it overcomes the difficulty of guaranteeing throughput in a distributed access system. In order to solve the problem, this thesis proposes a new closed-loop transmission rate control scheme in which an access point informs the wireless terminal of congestion status by setting one information bit in the ACK frame. In Chapter 6, this thesis discusses how to guarantee the transmission delay of machine to machine (M2M) communications systems, which are expected to receive a lot development in the near future. In M2M communication, one access point accommodates a large number of wireless terminals. Given the dynamic range of traffic received by an access point, it is difficult to guarantee the transmission delay of each wireless terminal. Our solution is a new transmission delay control scheme that guarantees transmission delay by controlling the transmitting rate of each wireless terminal by computing the optimum access parameters values from the amount of traffic that an access point receives. Finally, Chapter 7 summarizes the research results of this thesis.

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Acronyms

ABR	Available bit rate
AC	Access category
ACK	Acknowledgement
ACR	Allowed cell rate
AEDCA	Adaptive EDCA
AIFS	Arbitration interframe space
AP	Access point
ARQ	Automatic repeat request
ARQ-NAK	ARQ negative acknowledgement
ATM	Asynchronous transfer mode
BCCH	Broadcast channel
Bch	Broadcast channel
BLD	Backward-link-delay
BRM	Backward RM
Cch	Control channels
CD	Collision detection
CDV	Cell delay variation
CI	Congestion indication
CLR	Cell loss rate
CRC	Cyclic redundancy check
CSMA	Carrier sense multiple access
CSMA/CA	Carrier sense multiple access with collision avoidance
CSR	Cell stream regeneration
CW	Contention window
CWmax	Contention window maximum
CWmin	Contention window minimum
DA	Demand assignment
DCF	Distributed coordination function
Dch	Data channels
DES	Destination end system
DLC	Data link control
DSA	Dynamic slot assignment
DSSS	Direct-sequence spread spectrum

EDCA	Enhanced distributed channel access
EFCI	Explicit forward congestion indication
EPRCA	Enhanced proportional rate control algorithm
ER	Explicit rate
FCCH	Frame control channel
FHSS	Frequency hopping spread spectrum
FLD	Forward-link-delay
FRM	Forward RM
HP	High priority
ICR	Initial cell rate
IFS	Inter frame space
IMT	International mobile telecommunications
IWS	Initial back-off window size
LCCH	Logical control channel
LP	Low priority
M2M	Machine-to-machine
MAC	Medium access control
MCR	Minimum cell rate
MIMO	Multiple-Input Multiple-Output
MT	Mobile terminals
OFDM	Orthogonal frequency division multiplexing
PAR	Project authorization request
PCR	Peak cell rate
PDU	Protocol data unit
PER	Packet error rate
PF	Persistent factor
PHS	Personal handyphone system
PHY	Physical
QAM	Quadrature amplitude modulation
QoS	Quality-of-service
QPSK	Quadrature phase shift keying
RA	Random access
RACH	RA channel
Rch	Random access channels
RDF	Rate decrease factor
RFCH	RA feed-back channel

RIF	Rate increase factor
RM	Resource management
RMS	Root mean square
RTT	Round trip time
SAR	Segmentation and reassemble
SAS	Slot assignment scheduler
SDU	Service date unit
SES	Source end system
STA	Station
TDD	Time division duplex
TDMA	Time division multiple access
TX	Transmission
TXOP	Transmission opportunity
UDCH	User data channel
UDP	User datagram protocol
VC	Virtual channel
VD	Virtual destination
VD/VS	Virtual destination/virtual source
VS	Virtual source
WALN	Wireless local area network
WATM	Wireless asynchronous transfer mode
WATM-RTT	WATM-associated RTT
W-CDMA	Wideband-code division multiple access
WEP	Wired equivalent privacy
WS	Window size
WT	Wireless terminals

Nomenclature

Operators

Max(a,b)	Maximum of a and b
Min(a,b)	Minimum of a and b

Scalar

Greek Symbols

τ_h	Transmission probability of HP class
τ_l	Transmission probability of LP class

Latin Symbols

ACR_k	k-th allowed cell rate
ACR_{VS}	Allowed cell rate on the VS side
BLD	Backward link delay
CCR	Current cell rate
CI_N	N-th congestion indicator
FLD	Forward link delay
I_N	N-th interval between BRM cell transmissions
L_d	The depth of thebucket
L_{d_prev}	The depth of thebucket before updating
L_r	Received user data size
MCR	Minimum cell rate
N	Number of WTs
$Na(n)$	Number of slots to be assigned to MT#n
N_c	Number of cells received on the VD side between t and t + TRT
N_d	Threshold for burstiness judegement.
n_h	the number of HP class WTs
n_l	the number of LP class WTs
$Nl(n)$	Limitation for MT#n
Nl_max	Maximum number of slots to be assigned
Nl_min	Minimum number of slots to be assigned
Nl_up	Limitation increase number
Npmt	Number of MTs which AP assigned insufficient number of slots to.
Nr(n)	Number of slots requested by MT#n

Nr(n)	Number of slots requested by MT#n
Nrb(n)	Number of slots previously requested by MT#n.
N_{remain}	Number of slots which remain unused.
p	Random nubmer within right-open interval from 0 to 1
PCR	PCR
p_{gh}	Data generation probability of HP class
p_{gl}	Data generation probability of LP class
p_h	RA failure probability of HP class
p_l	RA failure probability of LP class
Q_S	Future queue length
Q_{SC}	Current queue size
Q_T	Predetermind queue threshold
RDF	Rate decrease facotr
Rg	Agreed rate
RIF	Rate increase factor
S	Throughput
t	current time
T_{cd}	Predetermind constant delay
T_{erss}	Effective RA slot length
T_M	MAC frame period
T_{now}	Current time
T_{prev}	the time at which the previous data frame was received
T_{RT}	WATM-RTT
W_{0h}	Initial backoff window size of HP class
W_{0l}	Initial backoff window size of LP class

Chapter 1
Introduction

About twenty years after the telephone was invented, Guglielmo Marconi, an Italian inventor, carried out the first radio transmission between the Isle of Wight and a tugboat 18 miles away. Thus, this is where the history of wireless communication began. Since then, the development of wireless communication systems has aimed at better communication quality and longer communication distances. Nowadays, there are many different wireless communication systems, including terrestrial fixed wireless communication, satellite communication, mobile wireless communication, and terrestrial and satellite broadcasting. Among these, mobile wireless communication systems have rapidly developed in recent years. In mobile wireless communications, the communication apparatus (i.e., the terminal) can be carried in movable objects, such as motor vehicles, railroad cars, seagoing vessels, and airplanes. The classification also includes man-portable terminals. The driving forces behind the development of mobile wireless communication systems are the wireless local area network (WLAN) and the mobile phone; the use of mobile phones, in particular, is increasing by leaps and bounds. Mobile phones are used not only for business but also for personal use. Many adults use mobile phones, and the number of children using mobile phones is growing. In fact, the number of contracts for mobile services including personal handyphone system (PHS) has exceeded

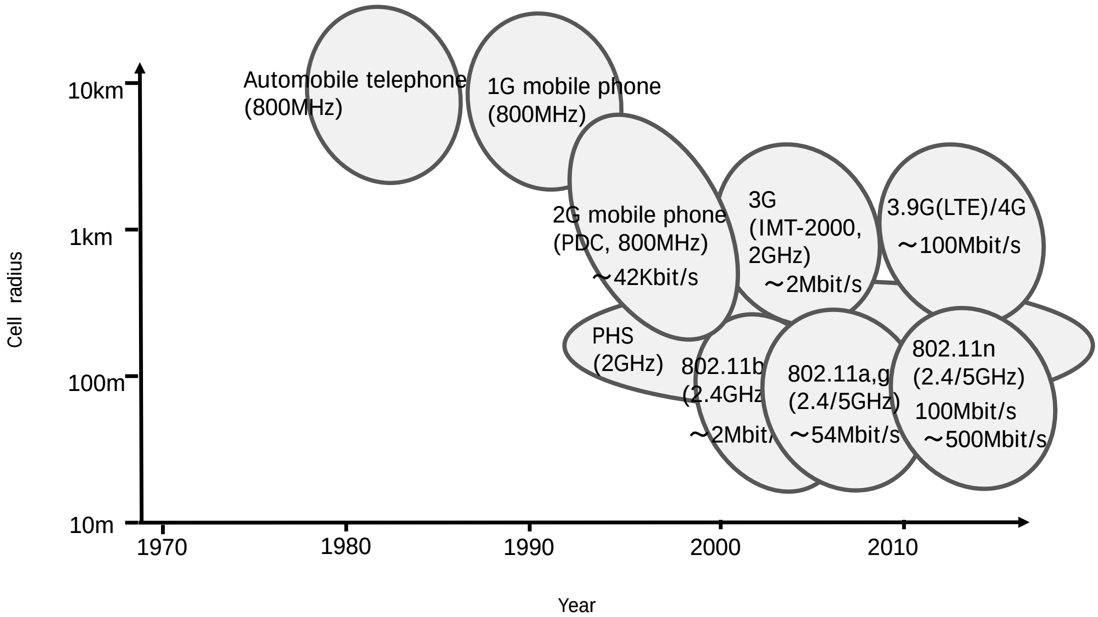


Fig. 1-1. Wireless communication systems and their cell radius.

the population of Japan. Statistically speaking, this means every Japanese person has one or more mobile phones.

Figure 1-1 shows wireless communication systems and their cell radius. Access speed has been improving with progress of technology. However, the cell radius became small in order to improve the efficiency of frequency utilization. A brief description of the history of mobile phone development is as follows. The advances in electromagnetic technology made during World War II led to the first commercial mobile phone service being deployed in the U.S. in 1946. This system used large zones and operated in the150 MHz band. In order to accommodate more users and to exploit radio resources effectively, the large zone system was eventually replaced with a small zone system made possible. In Japan, research and development on automobile telephone systems using analog transmission started in the late 1950s. An automobile telephone service using the 800 MHz band started in Tokyo in 1979. Such systems are the basis of today's mobile phone systems.

The large zone system transmitted electromagnetic waves over long distances by using a high gain antenna and high transmission power. A small zone system, on the other hand, used two or more base stations covering a comparatively small area called a cell. Cellular systems such as this appeared in various countries at the end of the 1970s and in the early 1980s. The most important feature of cellular systems is their repeated use of the same frequency in two or more base stations separated by some distance. If the cell radius is small, the same frequency can be used by base stations separated by shorter distances; this improves frequency utilization efficiency. The 2nd generation cellular system made the change from analog to digital. The personal digital cellular (PDC) system, a digital cellular system, was put into service in 1994. This development was followed by the widespread deployment of international mobile telecommunications (IMT)-2000 or 3G systems. In particular, the wideband code division multiple access (W-CDMA) system started service in October 2001 and CDMA 2000 1X service started in April 2002.

As the telephone of the wired network developed into the mobile phone, also in the data communications represented by the Internet access, it started in the wired network and developed into wireless communications. The beginnings of radio packet communications can be traced back to 1971. The early wireless systems used analog signals, whereas the wireless packet communications system ALOHANET [1]-[3], developed at the University of Hawaii in 1971, communicated with digital bit strings that were turned into signals called packets and transmitted in burst signals. ALOHANET connected a central computer on the island of Oahu to computers on the other Hawaiian islands through 9600 bit/s digital radio links. The network architecture had a star topology in which the central computer establishes a two-way communication link to other computers. ALOHANET's wireless data packet communication was enabled through a protocol including a radio channel access scheme and a routing method.

In 1991, a Project Authorization Request (PAR) was submitted to the IEEE 802 committee on establishing a standard for wireless local area networks (WLANs) in which mutual connections would be possible. The IEEE 802 committee followed its usual rules and moved towards making a standard that would be applicable in every corner of the Earth. A task group for a 2.4 GHz band system was established. A draft of the standard was submitted for approval in 1993, and it was recognized as the IEEE 802.11 standard in June, 1997. The first 802.11 standard supported data-communication speeds of 1 Mbit/s or 2 Mbit/s, and defined data transmission systems using infrared rays or the 2.4 GHz radio band, data encryption, wired equivalent privacy (WEP), direct-sequence spread spectrum (DSSS) modulation, and frequency hopping spread spectrum (FHSS) modulation. However, since DSSS and FHSS were specified in the standard as a compromise between manufacturers, the principles of their operation were incompatible with each other, and when they were used at the same time, they caused mutual interference that reduced their throughput. Despite this drawback, the first 802.11 standard opened up a completely new era of versatile communications and became the foundation of future WLAN development. After that, WLANs adopted orthogonal frequency division multiplexing (OFDM) modulation technology, which used quadrature phase shift keying (QPSK) and quadrature amplitude modulation (QAM), and their transmission rates were accelerated. This development affected the development of the mobile phone systems greatly, and the OFDM modulation was adopted in 4G cellular systems.

With the spread of WLAN technology, interest began to grow in the idea of wireless "machine to machine", or M2M, communication. Figure 1-2 shows the concept of an M2M network. The main feature of M2M communication is that the apparatuses operate autonomously and communicate without human intervention. The basic components of an M2M network are sensors that detect certain states in the apparatus or the environment and actuators for operating something. Since the purpose of the network is mainly to send information from sensors and measuring instruments and issue control commands, M2M communication needs to transmit only a small number of bytes at high speed correctly. Moreover, since M2M networks would be useful in places where electric power is scarce or impossible to provide, the communication devices must have extremely low power consumption and be able to operate for several years by battery. An M2M network which uses radio links for the whole access system is called a wireless M2M. Wireless M2M networks using a radio link are free of the cost of laying and maintaining telecommunication cables, meaning that they are in principle low cost. Moreover, since wiring is unnecessary, wireless M2M networks can be easily extended after installation. Furthermore, wireless M2M can be used in homes, offices, stores, hospitals, schools, and factories. The service scope is very broad and thus M2M networks would be ideal for a wide range of monitoring purposes, including electricity, gas, and water metering, disaster prevention, environmental monitoring, energy monitoring, factory automation, vending machines, inventory control, health care, smart household appliances, physical distribution surveillance, and agriculture. There is a wide area ubiquitous wireless network as a

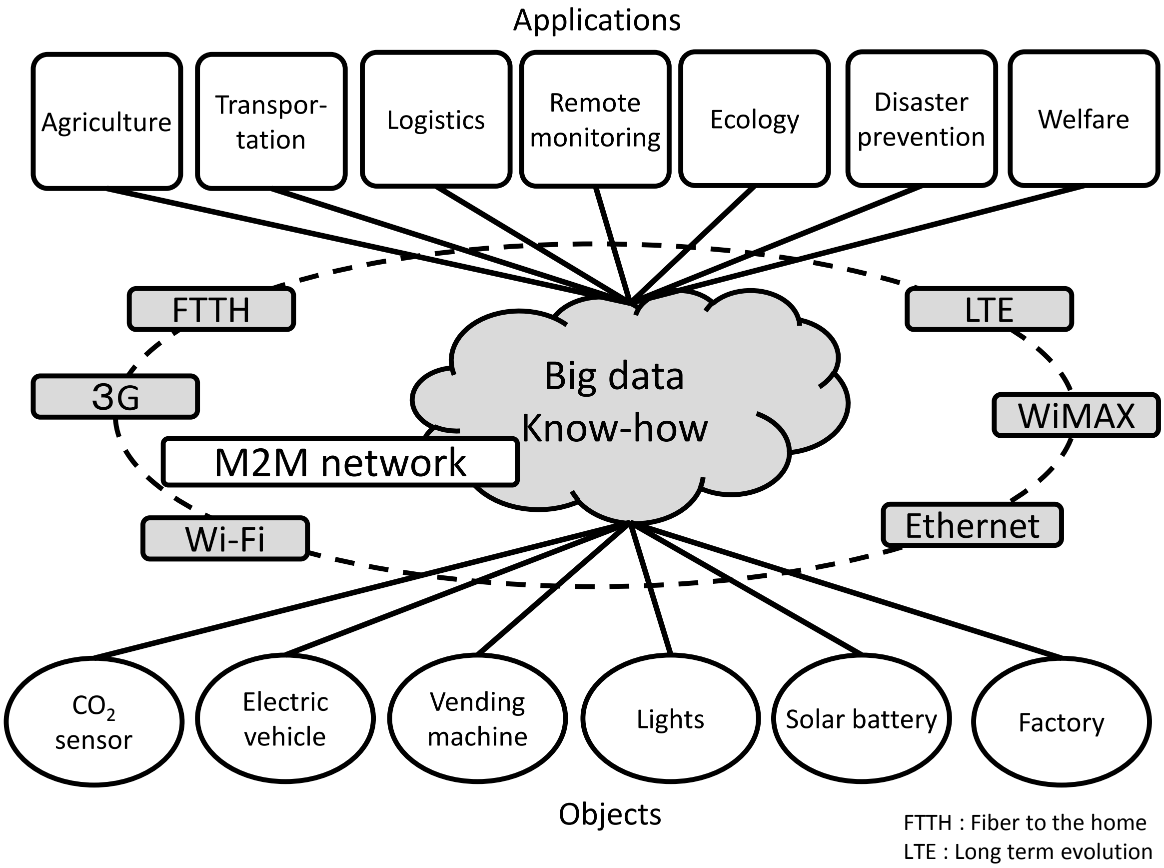


Fig. 1-2. Concept of M2M network.

means to realize wireless M2M networks cost-efficiently. In the wide area ubiquitous wireless network that is currently being developed and is considered in this thesis, the radio link transmission rates are very low in order to enlarge the service area, and each base station (BS) will accommodate many wireless terminals (WTs) because the sensors will send comparatively short data packets at long transmission intervals. Moreover, the network needs to provide users with a wide variety of services. Therefore, it needs to have several Quality-of-service (QoS) levels.

Our interest lies in a mobile communications system, in particular, a radio communications system that transmits data in packets (e.g., a WLAN or wireless M2M). We will study its QoS control schemes.

1.1 Wireless local area networks

The greatest factor that restricts the access speed of WLANs is multipath propagation. Multipath propagation is a situation in which the transmitted signal arrives at a receiving antenna through two or more paths. In a multipath environment, signals come from different directions, i.e., directly from the transmitting antenna and reflected off walls, fixtures, etc., and two or more signals are compounded intricately at the receiving antenna. Moreover, these multipath signals differ in their received power level, phase, and time of arrival. In such cases, multipath distortion and inter-symbol interference get worse as the delay of the signals and the difference in the power level increase.

The OFDM modulation scheme has achieved a high transmitting bit rate in a multipath environment. OFDM divides high-speed data into many low-speed data rows and performs parallel transmission using many subcarriers. Multipath distortion can be reduced by using many low-speed data rows, since the symbol time of each subcarrier signal becomes relatively long compared with the root mean square (RMS) delay spread. In IEEE 802.11, WLANs use access control technology so that two or more terminals can share the same radio channel. This access control technology is called carrier sense multiple access with collision avoidance (CSMA/CA), and it is a successor to CSMA with collision detection (CD) that had been used with cabled Ethernet. In particular, the CSMA/CA protocol is used for the access control of the distributed coordination function (DCF). Here, each radio terminal detects whether the channel is in use and determines the transmit timings of packets autonomously. For this reason, even if two or more radio cells using the same channel overlap on the whole or partially, the same channel can be shared and wireless terminals which exist in the radio cells can communicate.

1.2 Quality-of-service control scheme

Communication quality can be guaranteed comparatively easily if more than enough radio resources (in terms of transmission power and radio frequency bandwidth) can be assigned. For example, throughput can be guaranteed if the number of terminals accommodated in one base station is smaller than the system throughput divided by the guaranteed throughput. However, the radio resources will not be used effectively if the maximum amount is always assigned. In addition, considering the present situation where wireless communications are briskly used for various purposes, it is strongly desirable to further raise the utilization efficiency of radio resources.

The limited radio resources can be effectively managed with a QoS control scheme. Such a scheme ensures that a certain parameter has the desired value. QoS control schemes can be classified according to how changes in communication quality are detected, how detection results are judged, and how communication is controlled using the judgments. In wireless communications, the parameters that can specify QoS include the bit error rate, packet error rate (PER), transmission delay time, transmission delay time dispersion, and throughput. The bit error rate and packet error rate in wireless communications may be much higher than those of wired network communications, and besides error correcting codes, there are various automatic repeat request (ARQ) schemes, such as Stop & Wait, Go back to N, and Selective repeat, that can be used.

ARQ is an error correcting system that works by resending data, and it can guarantee an arbitrary PER by enabling the user to choose the maximum number of times one unit of data can be sent. For example, let us say that the PER in a radio link is 10^{-2}. By using ARQ, it can make the PER of the radio link equals to or less than 10^{-20} by setting the maximum number of resending times to nine. However, supposing again that the maximum number of resending times is nine, the maximum transmission delay of the packet may be up to 10 times the delay when the system does not have a resending mechanism. However, the probability that the maximum transmission delay may occur is very low when the number of resending times is large, so there are very few cases in which the QoS is actually determined by the maximum transmission delay. In other words, there aren't many QoS control schemes that dynamically control the maximum number of resending times and the maximum number of resending times tends to be a constant system parameter.

The QoS control schemes offering a throughput guarantee can be further classified into ones that guarantee the average throughput, ones that guarantee the minimum throughput, and hybrids thereof. Moreover, the IEEE 802.11e standard (2005) adds QoS control to the MAC layer in response to the spread of applications requiring low delay, such as VoIP and video streaming. Enhanced distributed channel access (EDCA) is a scheme that offers QoS through a random access mechanism. In EDCA, the desired QoS is guaranteed by choosing an arbitration interframe space (AIFS) length and contention window (CW) size for every access category (AC). Moreover, in EDCA, exclusive channel use is enabled by transmission opportunity (TXOP) control. In EDCA, AIFS is used as the carrier sense time according to the AC priority. Priority control between ACs is achieved by giving a short AIFS to ACs with high priority. In the same way as with AIFS priority control, smaller CW size is assigned to a higher priority AC. TXOP is a parameter determining the period during which a wireless station can use a channel exclusively after it has acquired the right to access it. The TXOP of an AC with high priority is not necessarily longer than that of an AC with low priority. Rather, the length of the TXOP is decided in consideration of the packet length of the application. If the TXOP for a certain AC is zero, the number of packets that can be transmitted after acquiring the right to access is one. Although the priority control in EDCA has a different access parameter for every AC, the access parameter in one AC has the same value and may be unable to guarantee QoS in cases such as when there is a drastic change in the amount of traffic.

In wireless M2M, devices autonomously communicate without a human having to perform any operations. Despite the potential convenience of such automatic operation, communication may be difficult to maintain when the outside propagation environment changes. If sufficient consideration is not put into a design, there is a possibility the system won't be able to deal with changes. Therefore, an adaptive control mechanism is required, and it must be able to control a system in consideration of various factors that are found in QoS control.

1.3 Problems affecting wireless quality-of-service and the contributions of this thesis

Following the advances in the features of wireless terminals, the number of WLAN users has exploded in recent years. This thesis deals with techniques related to WLANs and wireless sensor networks, which represent a sophistication extension of WLANs. Generally, compared with wired communications, the data-transmission capacity of wireless communications is highly variable and the bit error rate is high. Therefore, one continuous effort has been to raise communication quality as close to that of wired communications as possible. The metrics that indicate communication quality are mainly throughput, throughput fairness, transmission delay, and transmission delay variation. With respect to throughput, the use of Multiple-Input Multiple-Output (MIMO) and multi-level coded modulation techniques has improved transmission speed to the extent that it has become equal to that of wired communication in recent years [4]-[6]. Even if neither bandwidth nor transmission power is strengthened, MIMO technology substantially improves throughput and lengthens communication distance. Multi-level coded modulation techniques raise the number of data bits transmitted per symbol in digital wireless communication and they can improve radio resource utilization efficiency and significantly increase the data-transmission capacity of wireless links.

Throughput fairness is a measure that shows whether radio resources can be equally used by all wireless stations. For example, even if a certain transmission control scheme achieves high system throughput, it cannot be considered to offer high communication quality if it does not assign radio resources equally to each wireless station in the system. That is, a good wireless communication system needs to offer both high throughput and fairness. Particularly in today's terminal environment where each wireless terminal runs two or more applications, a wireless communication system needs to provide both services that guarantee throughput to a terminal and best effort services that distribute surplus radio resources equally. To take one example, a single radio terminal may need to perform real-time applications that require a fixed traffic level, such as providing video streams or voice phone calls, and non-real-time applications requiring best effort service, such as providing Web access or e-mail services. In that case, it is necessary to guarantee minimum throughput to the real-time applications and assign radio resources fairly to the non-real-time applications.

A number of schemes have been proposed that assign radio resources to each terminal constantly in order to guarantee throughput. Other schemes have been proposed that provide best effort services to guarantee fairness. However, there have been no reports of schemes that guarantee minimum throughput with best effort service while assigning surplus radio resources to best effort services equally and effectively. In Chapter 3 we describe a radio resource management scheme that assigns surplus radio resources efficiently so as to guarantee both minimum throughput and fairness.

The proposed scheme dynamically selects one polling scheme or one random access scheme as a radio resource request method in order to exploit radio resources efficiently. The proposed scheme keeps the MAC delay of normal terminals within seven frames even when heavy traffic terminals exist.

With the goal of mobile multimedia communications, demands for wireless asynchronous transfer mode (WATM) have always existed and a number of schemes that use radio links to accommodate asynchronous transfer mode (ATM) terminals have been studied. ATM systems have an available bit rate (ABR) service that offers throughput fairness across the entire network [7]-[9]. This service maintains fairness in the whole end-to-end connection. It ascertains the congestion situation in a connection and uses closed loop control to control the transmitting rate of the terminal. The problem with it, however, is that fairness collapses when round trip time (RTT) in the control loop is long and control delay is large. The virtual destination/virtual source (VD/VS) option has been proposed as a solution to this problem. This option maintains fairness by dividing one big loop into two or more small loops and shortening the control delay time of each loop. In the event of excess loop delay, the ABR service can maintain fairness by using VD/VS. However, the conventional VD/VS technique cannot solve the problems caused by the delay that occurs in a radio link because it cannot divide the link's control loop. In Chapter 4 we examine a scheme that guarantees fairness through the whole connection including the radio link and introduce a new transmission control scheme that offers stable transmission rate control and guarantees fair throughput sharing even when the round trip time (RTT) of the radio link is large. The proposed scheme can make the fairness index completely converge to one without fluctuation even when the RTT is as large as 8 msec.

As mentioned at the beginning of this thesis, the driving forces behind the development of mobile wireless communication systems are the WLAN and the mobile phone; the use of mobile phones, in particular, is increasing by leaps and bounds. When discussing the communication quality in WLANs one cannot fail to mention the distributed control method, which is the dominant system in the market today. On the other hand, if throughput guarantees are to be considered, centralized control schemes can more easily manage the throughput of each wireless terminal than distributed control schemes since they manage all the radio resources by themselves. So far, many centralized throughput control schemes have been proposed. However, only a few attempts have been made to study throughput control schemes based on distributed MAC protocols. One such scheme that differentiates priority for each QoS class has been proposed. However, no studies have been made of control schemes that quantitatively guarantee throughput at a certain predefined rate. Therefore, in Chapter 5 we discuss throughput control schemes that are based on a distributed MAC protocol and that guarantee different rates for each wireless terminal. The propose scheme employs a new closed-loop transmission rate control mechanism in which an access point informs the wireless terminal of congestion status by setting one information bit in the ACK frame. The propose scheme keeps the throughput of each high priority wireless terminal at the required rate regardless of the number of low priority terminals.

Since the range of traffic fluctuation increases with the number of terminals accommodated by one AP, it is more difficult to control the transmission delay to a desired value. In present-day WLANs, each AP accommodates several tens of wireless terminals. However, as M2M communication networks become more widespread it is expected that they will have 100 or even 1000 times more terminals per AP than WLANs. This is because the wireless terminals of M2M networks transmit less data than those of WLANs and their transmission intervals are very long. Chapter 6 of this thesis discusses a transmission delay control scheme based on distributed MAC protocols that functions effectively even when an AP accommodates several tens of thousands of wireless terminals. In this chapter, we clarify the problem of conventional transmission delay control schemes and propose a new solution that guarantees the transmission delay by controlling the transmitting rate of each wireless terminal by computing the optimum access parameters values from the amount of traffic that the AP receives. The proposal prevents the transmission delay of high priority terminals from increasing more than fifty percent regardless of the number of low priority terminals.

Chapter 2
Quality-of-Service Control Scheme for Wireless Local Area Networks

This chapter introduces QoS of network communications. First, the need for QoS and the parameters that indicate the level of QoS are shown. The inherent problems of wireless communications, which include lower throughput and higher bit error rates, make QoS guarantees more difficult to achieve in wireless communication than in wired. Therefore, techniques beyond those used in wired communications are needed for wireless communication. In particular, in order to exploit the limited radio resources efficiently, new techniques for the MAC layer are needed. This chapter also introduces some wireless multiple access protocols so as to assist make the proposals described in later chapters easier to understand.

2.1 Quality-of-service

QoS expresses the quality of service offered on a network. QoS is represented by various indices, such as throughput, transmission delay time, and fairness. QoS is an important parameter showing network validity, and applications that need a QoS guarantee have been increasing in recent years with the spread of networks. Since the beginning of the 1990s, real-time applications, such as voice telephone calls and video delivery, are being hosted more frequently on the Internet. When exchanging voices and real time videos across a network, delay in data communications takes forms that users can readily perceive, such as one-way pieces of conversation and dropped video frames. Therefore, the delay in data communications is directly linked with degraded quality. In fact, a network must handle various data flows. Occasionally, large volumes of data are injected into the network, causing congestion which may interfere with the functioning of the network and degrade network performance. This raises the importance of QoS control; it is needed to clearly handle and segregate data flows with the priority demanded by the user by keeping one or more measurable QoS metrics within certain specific values. In the broadband era, the necessity for QoS is high. Although the cost per bit to transmit has been reduced sharply, users have become more demanding in terms of service quality even while asking for greater communication bandwidth and greater numbers of service that rely on high speed reliable communication. With the emergence of the multimedia age, which circulates rich contents such as high resolution images and full sound, QoS control is attracting even more attention because demands for different QoS levels for every application continue to increase.

The greatest factor pushing the need for QoS control is congestion. If there were no congestion, there would be no need to guarantee QoS. However, congestion does indeed occur in almost every part of network and triggers many other problems. In the presence of congestion, the full communication bandwidth required for each terminal is unavailable so data transmission sessions take longer to complete, and packet loss due to buffer overflow in entities such as routers and L2 switches becomes more frequent. Methods of avoiding congestion in a network include admission control. In admission control, when establishing a connection, a service request is refused if its QoS level cannot be guaranteed. Admission control is effective when the traffic can be predicted beforehand and the required communication bandwidth can be computed like voice or video communication. However, the traffic generated by data communications represented by the Internet is bursty in nature. If the required communication bandwidth is estimated conservatively and the amount of assigned bandwidth is higher than the average traffic rate, the utilization efficiency of bandwidth will be low and communication resources will be wasted. Conversely, if the required communication bandwidth is estimated without regard to traffic bursts, the situation where QoS cannot be maintained will arise frequently. Therefore, in data communications, where traffic load dynamically changes, QoS must be assured by some form of QoS control, not admission control.

2.1.1 Throughput

QoS control methods for guaranteeing throughput fall into three types. The first type guarantees constant throughput. In this type of QoS control scheme, the user is always provided with the same amount of throughput. The amount is defined when the connection is established, and the amount is kept .This type of QoS control method is used in order to accommodate constant bit rate traffic. In this case, transmission delay time is also held at a steady value. The second type of QoS control method guarantees minimum throughput. This QoS control scheme holds throughput more than the pre-defined value. The third type of QoS control method does guarantee throughput as much as a user requires. This type of QoS control method assigns as much radio resource as the user requires without exceeding the maximum throughput of the radio link.

2.1.2 Transmission delay and delay variation

Network bandwidth assignment becomes increasingly important with an increase in the kinds of applications that communicate using a network. In particular, it is necessary to understand network transmission characteristics for applications treating voice and/or video in real time. One of the most important characteristics is "transmission delay". When information is transmitted via a medium, "delay" always occurs. Delay that occurs in a communication network can be eliminated neither by a high-speed network nor highly efficient equipment. Delay time cannot be eliminated, only shortened. Network delay is defined as the period from data transmission on the transmission side to reception on the receiving side. It is caused by various factors such as latency, packet loss, and processing time.

Delay variation is a characteristic strongly related to transmission time. Transmission delay variation expresses the degree of fluctuation of transmission delay. Since even small transmission delay distribution may also cause buffer underflow on the data receiving side, it may be necessary to guarantee delay variation when accommodating traffic of a real-time system stream. In particular, delay variation in ATM is specified in detail and a severe standard is applied. Therefore, it is necessary to guarantee delay variation on radio links in wireless ATM according to the standard of the wired network.

2.1.3 Fairness

Wireless communications or wired communications also use limited resources in order to achieve data transmission. When two or more user terminals are exploiting limited resources, the amount of demand may exceed the quantity of applicable resources that can be offered. In this case, it is necessary to specify how the limited radio resources should be shared among users. One of the indicators is fairness. In data communication networks there are various kinds of fairness: fair throughput assignment, fair bandwidth distribution, fair sharing of medium access rights and so on. Various control schemes have been proposed for the type of fairness guaranteed. In fair throughput assignment, an equivalent throughput is assigned to all the users. When providing fair assignment, a center controller that controls the transmission of user terminals measures the throughput assigned to each terminal and accordingly controls future throughput assignment. In fair bandwidth distribution, instead of throughput, the bandwidth used for the communication is equally assigned to each user. In fair sharing of medium access rights, the equality of the probability by which a user acquires a medium access right is guaranteed. The index showing how much fairness is achieved is called the fairness index. A typical fairness index is the standard deviation of the quantity of the parameter assigned to each user terminal.

2.1.4 Packet error rate

In wireless communication, various factors can degrade radio link quality. For example, the quality readily falls when the received power declines due to fading, noise, or interference from other systems. This degradation increases the number of errors in the received data, as indicated by the error rate. The error rate types include bit error rate, frame error rate, and packet error rate (PER). In general, wireless links have much higher PER than wired links such as optical fiber links. Therefore, in wireless communication networks, the packet error correction function plays an important role in improving the overall communication quality. It can be said that the PER in mixed communication systems is dependent on the transmission quality in the wireless communications sections, i.e., the PER of wireless communications.

The radio link error rate in wireless communications is managed by controlling factors such as transmitted power at antennas, modulation methods, coding rates, and error correction codes. Although radio links have higher PER than cable links, it is still only about 10^{-5}. Therefore, it requires much longer time to measure PER correctly. For example, at 2 Mbps access speed, a PER measurement period of 30,000 seconds is needed to check whether the PER is less than 10^{-10} with 95% reliability. Therefore, virtually no control scheme feeds the measured values back to the transmitting side by directly measuring the wireless link error rate on the receiving side. Almost all schemes use indirect measurements in controlling the error rates. As an example, let us consider the case where the radio link error rate is managed by controlling transmitted electric power. In this case, simulations or experiments are performed beforehand to clarify the required received power on the receiving side that yields the desired bit error rate on the receiving side. The desired error rate is then indirectly attained by controlling the transmitted electric power to achieve the desired received power.

2.1.5 ATM service categories

Network-provided services in ATM are categorized into four types: constant bit rate (CBR) services, variable bit rate (VBR) services, available bit rate (ABR) services, and unspecified bit rate (UBR) services. They are classified by their respective throughput characteristics but their transmission delay performances also differ.

CBR services are applicable for connections that convey traffic at a constant bit rate, where there is an inherent reliance on time synchronization between the traffic source and destination. CBR applications are tailored for any type of data for which the end systems require predictable response time and a constant amount of bandwidth continuously available for the life-time of the connection. The amount of bandwidth is characterized by the peak transmission bit rate. These applications include services such as video conferencing, telephony (voice services). For telephony and native voice applications, CBR provides low-latency with predictable delivery characteristics,

While CBR services are for constant bit rate traffic, VBR services are provided to accommodate variable bit rate traffic. These services are applicable for traffic that is generated at variable rate and/or consists of bursty data. They are applicable for connections that convey traffic at variable rates and traffic that relies on accurate timing between the traffic source and destination. An example of traffic that requires this type of service category is variable-rate, compressed-video streams for which there is a need to try to achieve a guaranteed bandwidth or latency. In VBR applications, bandwidth is assigned to users on the basis of their demands in a range that is defined by the maximum rate. Sources that use VBR connections are expected to transmit at a rate that varies with time (for example, traffic that can be considered bursty).

The third service type, called ABR services, guarantees minimum bandwidth for data transmission and shares surplus bandwidth among the users who demand more bandwidth to transmit data. The ABR service is similar to VBR, because it also is used for connections that transport variable bit rate traffic. However, there is no reliance on time synchronization between the traffic source and destination, and for cases in which no guarantees of latency are required ABR provides a best-effort transport service, in which flow-control mechanisms may be used to adjust the amount of bandwidth available to the traffic originator. The ABR service is designed primarily for any type of traffic that is not time sensitive and expects no guarantees of service quality. This category generally is considered preferable for TCP/IP traffic, as well as for other LAN-based protocols that can modify their transmission behavior in response to the ABR’s rate-control mechanics.

The UBR service also is similar to the VBR and ABR services because it is used for connections that transport variable bit rate traffic for which there is no reliance on time synchronization between the traffic source and destination. However, unlike ABR, there are no mechanisms to dynamically adjust the amount of bandwidth available to the user. Although UBR is one of the best effort type services, it does not guarantee any throughput. Therefore, much bandwidth is assigned to users when there are available resources. However, UBR service users can’t transmit any data when other service users consume all resources for their communication. The UBR service generally is used for applications that are tolerant of delay and packet loss. It has enjoyed success in Internet LAN and WAN environments for store-and-forward traffic, such as file-transfer and e-mail. Similar to the way in which upper-layer protocols react to ABR’s traffic-control mechanisms, TCP/IP and other LAN-based traffic protocols can modify their transmission behavior in response to latency or packet loss in transport networks.

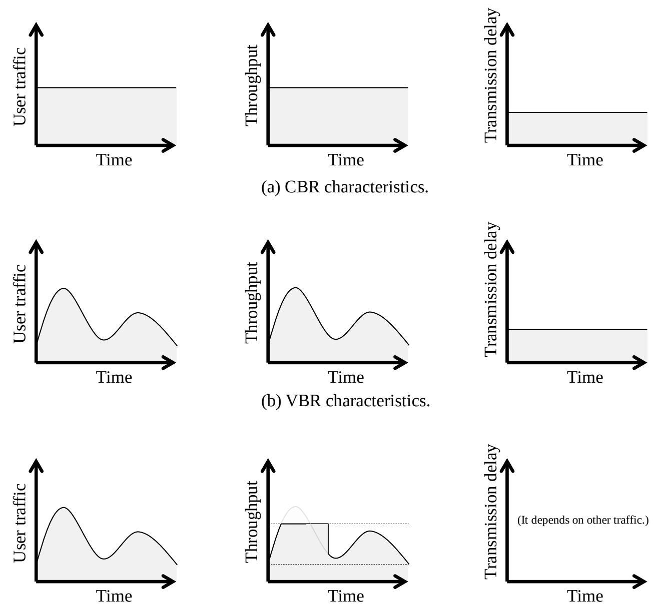


Fig. 2-1. Service characteristics of traffic, throughput, and transmission delay.

Figure 2-1 shows the characteristics of the four service types. The CBR services are used in order to accommodate traffic at a constant bit rate while providing constant throughput and constant delay. The VBR services are used in order to accommodate traffic at a variable bit rate; throughput changes according to the traffic. As with CBR services, transmission delay is constant in VBR. The ABR services are also used in order to accommodate traffic at a variable rate, but only minimum throughput is guaranteed. When the network is crowded, the throughput is restricted and delay increases are possible.

2.2 Multiple access protocols

The multiple access protocols, addressed in this section, are those used in communication systems in which the resource to be shared is the communication channel. A multiple access protocol is defined as an agreement and a set of rules among users for successful transmission of information across a common medium. In this section, multiple access protocols are classified into two types: the centralized multiple access protocols and the distributed multiple access protocols. In the centralized multiple access protocol, a coordinator such as a base station or an access point basically controls transmission timings of all wireless terminals to avoid the collision of transmission signals. The distributed multiple access protocol avoids packet collision by using a distributed access control algorithm.

2.2.1 Centralized multiple access protocols

Centralized multiple access protocols manage the transmission timing of all users and prevents users from trying to access the same radio channel at the same time by scheduling the access timing of all users. Time division multiple access (TDMA) and frequency division multiple access (FDMA) are typical protocols.

2.2.1.1 Time division multiple access

In TDMA, which is one of the centralized multiple access protocols, transmission time is divided into frames consisting of several time slots [10]-[12]. Figure 2-2 shows an example of a TDMA frame structure. In TDMA, each user is time-exclusively assigned one or more different cyclically repeating time slots. The relationship between a user and his/her assigned time slots is fixed in time and users are able to use their time slots periodically. Because time slots are periodically assigned to users and transmission is not continuous, transmission delay caused by frame structure is unavoidable. However, TDMA has an advantage that it is simple to assign multiple channels to a single user by simply assigning the user multiple time slots.

2.2.1.2 Frequency division multiple access

FDMA is one of the centralized multiple access protocols. In FDMA, the system frequency bandwidth is divided into non-overlapping frequency bands called channels. FDMA assigns individual channels to individual users. Each user is allocated a unique frequency band or channel. These channels are assigned on demand to users who request service. During the period of a call or data communication, no other user can share the same frequency band. In FDD systems, the users are assigned a channel as a pair of frequencies; one frequency is used for the forward channel, while

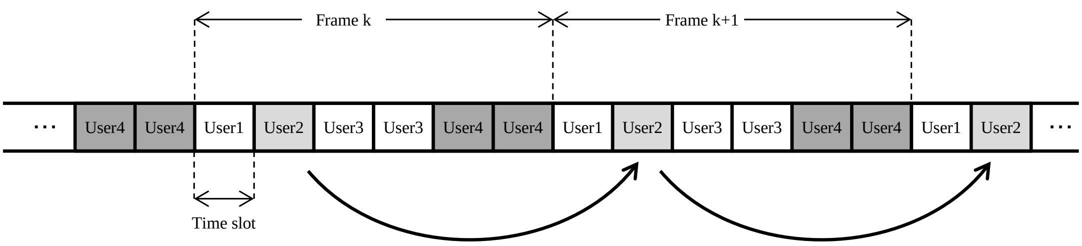


Fig. 2-2. TDMA frame structure.

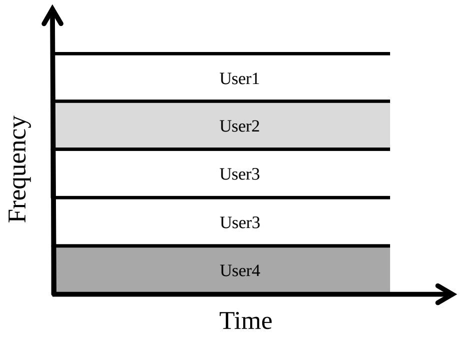


Fig. 2-3. FDMA channel allocation.

the other frequency is used for the reverse channel. Figure 2-3 shows an example of channel allocation in FDMA. Since FDMA is a continuous transmission scheme, fewer bits are needed for overhead purposes than TDMA.

2.2.1.3 Polling

Scheduling protocols avoid the situation in which two or more users access the same channel at the same time by scheduling the transmission of all users. This can be done in a demand-assigned fashion where the scheduling only takes place between the users that have something to transmit. Protocols of this type are called demand-assignment scheduling protocols. One such protocol is the polling protocol. Figure 2-4 shows the access sequence of the polling protocol. In the polling protocol, a base station (BS) asks all the wireless terminals (WTs) whether they have something to

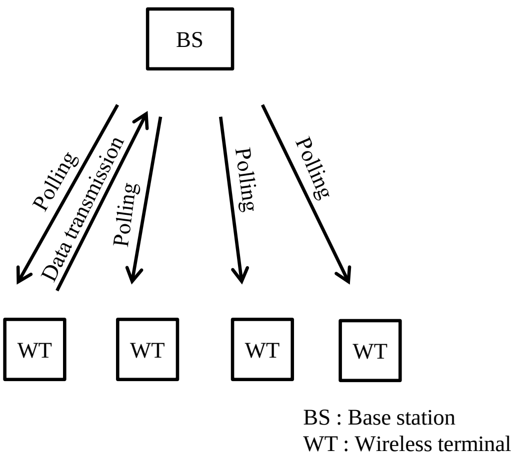


Fig. 2-4. Access sequence of polling protocol.

transmit or not. If the BS finds a WT that has data, it assigns access rights to the WT. After that, the WT transmits its packets. Thus, a polling protocol is a communication method in which a BS has the leadership and controls a WT's transmission. Therefore, even if a WT has data to transmit, it is necessary to stand by until it is polled, and consequently a time lag occurs. However, the polling protocol completely avoids WT packet collisions. The polling protocol uses the limited radio resource more efficiently than TDMA or FDMA because it assigns radio resource only to users that need to transmit data; it does not waste the resource.

2.2.2 Distributed multiple access protocols

When the target application generates continuous traffic, such as voice and video data, a dedicated channel facilitates good performance. However, most data applications do not require continuous transmission because data are generated at random times. Therefore, dedicated channel assignment can be extremely inefficient. Distributed multiple access protocols are used in such systems to efficiently assign radio resources to only active users. If packets from different users overlap in time, collisions occur. In this case, packets may be decoded unsuccessfully.

It is essential to use random access techniques when users transmit their packets in one common channel using a distributed access control algorithm. If the random access technique used has low processing efficiency, both the number of channels required for communication and transmission delay will increase, and service quality will deteriorate. Therefore, random access techniques are important for recent wireless access systems.

2.2.2.1 ALOHA

With distributed multiple access protocols there is no scheduling of transmissions. This means that a user is not able to know when he/she can transmit without interfering with the other users. Thus, if several ready users start their transmissions more or less at the same time, all of the transmissions will fail. The random access protocol should resolve the contention that occurs when several users transmit simultaneously.

In 1970, the University of Hawaii proposed the pure ALOHA system. The pure ALOHA random access protocol is simple. In pure ALOHA, users transmit data packets as soon as they are formed. If we ignore the capture effect, then packets that overlap in time are assumed to be received in error and must be retransmitted. The reason for the inefficiency of pure ALOHA is the fact that users can start their packet transmission at any time, and any partial overlap of two or more packets

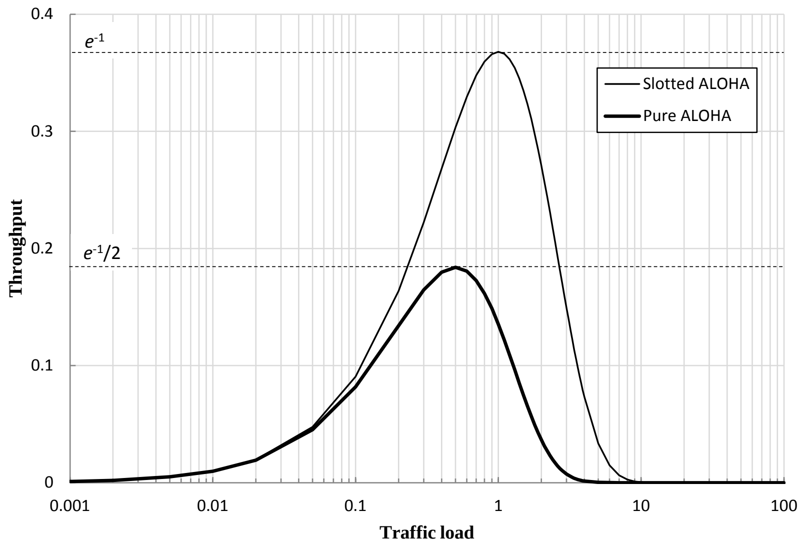


Fig. 2-5. Throughput performance of ALOHA schemes.

destroys the successful reception of all packets. If a positive acknowledgement can't be received within a predefined interval, a packet the same as the error packet should be retransmitted on the transmission side. By synchronizing users such that all packet transmissions are aligned in time, the partial overlap of packet transmission can be avoided. This is the basic premise behind slotted ALOHA [13].

2.2.2.2 Slotted ALOHA

In slotted ALOHA, time is assumed to be slotted in timeslots and users can only start their packet transmission at the beginning of the next timeslot after the packet has been formed. Thus, there is no partial overlap of transmitted packets. Therefore, the slotted ALOHA provides higher throughput than the pure ALOHA does. Note that slotted ALOHA requires synchronization of all user terminals in the wireless network, which can entail overhead. In general, the synchronization between user terminals is achieved by receiving broadcast signals from a base station. Figure 2-5 shows the throughput performance of the two ALOHA schemes. By avoiding partial collisions, slotted ALOHA achieves twice the throughput of pure ALOHA.

2.2.2.3 Carrier sense multiple access

Collisions can be reduced by carrier sense multiple access (CSMA), where terminals sense the channel and delay transmission if they detect that another user is currently transmitting [14]-[16]. In CSMA, a terminal getting ready to transmit its packets senses the radio channel before transmitting. If any other users are transmitting their packets and the radio channel is busy, then the user postpones packet transmission until the radio channel becomes idle so as to avoid collisions. After the channel changes to idle, the user who detected the channel was once busy typically waits for a random time period before transmitting. This operation is called backoff. After backoff, the user senses the radio channel. If the channel is idle, it starts to transmit its packets. For this to be effective, detection time and propagation delays in the system must be small.

Chapter 3
Quality-of-Service Control Scheme in Centralized Access Control System

This chapter proposes a new polling-based dynamic slot assignment (DSA) scheme. With the rapid progress of wireless access systems, wireless data communication will become more and more attractive. In wireless data communication, an efficient DSA scheme is required to enhance system throughput, since the capacity of radio links is often smaller than that of wired links. A polling-based DSA scheme is typically used in centralized slot assignment control systems. It, however, is difficult to assign the slots to the targeted mobile terminals in a fair-share manner if only a polling-based scheme is used, especially in unbalanced-traffic circumstances, as revealed later. To solve this problem, we propose the exponential decreasing and proportional increasing rate control as is employed in available bit rate (ABR) service in ATM so that fair slot assignment is achieved even in heavily-unbalanced-traffic circumstances. Moreover, so that an AP operating with a large number of MTs can avoid long transmission delays, a polling-based resource request scheme with random access is featured in a new algorithm. Simulations verify that the proposed scheme offers fair slot assignment for each user while maintaining high throughput and short delay performance.

3.1 Overview

The demand for high-speed-multimedia communication is increasing with the popularity of data communication services such as the Internet [17]. Packet-based technology suits data communication because it enables the bandwidth of the transmission medium to be more efficiently used. It is especially beneficial in wireless communication systems, whose bandwidths are often far less than those of metallic wire or optical fiber.

There are two approaches to radio medium access control: distributed control schemes [18][19] and centralized control schemes. In general, centralized control schemes offer better performance in terms of resource utilization efficiency than distributed control schemes, although the former are simple to implement. Also, centralized control schemes may support quality of service (QoS) in a more sophisticated manner. Accordingly, many centralized control schemes

The contents of this chapter are based on [69].

have been proposed [20]-[22] and are being discussed in several standardization organizations.

The polling-based DSA scheme is one of the simplest centralized control schemes. In a system using this scheme, an access point (AP), which controls the radio medium access of all mobile terminals (MTs) in its coverage area, polls the MTs to determine how many slots they require to transmit their packets. The AP then assigns slots to the MTs according to their requests and the resources available.

One-time transmission of the entire packet in hand (this is referred to as bulk transmission, hereafter) can be achieved if an MT is assigned all the slots it requests. Bulk transmission in the medium access control (MAC) layer yields good performance in the network layer as explained in detail later. On the other hand, when an insufficient number of slots are assigned, a protocol data unit (PDU) in the network layer may be sent using several MAC PDUs. In this case, the transmission delay of the network layer can be considerable. In other words, long delay in the network layer may happen when the input traffic exceeds the capacity of the radio link. This means that when even just one MT carries heavy traffic such that the total traffic exceeds the radio link capacity, the other MTs may suffer degraded performance such as very low throughput and long delay. This is a serious problem for systems that aim to support QoS guarantees such as minimum throughput per MT or maximum delay.

To avoid traffic congestion in the MAC layer, a traffic shaping mechanism can be employed in the network layer. When the traffic of each MT in the MAC layer is always held to under one N-th of the radio link capacity (N: the number of MTs connected to an AP), i.e. traffic shaping, traffic congestion will not happen in the MAC layer. However, when one MT outputs heavy traffic while the others output very light traffic, the heavily-loaded MT suffers insufficient bandwidth even though the capacity of the radio link is not used completely. In other words, traffic shaping in the network layer blocks realization of statistical multiplexing gain in the MAC layer. Therefore, it becomes crucial to assign an appropriate number of slots to each MT considering the traffic of each MT and total traffic load in the MAC layer.

To date, several studies [23][24] on DSA schemes have been published. They examined total system performance in evaluating their proposed schemes; however, to the best knowledge of the authors, degradation (e.g., long delay time, less throughput) in MT performance caused by other MT’s traffic (i.e., inter-MT degradation) have not thoroughly been examined for DSA-based wireless access systems. As described above, it is now essential to consider inter-MT degradation to provide services with QoS guarantees.

Given the above discussion, we propose a new slot assignment scheme. The proposed scheme observes the traffic of each MT and when the amount of all traffic exceeds the radio link capacity, and then it restricts the outputs of MTs which excessively require the large number of slots to avoid degrading the performance of other MT’s. Specifically, the traffic is controlled by newly combining the exponential decreasing and proportional increasing rate controlling mechanism in the slot assignment scheme. Furthermore, a new resource request scheme is also incorporated into the slot assignment scheme to enhance the delay performance.

In addition to slot assignment, resource request scheme can be essential, since, in DSA, MTs may request slots for data transmission, rather than having fixed amounts of data to transmit. A polling scheme can be used to handle resource requests; however, it is inefficient that each request consumes a constant amount of bandwidth even though the MT has no user data to transmit. An alternative is the random access resource request scheme. It, however, can incur rather long delays depending on MT traffic. To alleviate this problem, polling can be added to random access management [24]. Although this scheme performs well for bursty data traffic, it is difficult to handle non-burst traffic well. Accordingly, this section proposes a new resource request scheme that adaptively changes its resource request mode explicitly from the random access mode to the polling mode or vice-versa according to MT traffic.

The remainder of this section consists of 5 sections. Section 2 describes the structure of the wireless communication system. Section 3 explains the mechanism of the polling-based DSA scheme. Section 4 clarifies the problem of polling-based DSA. The operating principle of the proposed scheme is then given in Section 4. Section 5 provides a performance evaluation of the proposed scheme. Finally, Section 6 concludes this section.

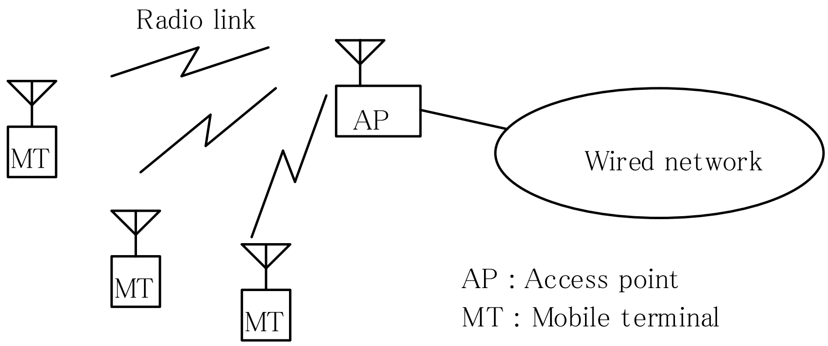


Fig. 3-1. Wireless access system.

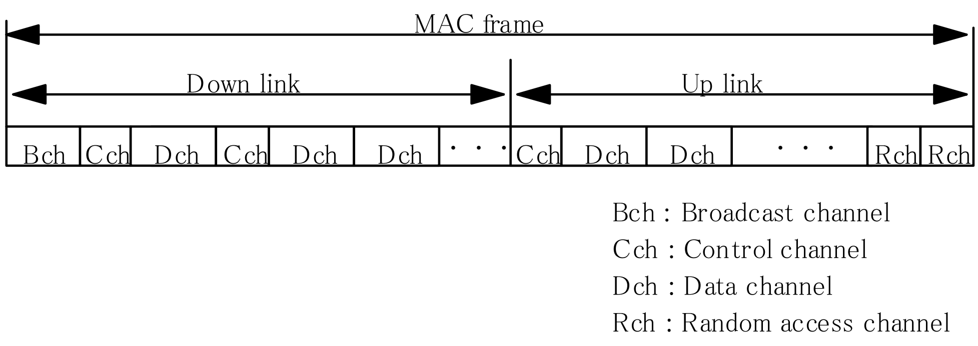


Fig. 3-2. Structure of MAC frame.

3.2 Assumed Wireless Access System

3.2.1 System Configuration

The configuration of the wireless access system considered in this section is shown in Fig. 3-1. The system is composed of one AP, which is connected to a fixed network, and several MTs. The system employs time division multiple access (TDMA) - time division duplex (TDD) in the medium access control (MAC) layer. The MAC frame structure assumed here is depicted in Fig. 3-2. The MAC frame consists of broadcast channel (Bch), control channels (Cch), data channels (Dch), and random access channels (Rch). Bchs are used to inform the attributes of the AP and MAC frame structure (i.e., the position and the number of Cchs or Dchs assigned to MTs in this MAC frame). Bch and Cch formats are shown in Fig. 3-3 and Fig. 3-4, respectively. Rchs may convey MT identifier and a resource request. Rchs are used for random access slot assignment.

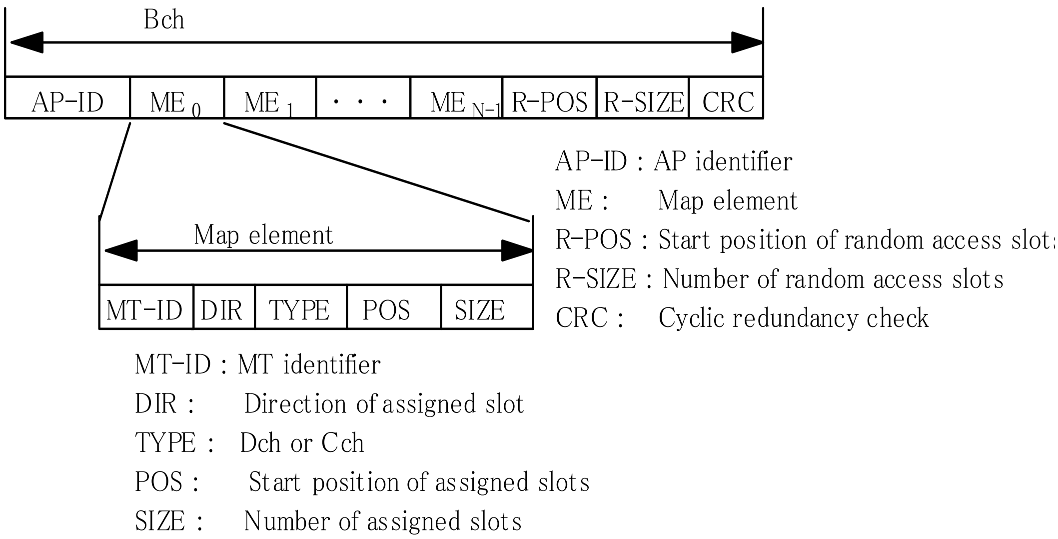


Fig. 3-3. Bch format.

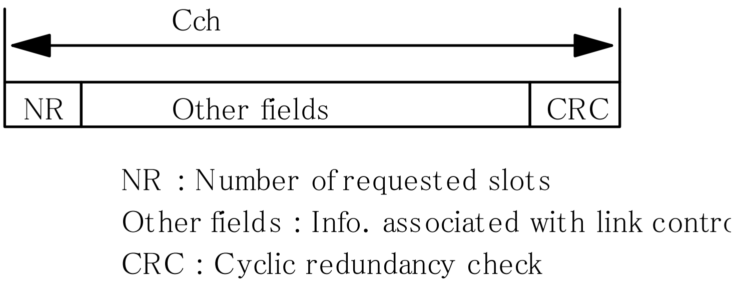


Fig. 3-4. Cch format.

3.2.2 PDU Structures

Figure 3-5 shows the relationship between the PDUs in each layer. On the transmitting side, the network layer PDU is segmented into several blocks of the same size as the DLC service date unit (SDU) in the segmentation and reassemble (SAR) layer. A DLC-PDU consists of a DLC header, DLC-SDU, and trailer. The DLC header part contains a sequence number, which is used to achieve ARQ, SAR flag, which indicates the last DLC-PDU in the same network layer PDU, and so on. The trailer is a cyclic redundancy check (CRC) unit. In the MAC layer, the DLC-PDUs are first clustered according to the number of assigned slots and then sent.

On the receiving side, each received DLC-PDU is inspected by CRC to determine if it has been corrupted and sent to the SAR layer if it is correct. When the SAR layer receives the last block of a network layer PDU, it reassembles the network layer PDU and sends it to the network layer.

3.3 Considered Dynamic Slot Assignment

3.3.1 User Data Transmission Sequence in Polling-based Scheme

The slot assignment and data transmission sequence is shown in Fig. 3-6. First, the AP sends a Bch at the top of a MAC frame to indicate the position at which an MT can transmit a Cch. Every MT receives the Bch and knows when to transmit their Cchs. Second, an MT transmits a Cch containing a slot assignment request. The AP knows of the number of the slots that the MT requires upon receiving the Cch. The AP then assigns usable slots to the associated MT. When there are not enough slots available in the current frame, the AP continues to assign slots in a

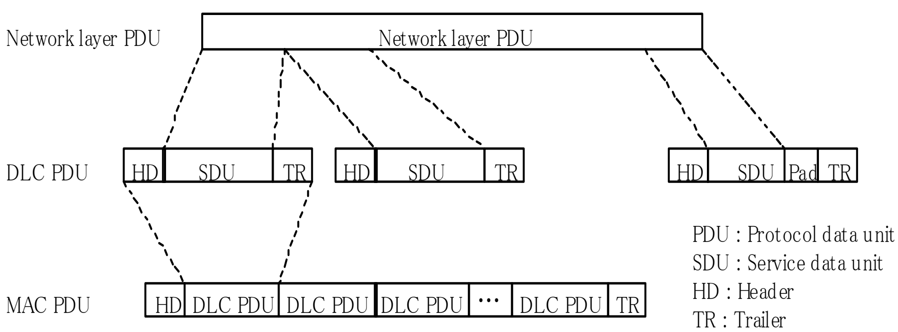


Fig. 3-5. Relationship of PDUs.

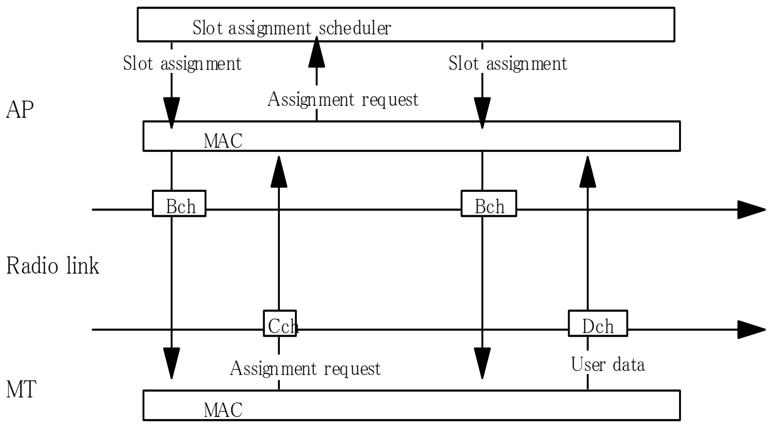


Fig. 3-6. User data transmission sequence in polling-based scheme.

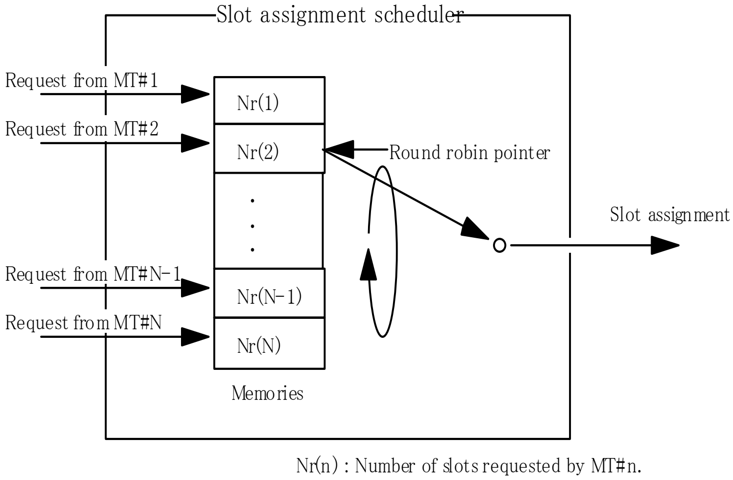


Fig. 3-7. Basic structure of SAS with conventional scheme.

subsequent frame or frames. After completing the slot assignment sequence for the MT, the AP assigns slots to the next MT in a round-robin manner.

3.3.2 Slot Assignment Scheduler in Polling-based Scheme

The AP has a function that schedules slot assignment according to the requests of MTs. This function is referred to as the slot assignment scheduler (SAS) hereafter. Figure 3-7 shows the structure of the SAS with polling-based DSA. The SAS has memories to store the number of slots requested by each MT. When a Cch is received by the AP, the number of requested slots is written to the memories. The SAS reads the memories in a round-robin manner and then assigns the slots. An example of a slot assignment pattern is shown in Fig. 3-8. First, the AP assigns slots for the up-link Cch. Second are the slots for Dchs to transmit user data. Finally, the slots for the down-link Cch are assigned in the next MAC frame. When the number of requested slots is zero, no slot is assigned for user data transmission, but the up-link Cch slot is always assigned to enable subsequent transmission requests.

The DSA scheme considered herein assigns slots to the associated MTs efficiently: especially, when the traffic is bursty, some statistical multiplexing gain can be expected. Transmitting a large number of slots of a certain MT at a time obviously contributes to higher system throughput efficiency, since the number of slots assigned for Cchs at a time is constant.

3.3.3 User Data Transmission Sequence in Random Access Scheme

Figure 3-9 shows the user data transmission sequence in the random access scheme. In this scheme, the AP sends a Bch at the top of a MAC frame to indicate the start position of random access slots and the number of the slots. When an MT has user data to transmit, it transmits an Rch containing the MT identifier and a resource request. When the AP receives the Rch, it assigns slots to transmit MT's user data in the same manner as the polling-based scheme. If the Rch collides with other Rchs that are transmitted by other MTs, the Rch is retransmitted. In this section, the random access scheme employs the exponential back-off algorithm [25][26] to avoid repeated collisions. The Rch retransmission sequence is depicted in Fig. 3-10. When an MT detects random access failure, it calculates the number of back-off slots using a back-off window to avoid subsequent collisions. It then retransmits the Rch after the back-off period. If the retransmitted Rch collides again, the MT doubles the back-off window size and recalculates the number of back-off slots. In this section, the initial back-off window size is four and the maximum window size is one thousand and twenty four.

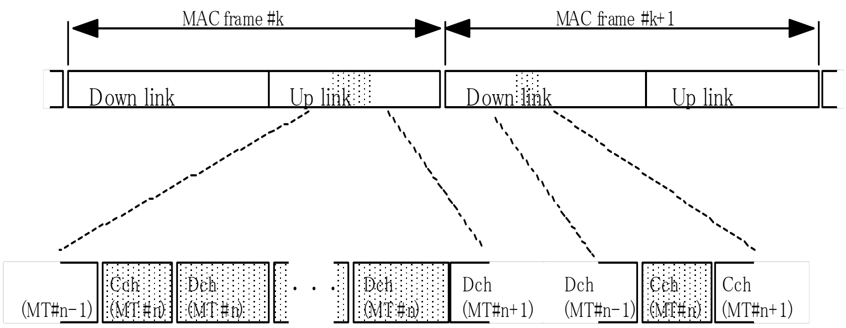


Fig. 3-8. Slot assignment pattern.

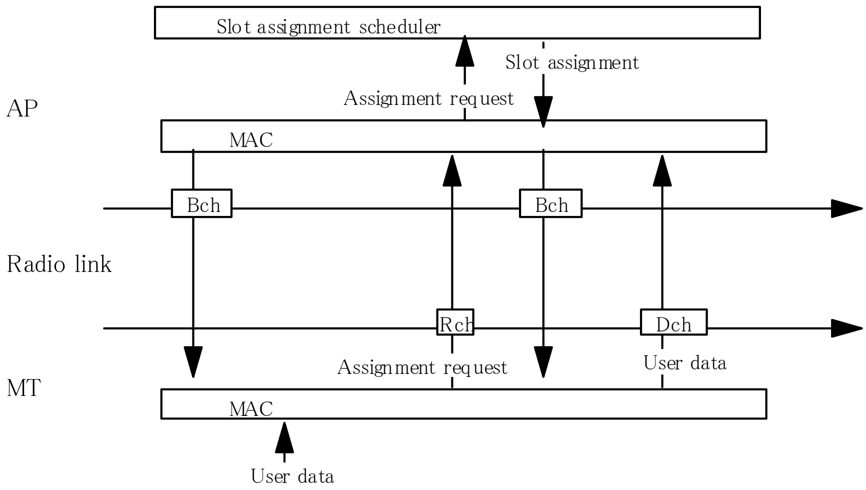


Fig. 3-9. User data transmission sequence in random access scheme.

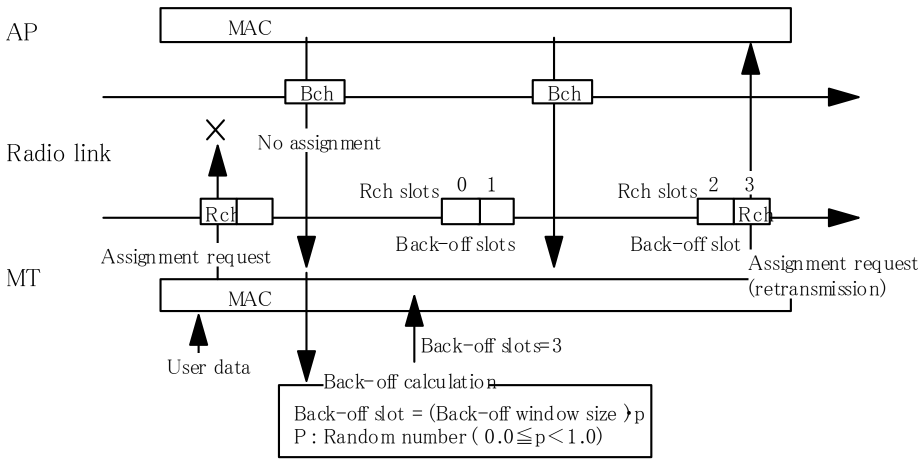


Fig. 3-10. Rch retransmission sequence (Back-off slots = 3).

3.4 Technical Challenges and Proposed Solutions

3.4.1 Technical Challenges

A. Fair Share Issue

The bulk transmission offered by the DSA considered in the previous section offers a crucial benefit: using contiguous MAC slots to accommodate a single network layer PDU minimizes the delay in the network layer. In particular, this aspect becomes essential for supporting TCP/IP-based applications, since long delays in the radio link can trigger the TCP's slow-start mechanism, which can totally degrade end-to-end throughput performance [27].

There is, however, one problem to be solved. When the scheduler blindly assigns all slots demanded by the MTs and the traffic of a certain MT is significantly heavy, it consumes the whole bandwidth of the radio link and the other MTs must wait for their turn to come around again. This obviously degrades the delay and throughput performance of the other MTs. Traffic shaping, which sets a certain limit to the allowed maximum throughput for each MT, is one solution. Namely, the number of slots that can be assigned to any MT at a time is restricted. This enables short round-robin intervals, even when an MT requests a large number of slots, allowing the other MTs to have the opportunity to transmit quickly. In this case, however, it is hard for the MTs to fully utilize the entire available bandwidth. For instance, when one MT outputs heavy traffic while the others output very light traffic, the heavy traffic MT can not transmit its all data at once because of the limit placed on the number of slots even if some slots remain unused in the MAC frame. Moreover, as described above, when PDU size in the network layer is large, the short-round-robin-interval approach can cause long delays in the network layer.

B. Transmission Delay Issue

In addition to the fair share issue described in the section above, transmission delay of the polling-based DSA scheme should be taken into consideration. In the polling-based DSA scheme, an AP periodically assigns a slot to MTs to send a slot assignment request. That is, a constant amount of bandwidth in the radio link capacity is reserved to handle the requests. The bandwidth thus used rarely affects the transmission performance (e.g., throughput, delay) when the number of connected MTs is small, since the consumed bandwidth is negligible compared to that available for data transmission. However, when a large number of MTs are connected to an AP, the transmission performance can be significantly degraded. In particular, long delay can occur if most traffic is bursty, even when the total traffic of the MTs is much less than the radio link capacity.

To avoid this degradation, a resource request scheme using random access has been proposed [28]. MTs request slots to send user data using random access slots only when user data are generated in the network layer. Therefore, the bandwidth is not assigned constantly so the requests do not affect transmission performance even when a large number of MTs are connected. However, this scheme has one essential problem in that sending a request via random access takes more time and this causes transmission delay. Especially when MTs generate user data very frequently, the random access time becomes very long, since MTs send slot assignment requests repeatedly causing collisions and retransmission of the requests.

Overall, random access is efficient when MT traffic is very bursty, but it can cause very long delays if the traffic is not bursty.

3.4.2 Proposed Scheme

A. Transmission Rate Control

To resolve the fairness issue, we propose a new DSA scheme. Basically, it identifies the MTs carrying heavy traffic, if any, and calculates the number of slots that can be assigned to each MT.

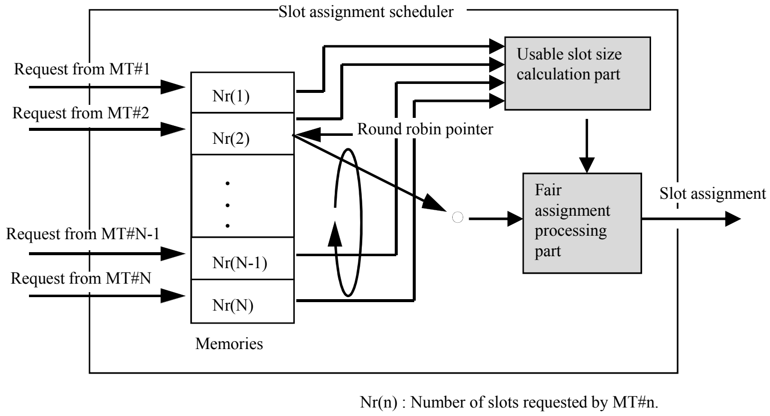


Fig. 3-11. Structure of SAS with proposed scheme.

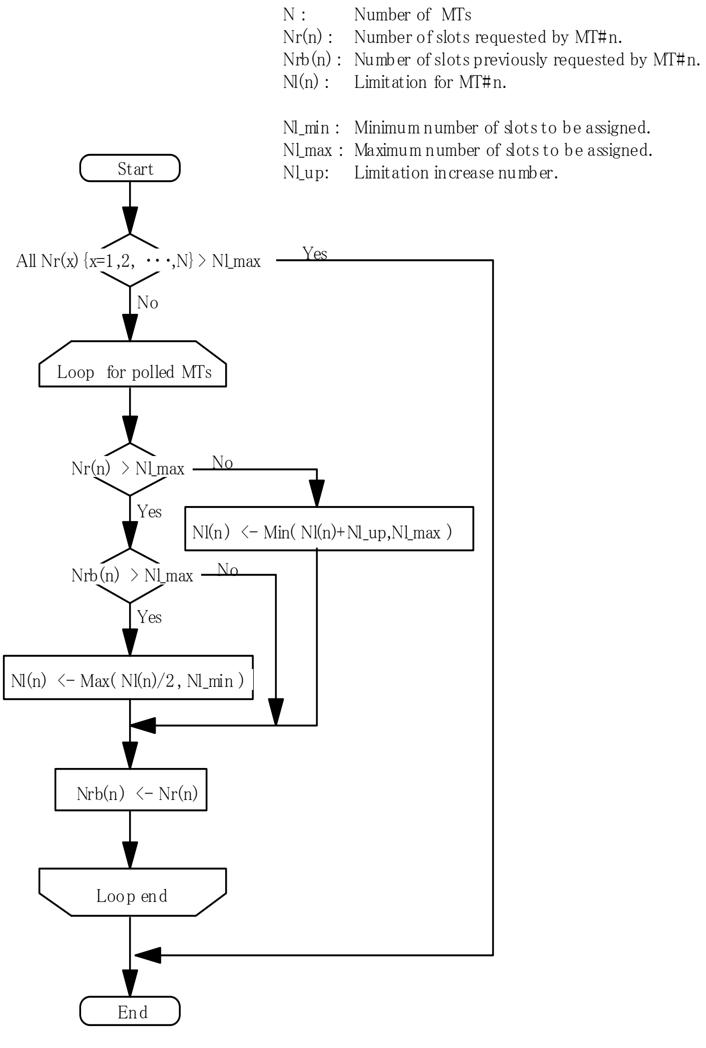


Fig. 3-12. Flow chart of usable slot size calculation.

An essential technical point of the proposed scheme is how to judge when an MT starts issuing heavy traffic and when it ends. The proposed SAS structure is shown in Fig. 3-11; the newly added usable slot size calculation and fair assignment processing parts are shown. The usable slot size calculation part calculates the number of slots assigned when an MT causes radio link congestion. On the other hand, it releases the limitation when the traffic of the corresponding MT eases. The fair

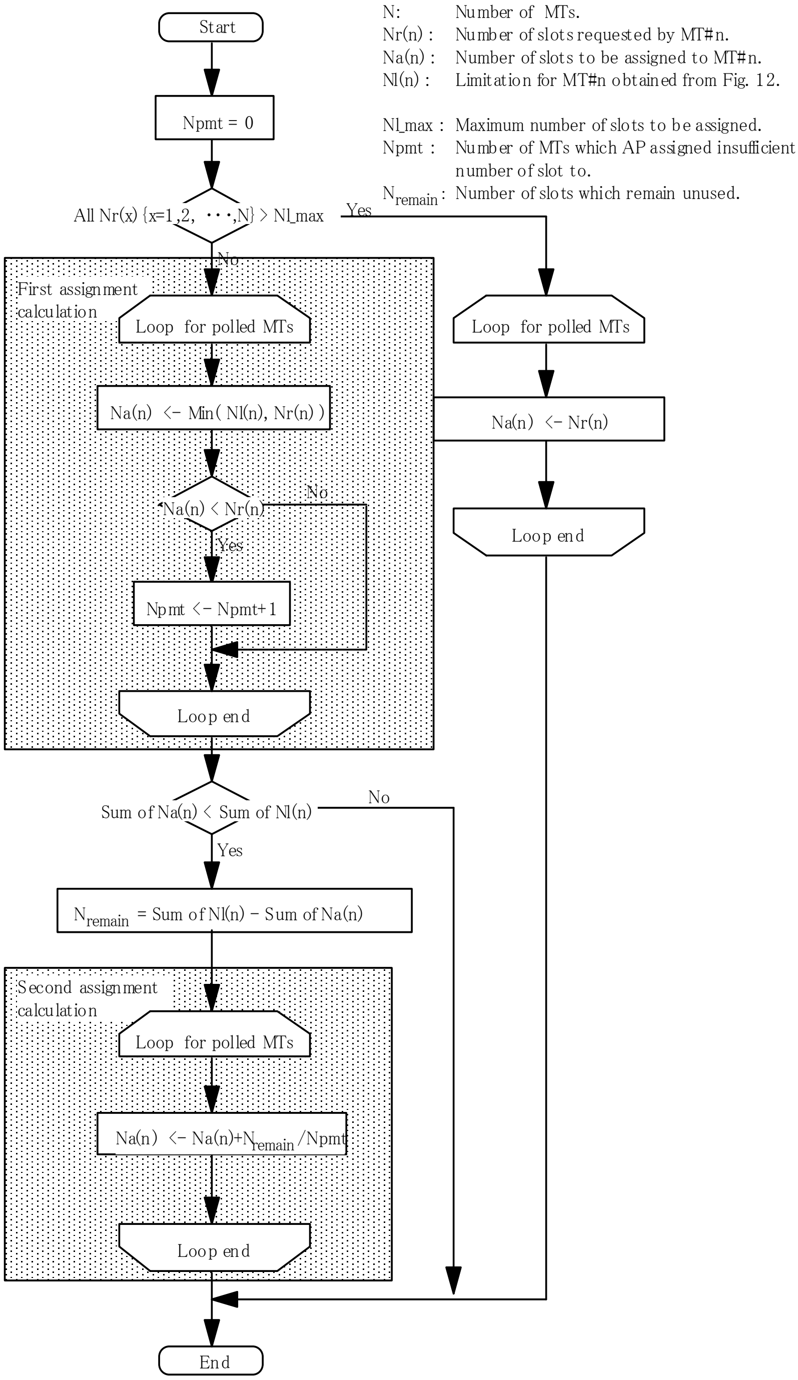


Fig. 3-13. Flow chart of fair slot assignment.

assignment processing part assigns slots to MTs applying the usable slot size for each MT.s

Figure 3-12 shows the flow chart to show how the usable-slot-size calculation part determines the usable slot size for each MT. Its basic mechanism is similar to the algorithm of ABR in ATM [29]. In the first phase, there is no limitation on the usable slot size. When the AP receives a Cch containing a slot request that exceeds the maximum assignment slot number, Nl_max, which is pre-defined as the limitation of possibly assigned slots per MT in the usable-slot-size calculation part, the corresponding-Cch-sending MT is regarded as a heavy traffic source. In the heavy traffic case, the proposed SAS then changes, Nl, which represents the current limitation number of slots to be assigned to the MT, to $Nl/2$ for the next processing. The AP continues this operation whenever it receives a Cch that contains a request exceeding Nl_max. On the other hand, if the next Cch from the heavy traffic source MT contains a request for less than Nl_max slots, the AP regards the MT as a light traffic source and changes Nl to $Nl+Nl\_up$.

In this mechanism, the usable slot size is exponentially decreased when an MT tries to emit heavy traffic. The usable slot size is proportionally increased when the MT emits light traffic. It is possible to reduce the usable slot size rapidly when congestion is detected, and to ensure limit control system stability, because of the proportional increase. Figure 3-13 shows the flow chart used to calculate the number of slots to be assigned, where the usable slot size, Nl, is obtained by the algorithm shown in Fig. 3-12. First, the SAS judges whether all MTs have requested slots exceeding Nl_max. If all MTs are in a threshold-exceeding status to the same degree, the SAS assigns the slots to MTs according to their requests without setting any slot limitation. This maximizes system throughput with fair sharing. When at least one MT is not a heavy traffic source (i.e., the requested slot size is less than Nl_max), the fair assignment approach is used. In this situation, after comparing the number of the requested slots against the usable slot size, the smaller value of the two is used to assign slots in the first assignment calculation. After the first assignment calculation is completed for all active MTs, the number of remaining available slots is calculated. The number, N_{remain} is obtained as

$$N_{remain} = \text{Sum of } Nl(n) - \text{Sum of } Na(n)\ , \tag{3.1}$$

where Sum of $Nl(n)$ is the number of slots prepared to use, and the sum of $Na(n)$ is the number of slots assigned in the first assignment calculation. If slots remain unused, the second assignment calculation is carried out. In this case, the remaining slots are assigned to the MTs requiring more slots. This two-stage slot assignment scheme efficiently assigns the slots unused after the first assignment, while supporting fair assignment to each MT.

B. Slot Assignment Requests

To solve the delay time problem, we propose a new resource request scheme. In this scheme, an MT has two requesting modes; the random access mode and polling mode, and an AP assigns slots only to MTs that are in the polling mode.

First, an MT is in the random access mode. The mode transition sequence from the random

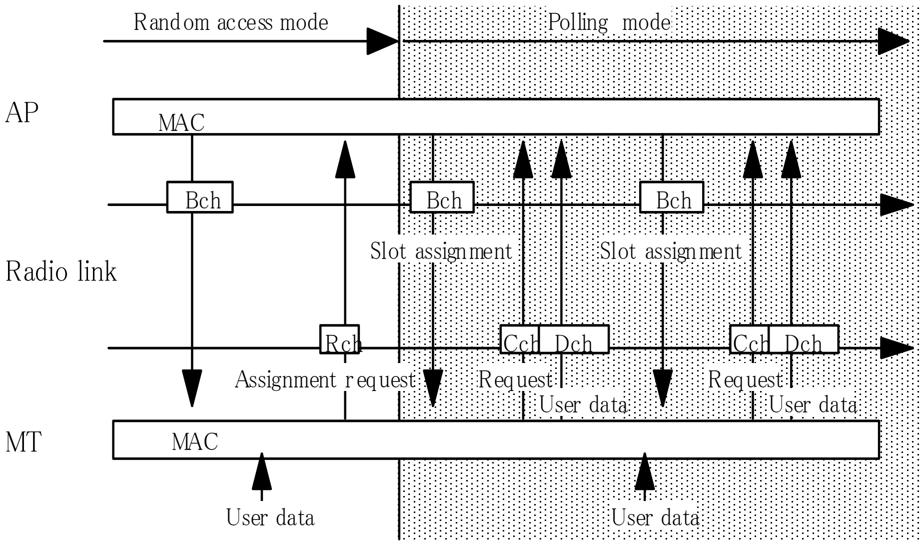


Fig. 3-14. Mode transition sequence from random access mode to polling mode.

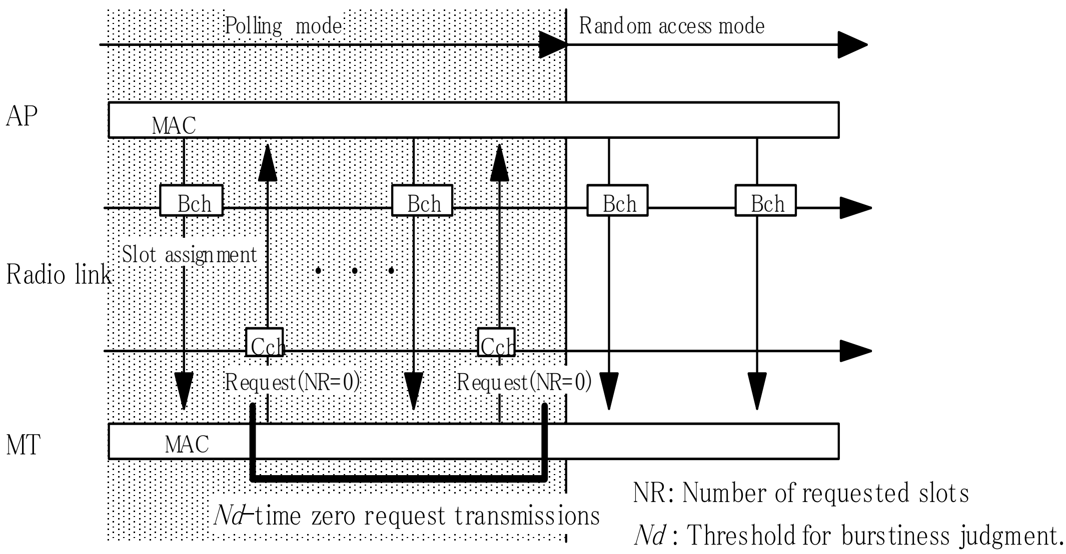


Fig. 3-15. Mode transition sequence from polling mode to random access mode.

access mode to the polling mode is shown in Fig. 3-14. In the random access mode, the MT sends slot assignment requests using random access when it has user data to transmit. After random access has succeeded, the MT switches to the polling mode. An AP can judge which MT is in the polling mode from the success of random access. The AP periodically assigns the slots in a round-robin manner to the MTs in the polling mode. MTs send user data and slot assignment requests using the assigned slots. An MT changes to the random access mode after sending subsequent *Nd*-slots with "no more slot" indication as shown in Fig. 3-15. The AP then assigns no slots to the MT. In short, *Nd* is referenced to as burstiness threshold.

When traffic is not bursty (i.e., MTs send small amounts of user data but do so frequently), the MTs switch to the polling mode and send their slot assignment requests using the slots assigned by the AP. Therefore, transmission delay equals that of polling-based DSA. When traffic is bursty (i.e., MTs send large amounts of user data infrequently), MT mode changes to the random access mode and the radio link capacity is not consumed by resource requests. In this case, the resulting transmission delay is kept low compared to that of the polling-based scheme.

Table 3-1. Main parameters of simulation model.

Parameter		Value
The number of MTs		5
DLC T buffer size		512 PDUs
MAC frame size		9000 bytes
Bch slot size		360 bytes
DLC SDU size		48 bytes
DLC PDU size		54 bytes
DLC control message size		6 bytes
Bit error rate		0.0
Maximum assignment slot number *Nl_max*		256
Traffic of MTs in network layer		
	Distribution of PDU length	Poisson
	Maximum PDU length	9600 bytes
	Distribution of occurrence	Poisson

3.5 Performance Evaluation

3.5.1 Evaluation of Fair Share Performance

Fundamental performance of the proposed scheme is examined by simulation. Specifically, the simulation focuses on the up-link, since the AP knows the down-link traffic and can easily assign slots accordingly. The main simulation parameters are listed in Table 3-1. The simulation model consists of an AP and five MTs. The sizes of DLC-SDU, DLC-PDU, and DLC control message follow those of the working assumption in the ongoing standardization activities of ETSI-BRAN in Europe and MMAC in Japan. In all simulations, the maximum assignment slot number of the proposed scheme, *Nl_max*, which is used to judge whether an MT is a heavy traffic source or not, is set to two hundred fifty five, because the *Nl_max* of two hundred fifty five is sufficient to prevent

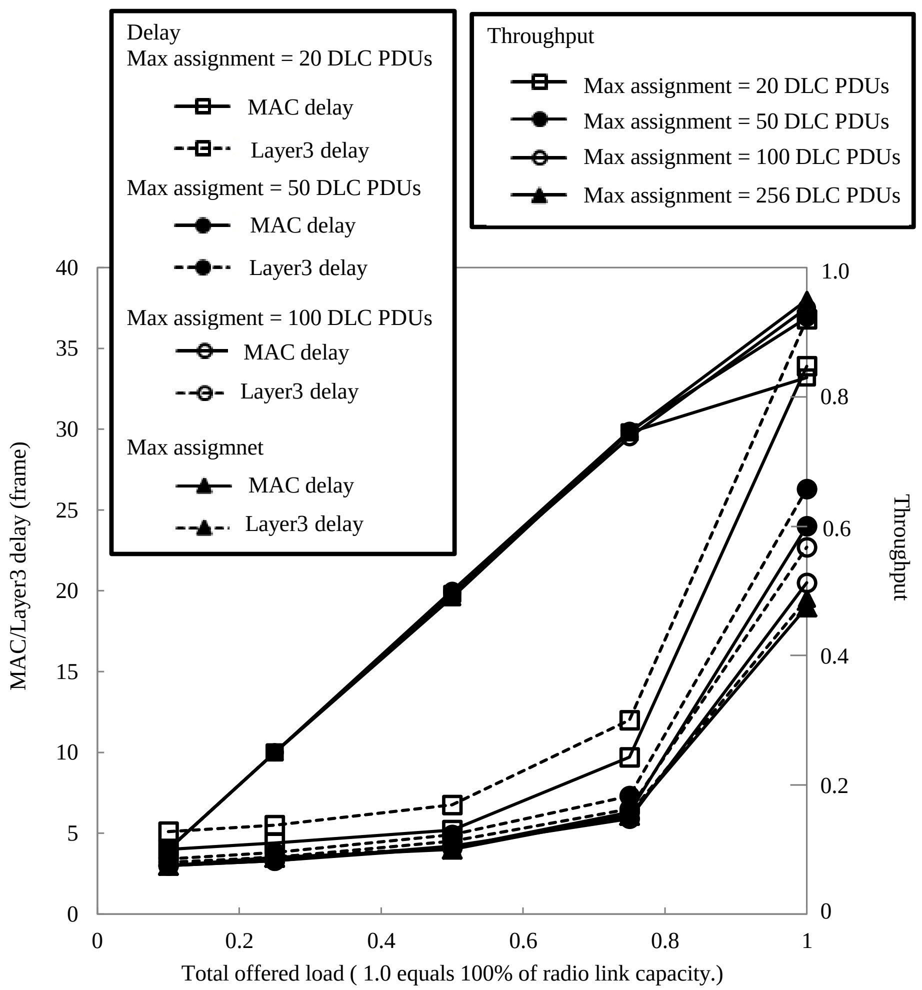


Fig. 3-16. MAC/Layer3 delay and throughput when all MTs have same traffic (Conventional, Case1).

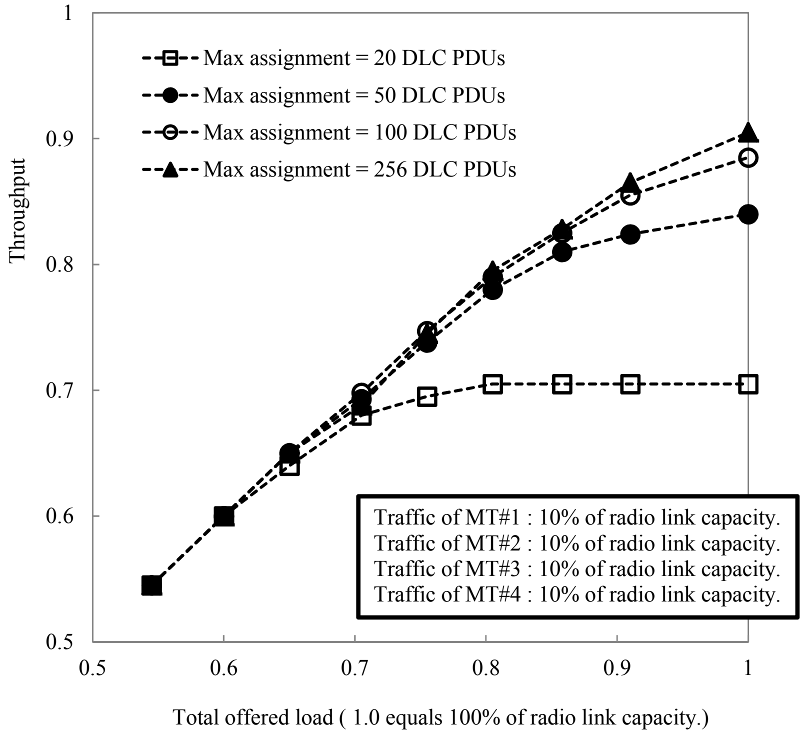


Fig. 3-17. System throughput when only traffic of MT#0 is changed (Conventional, Case 2).

light-traffic MTs from incurring their degraded throughput, as revealed later.

MT traffic is defined using a probability distribution of PDU lengths and PDU occurrence intervals in the network layer on MT side; both follow a Poisson distribution. This section examines two cases. In Case 1, all MTs output traffic that has the same statistical characteristic. In Case 2, the traffic of one MT changes while the other MTs output the same constant traffic. Note that the simulation parameters and models reveal the potential performance of the proposed scheme through a comparative evaluation.

A. Fair Share Performance of Conventional Scheme

First, for Case 1, the delay and throughput performances of the conventional scheme that constantly uses the given maximum assignment size for slot assignment are calculated. The results are depicted in Fig. 3-16. As the maximum assignment size increases, the throughput increases while the delay falls. When the maximum assignment is large, the difference between the delay in the network layer and that in the MAC layer is nearly zero. This indicates that the PDUs of the network layer are sent as one MAC PDU. It can be said that the delay and throughput performances improve as the maximum assignment size increases.

The delay and throughput performance for Case 2 is shown in Figs 3-17, 3-18, and 3-19, where MT#0 represents the heavy traffic source and the other MTs (MT#1-4) use no more than 10% of the radio link capacity. Figure 3-17 reveals that as the maximum assignment size increases, the system throughput becomes high. Figure 3-18 indicates that the delay of the light traffic sources (MT#1-4) is more severely affected by the heavy traffic source (MT#0) when the maximum assignment size is large. On the other hand, when the maximum assignment size is small, the influence from the heavy traffic source is not significant, but the MAC and network layer delays are large as shown in Fig.3-19. From Figs 3-16 and 3-17, a larger maximum assignment size is seen to be preferable. Thus, it can be said that the technical challenge is to suppress the effect of the heavy traffic source on the light traffic sources in Case-2-like circumstances, without degrading throughput and delay performance.

B. Fair Share Performance of Proposed Scheme

Figure 3-20 shows the throughput and delay performance of the proposed scheme for Case 1. Compared to Fig. 3-16, the proposed scheme and conventional scheme offer the same performance when the maximum segment size is 256. This verifies that the proposed scheme is effective when

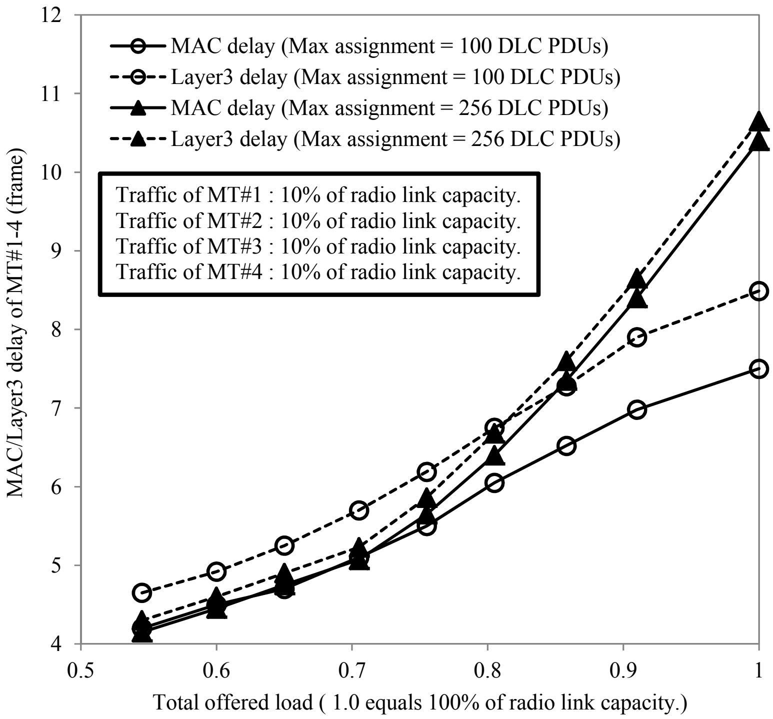


Fig. 3-18. MAC/Layer3 delay of MT#1-4 when only traffic of MT#0 is changed (Conventional, Case 2).

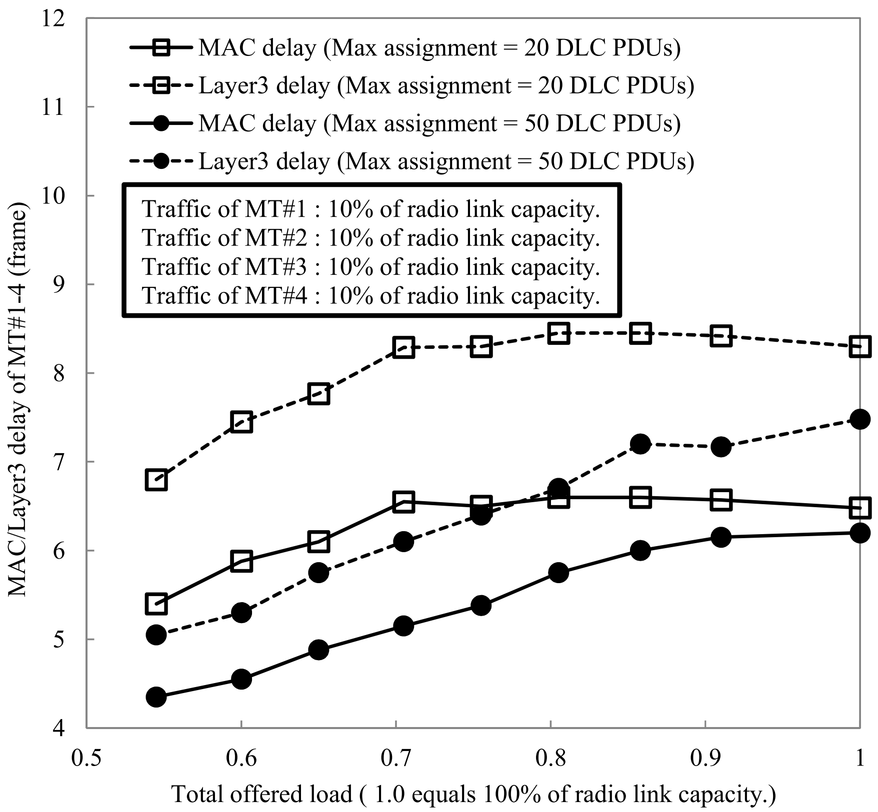


Fig. 3-19. MAC/Layer3 delay of MT#1-4 when only traffic of MT#0 is changed (Conventional, Case 2).

all MTs output the same amount of traffic as well.

The throughput and delay performance of the proposed scheme for Case 2 is shown in Fig. 3-21. This figure reveals that the throughput performance of the proposed scheme nearly equals that of the conventional scheme with the maximum assignment size of 100. As for the delay performance, when the total traffic increases, only the delay of the heavy traffic source increases while the delays of the other MTs remain quite small, unlike the conventional scheme. Comparing Fig. 3-21 with Figs 3-18 and 3-19 reveals that the proposed scheme enables the heavy traffic source to much less affects the others. Also, the proposed scheme achieves smaller delay in the network layer than that of the conventional scheme.

These results indicate that the proposed scheme performs efficiently in different traffic environments: the number-of-slots-limiting mechanism is not applied in Case 1 providing high throughput performance when MTs output the same traffic, and it is activated in Case 2 where there is a heavy-traffic MT. It should be emphasized that, when the number of MTs is large, the conventional scheme suffers more severely from the above-revealed negative influence of the heavy-traffic MT. The same goes for the case wherein many MTs are heavy-traffic sources.

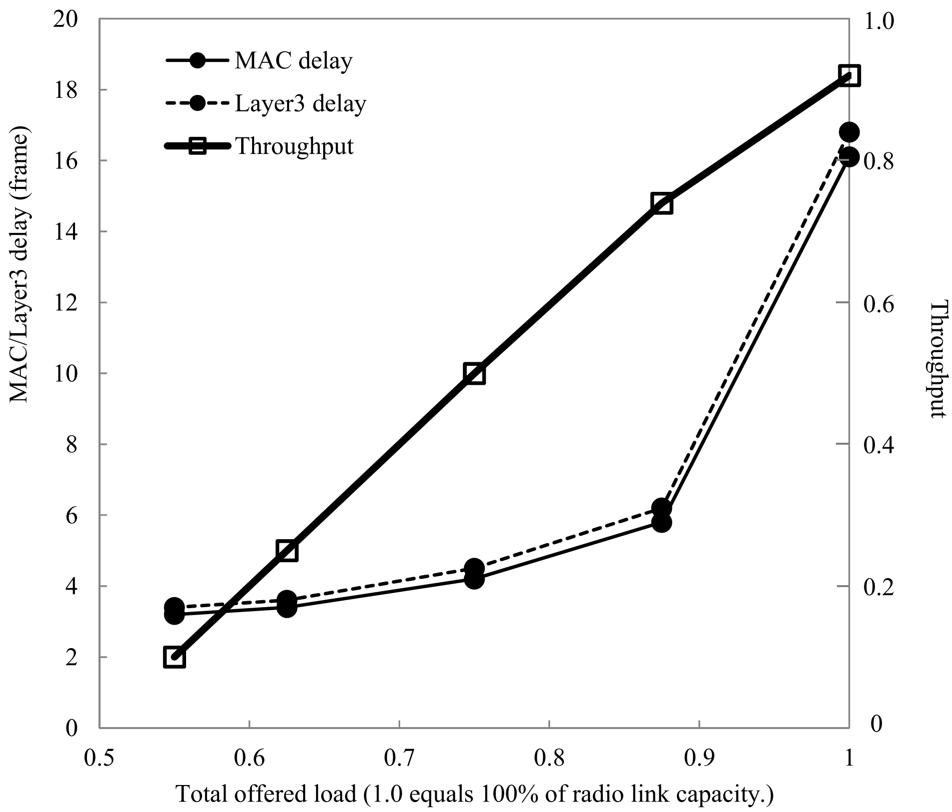


Fig. 3-20. MAC/Layer3 delay and throughput when all MTs have same traffic (Proposed, Case1).

Figure 3-22 shows the throughput versus delay of the conventional and proposed schemes in Case 2. These results reveal that the conventional scheme with maximum assignment size of 20 prevents the delay of light traffic MTs (MT#1-4) from increasing when one MT outputs heavy traffic. However, the throughput is quite limited. On the other hand, the conventional scheme with maximum assignment size of 256 can make the delay of all MTs long if the throughput is high; the scheme does not limit the throughput. The proposed scheme keeps the delay of light traffic MTs low and does not restrict the throughput.

3.5.2 Evaluation of Delay Time with many MTs

The delay of the polling-based DSA scheme with the proposed resource request is examined by simulation. The simulation model consists of an AP and one hundred and twenty eight MTs. The other parameters are the same as in the previous simulation. In this simulation, the minimum number of random access slots is constantly assigned in each MAC frame. If slots remain unused, these slots are used as random access slots. Constant assignment of random access slots degrades

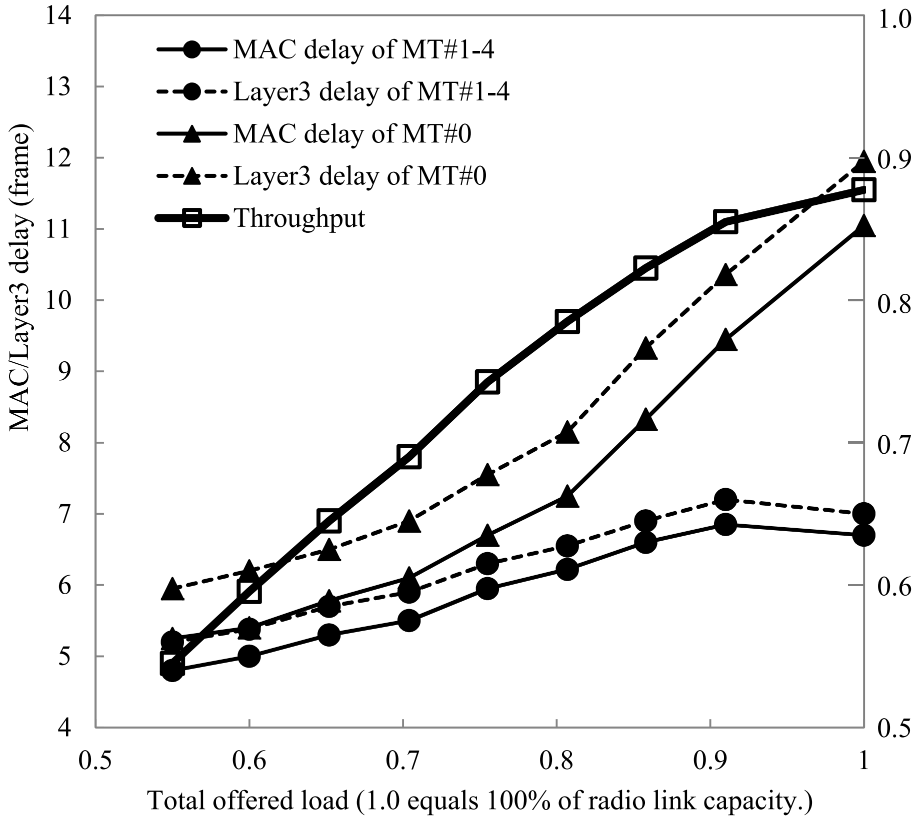


Fig. 3-21. MAC/Layer3 delay and throughput when only traffic of MT#0 is changed (Proposed, Case2).

the total throughput, because the slots may be used. Therefore, the throughput is improved by using few constantly assigned slots. However, at least one slot should be assigned to transmit resource requests even when user data traffic is heavy and no slot remains unused. Therefore, in these simulations, the minimum number of random access slots was one. The proposed scheme employs the exponential back-off algorithm to avoid successive collisions in the random access mode.

Delay versus traffic burstiness curves of the three resource request schemes, the random access, polling-based and proposed scheme, are shown in Fig. 3-23, where the total offered MT loads are 20 percent and 50 percent, respectively. In these figures, the occurrence interval of the network layer PDU is used as the measure of burstiness: the burstiness increases with the occurrence interval when the transmission rate is constant. In the proposed scheme, an MT switches to the random access mode from the polling mode after sending Nd sequential requests indicating zero slots. The simulation results are shown for Nd values of 20, 40 and 60.

Figure 3-23 shows that delay of the polling-based resource request scheme is small when the burstiness of MT traffic is low. This is because the assigned request slots are used efficiently since

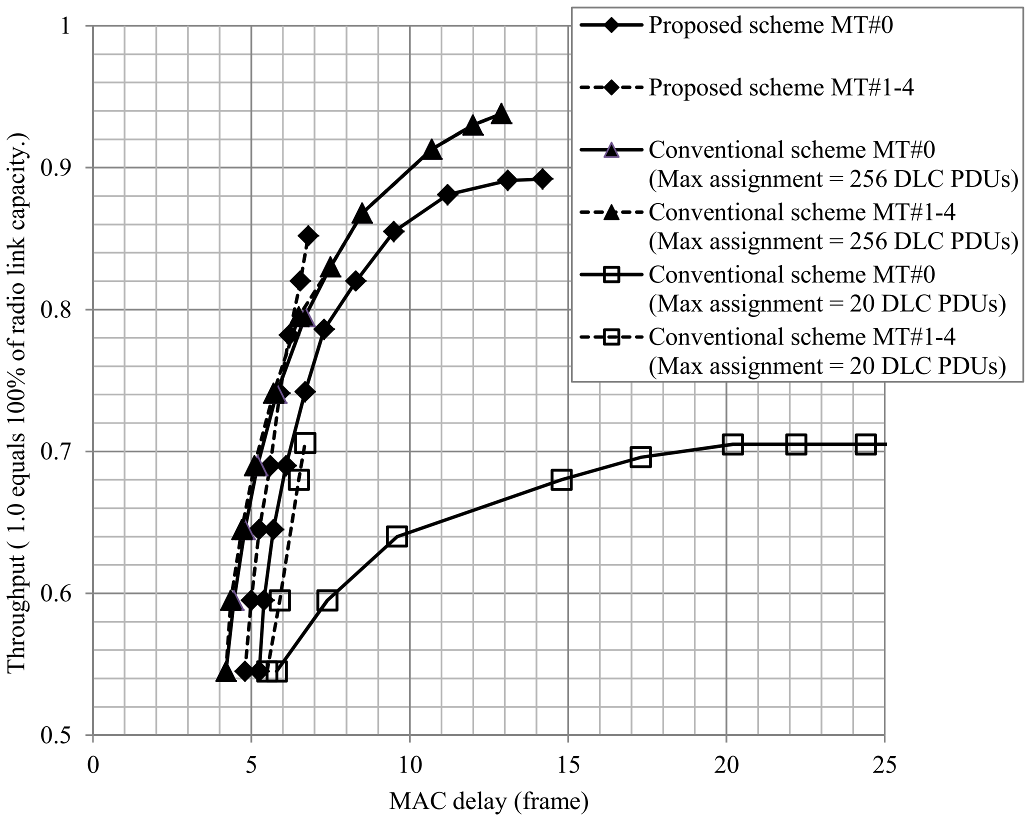


Fig. 3-22. Throughput vs. MAC delay when only traffic of MT#0 is changed (Conventional and proposed, Case 2).

MTs usually have user data to send. When the burstiness becomes high, the delay increases remarkably. This is because the slots assigned to send requests are not used efficiently because many MTs have user data to send only sporadically. On the other hand, the delay of the random access scheme is longer than that of the polling-based scheme when the burstiness is low due to collisions. However, the delay is kept lower than that of the polling-based scheme when the burstiness is high. The reason is that slots are not periodically assigned to MTs that do not have user data.

When the proposed scheme is applied, the polling mode is correctly selected when burstiness is low; the simulation results of the proposed scheme match with those of the polling-based scheme. On the other hand, as the burstiness increases, the resource requests are sent by random access. Consequently, the delay is as short as is offered the random access scheme. As the total offered load becomes high, the superiority of the proposed scheme, low-delay, strengthens.

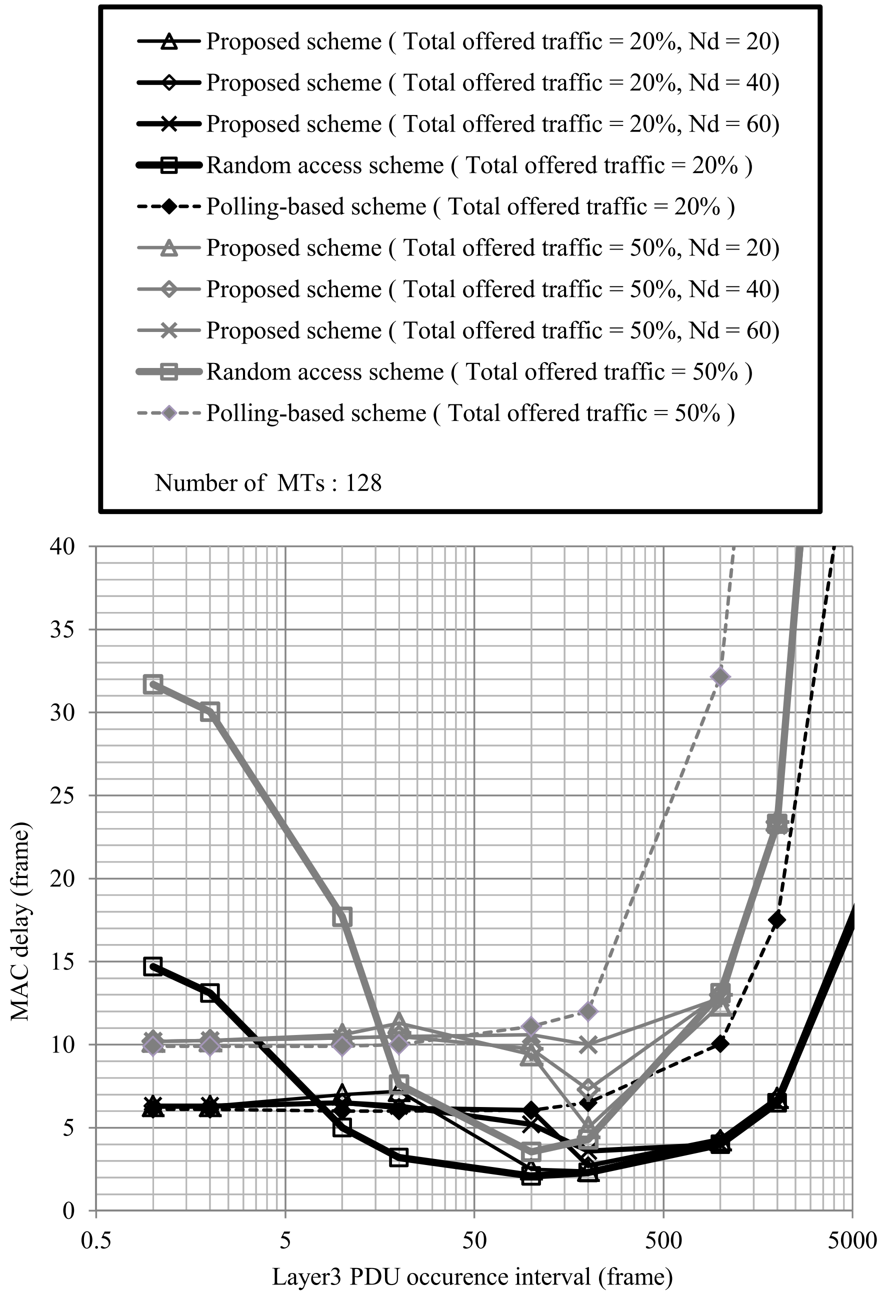


Fig. 3-23. MAC delay vs. burstiness (Conventional and proposed, Case 1).

3.6 Conclusions

This section proposed a new polling-based DSA scheme for the broadband wireless access systems that allow the AP to assign slots. It was revealed that the conventional constant length assignment scheme can have difficulty in achieving satisfactory throughput performance and short delay performance at the same time, especially when the traffic of one or more MTs is excessive. Unlike to the conventional scheme, the proposed scheme exhibits excellent performance; it offers fair slot assignment while maintaining high throughput and short delay performance even in unbalanced-traffic circumstances, mainly by employing the exponential decreasing and proportional increasing rate control scheme. We also proposed a new resource request scheme that combines polling and random access and clarified its performance. The proposed scheme uses polling access when traffic is not bursty and random access otherwise. Consequently, it minimizes the transmission delay by matching the MT's traffic characteristics.

Combining the proposed slot assignment scheme and the resource request scheme allows us to use the given radio bandwidth efficiently while guaranteeing low delay with robustness against other MT's traffic. Since, in practice, some MTs may yield significantly large traffic compared to the other MTs, the proposed scheme offers significant benefit for wireless data access systems that aim to provide stable services.

Chapter 4
Quality-of-Service Control Scheme in Wireless Asynchronous Transfer Mode

This chapter explores virtual destination (VD) / virtual source (VS) –based available bit rate (ABR) flow control performance, targeting wireless asynchronous transfer mode (WATM) application that can incur long link-delays because of employing radio-medium sharing and/or radio-specific data link control schemes. As this chapter reveals, the conventional VD/VS scheme has difficulty in sustaining satisfactory ABR performance, when it is applied to long-delay-causing WATM; it suffers from significant increase in the necessary buffer capacity. To ensure the ABR performance in WATM, this chapter proposes a new VD/VS coupling scheme using a feed-forward congestion indication. The proposed scheme controls the allowed cell rate of a source end system in a feed-forward manner by predicting the queue length at the time the WATM-associated-round-trip ahead. Simulation results show that the proposed scheme exhibits excellent ABR performance with a long delay of the divided loop on the radio-link side. It is also verified that the proposed scheme is rather robust against uncertainty and/or time variation regarding the predetermined radio link delay.

4.1 Overview

With the goal of mobile multimedia communications, the demand for wireless ATM (WATM) is continuing to grow [17]. The available bit rate (ABR) service category [29] defined by the ATM Forum is intended for services and applications that require a low cell loss rate (CLR) but can tolerate delay and delay variation. ABR will be widely used in asynchronous transfer mode (ATM) networks, since it efficiently manages traffic and provides fairness with respect to throughput per user. It is, therefore, essential for WATM, which features simultaneous multi-user operability, to support ABR services.

To avoid traffic congestion in the ATM layer, ABR employs a rate-based closed-loop flow control [29]. In this flow control, resource management (RM) cells are used to control the transmission rates of the user terminals. RM cells are periodically sent from the source end system (SES) to the destination end system (DES) to detect congestion in each virtual channel (VC). The RM cell for this case is called a forward RM (FRM) cell. Upon receiving an FRM cell, the DES

The contents of this chapter are based on [70].

returns an RM cell in the same VC as a response to the FRM cell. The RM cell is referred to as the backward RM (BRM) cell. The bandwidth allocated for ABR services must be shared equally and efficiently among all active sources, that is, throughput fairness. Unfortunately, as revealed later, ABR performance such as throughput fairness, required buffer capacity, and convergence speed when the number of active ABR sources is changed is degraded when the round trip time (RTT) in the ABR flow control loop is long. Note that the RTT is defined through this chapter as the duration between when an SES sends an FRM cell and when the SES receives the BRM cell sent by the DES that received the FRM cell.

In providing satisfactory ABR services, RTT is a critical factor to be considered. WATM can cause a considerably long RTT (e.g., 8 ms in [30]) compared to those of fixed networks. Specifically, WATM systems will mostly employ a time division multiple access (TDMA)-time division duplex (TDD) based approach for their medium access control (MAC) as discussed in several standardizing organizations (e.g., [31]). These systems queue ATM cells and send them at the assigned time-slot in a TDMA-TDD frame. Furthermore, their data link control (DLC) may employ an automatic repeat request (ARQ) scheme to compensate for error-prone radio link transmission. These MAC and/or DLC methods can contribute to the long WATM-associated RTT (WATM-RTT). Note that other delay factors including radio propagation and signal processing can be neglected compared with the WATM-RTT caused by MAC and DLC. As revealed later, a long RTT can substantially increase the necessary buffer capacity for assured ABR fairness performance; otherwise, it results in significant degradation of fairness. Especially, when a WATM network inter-works with a fixed network, the WATM-RTT can considerably degrade the whole network-ABR performance, even when the fixed network causes a short link delay.

To ensure ABR performance in long RTT-causing networks, using a virtual destination (VD) / virtual source (VS) loop segmentation is suggested [29]. VD/VS loop segmentation aims to shorten the RTT by dividing the end-to-end flow control loop into several shorter loops. VD/VS methodology to couple two adjacent loops is implementation specific. To data, several VD/VS coupling schemes based on feedback flow control approaches have been proposed [32],[33]. They perform well when the RTT of the divided loop is moderate; however, they require an unfavorably large buffer capacity to cope with the WATM-RTT which is hardly reduced by segmentation.

Given the facts above, this chapter proposes a new ABR flow control scheme incorporated with VD/VS loop segmentation using feed-forward congestion indication. The proposed scheme predicts the queue length of the WATM-RTT; then, it controls the allowed cell rate (ACR) of an SES in a feed-forward manner. It achieves excellent ABR performance with a rather small buffer capacity at the associated ATM switch.

This chapter comprises six sections. Section 2 describes the generic ABR flow control. Section 3 clarifies the WATM-DLC/MAC-related assumptions made by this chapter and the technical challenge to be solved. The operating principle of the proposed scheme is then given in Sect. 4. Section 5 deals with the performance evaluation of the proposed scheme. Finally, Sect. 6 concludes this chapter.

4.2 Generic ABR Flow Control and VD/VS Segmentation

The Traffic Management Specifications issued by the ATM Forum technical committee specify the SES, DES and switch behaviors for the binary feedback and those for the explicit rate feedback [29],[34]. The primary parameters of the ABR flow control are the peak cell rate (PCR), minimum cell rate (MCR), initial cell rate (ICR), ACR, rate increase factor (RIF), rate decrease factor (RDF), and Nrm (Nrm: the maximum number of cells that a source may send between two consecutive FRM cells). These specifications expect that the ABR-available bandwidth will be divided into the active traffic sources in a fair, efficient, and agile manner.

First, on virtual channel (VC) establishment, the SES sends data and RM cells at a transmission rate that does not exceed the ACR of the SES. The initial value of the ACR is set by the ICR. The SES can send an FRM cell, the DES returns a BRM cell using the same VC as a response to the FRM cell. The FRM and BRM cells basically carry VC congestion information using several fields such as the congestion indication (CI) and explicit rate (ER). For notification of VC congestion, the ATM switches, through which the cells travel, set the explicit forward congestion indication (EFCI) bit of the cells to one if the switches are in a congested state. Note that the EFCI bit is one of ATM cell header bits and is used to inform the DES of VC congestion. When the DES sends a BRM cell, the BRM-cell’s CI bit is set to one if the last-received-cell’s EFCI bit was set to one. Through this procedure, the DES informs the SES whether there is congestion in the VC. When the SES receives a congestion-indicating BRM cell, the ACR is decreased as

$$ACR \Leftarrow \mathrm{Max}(\, ACR - ACR \times RDF,\, MCR\,). \tag{4.1}$$

On the other hand, if the CI bit is not set to one (namely, no congestion), the ACR is increased as

$$ACR \Leftarrow \mathrm{Min}(\, ACR + PCR \times RIF,\, PCR\,). \tag{4.2}$$

Parameter ER is used to control the ACR directly. ATM switches may use the ER field of the BRM cell to explicitly inform the SES of the allowable cell rate. Each ATM switch calculates an allowable cell rate for the SES according to the current quantitative congestion indication. Then, if the ACR is larger than the ER, the SES changes the ACR to the ER. In general, however, the

CI-based rate control is simpler in terms of implementation complexity than the ER-based rate control [35]. Therefore, the CI-based rate control rather than the ER-based rate control is the focus of this chapter. As verified later, by employing the CI-based rate control, the proposed scheme achieves satisfactory ABR performance.

A VD/VS structure is shown in Fig. 4-1, where the CI-based rate control is considered [29]. ATM cells from the SES are once queued in the VD/VS coupling section, which links the VD-side and VS-side. They are then sent to the ATM switch according to the current ACR on the VS-side. Figure 4-2 shows the processor structure of the simple VD/VS coupling scheme. The simple VD/V S coupling scheme uses pre-defined queue threshold QT (see Fig. 4-2) to identify VC congestion by comparing the current queue length with QT [36]. This is based on the fact that the queue length indicates the degree of congestion in the adjacent segment. If the queue length exceeds QT, the BRM-cell's CI bit for the SES is set to one. This VD/VS coupling scheme shortens the actual RTT by loop segmentation.

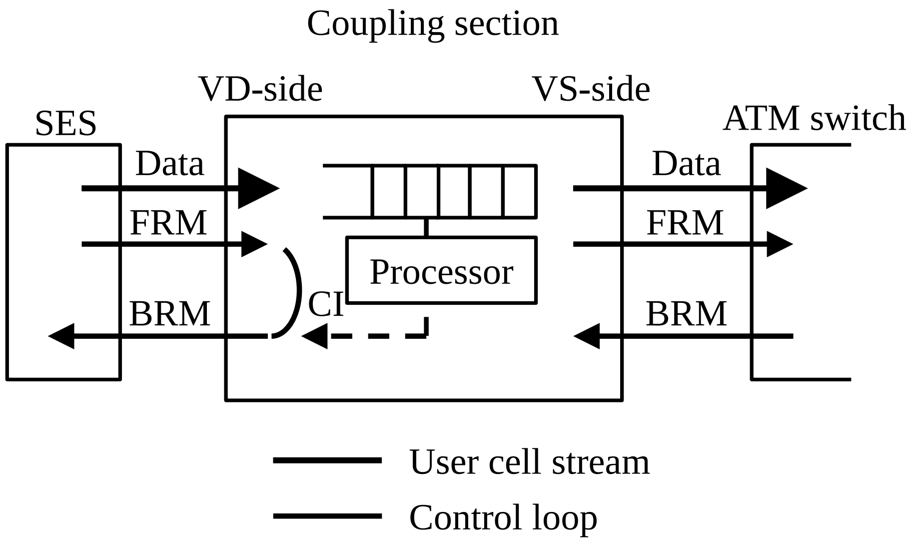
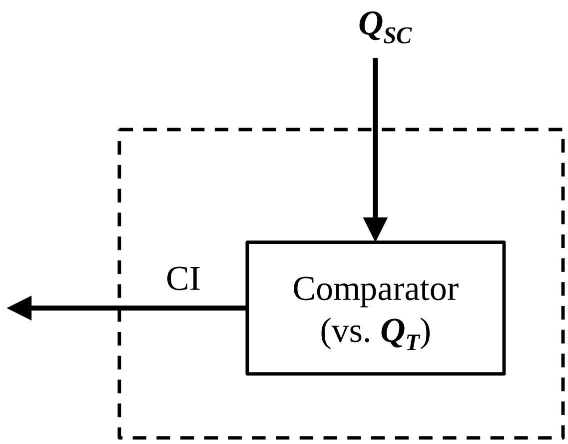


Fig. 4-1. VD/VS structure.



Q_{SC}: current queue size
Q_T : queue threshold (predetermined)
CI : congestion indication

Fig. 4-2. Processor structure (simple VD/VS coupling scheme).

4.3 WATM-DLC/MAC-related assumptions and link-delay-related issue

This section first clarifies the WATM-DLC/MAC-related assumption. The issue caused by WATM link delay is also presented.

4.3.1 WATM-DLC/MAC-related assumptions

This chapter supposes that cell stream regeneration (CSR) is employed in the DLC/MAC layer [37], and the proposed VD/VS coupling scheme handles a predetermined constant delay (see Fig. 4-3). As shown in Fig. 4-3, the information cells incoming from the ATM layer are usually packed into a radio packet in the DLC/MAC layer with the time stamp indicating the arrival time of each cell. On the receiving side, the cell stream is regenerated based on the added time stamps so as to satisfy a given cell delay variation (CDV) requirement (e.g., less than 1 msec in [31]). When ARQ is used to retransmit error cells, the time for ARQ should also be considered. In this case, to eliminate an ARQ-causing large CDV, a rather long predetermined constant delay is given on the receiving side, considering the MAC frame period and ARQ-related parameters (e.g., the buffer size, the number of re-transmissions, etc.). Based on the assumptions above, this chapter supposes that the WATM-RTT is given as a constant.

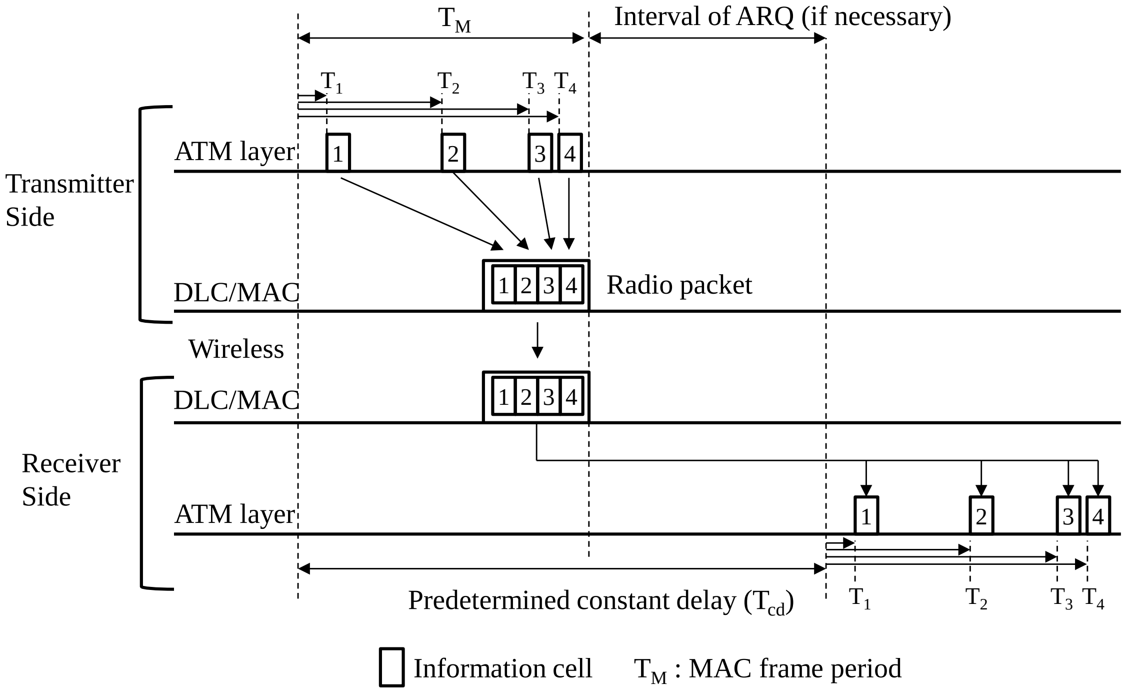
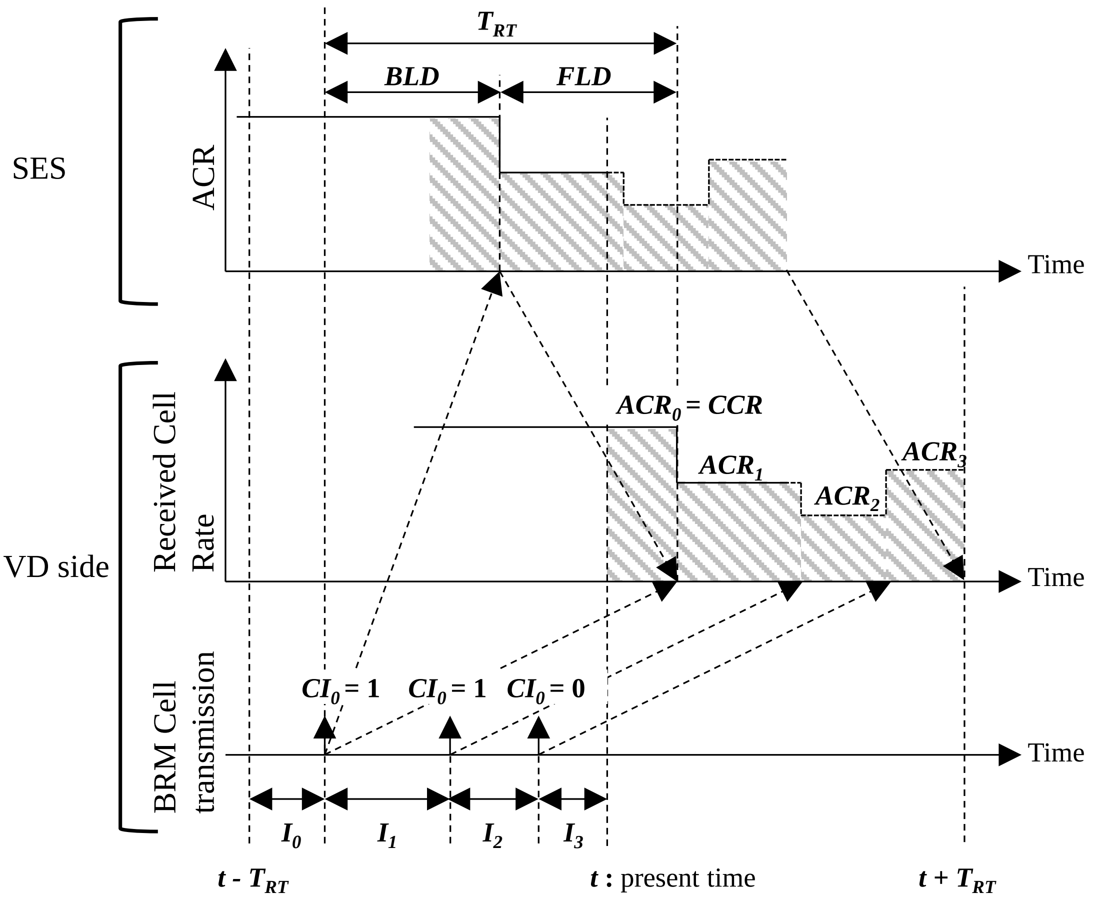


Fig. 4-3. Assumed cell stream regeneration (an illustrative example).

4.3.2 Link-Delay Related Issue

Figure 4-4 shows how the received cell rate changes on the VD-side according to the CI bits of BRM cells once sent from the VD-side. When a BRM cell is sent from the VD-side to an SES, the SES changes its ACR at the time the-backward-link-delay (BLD)-later than the time when the BRM cell transmission on the VD-side. The change in the ACR is then observed on the VD-side at the time the-forward-link-delay (FLD)-later than the time when the ACR changes at the SES. That is, the sum of the BLD and FLD time is the WATM-RTT, TRT. In short, it takes TRT for the VD-side to recognize the ACR change caused by the CI bit which was sent by the BRM cell from the VD-side. Moreover, Fig. 4-4 indicates the relationship between the BRM-cell CI bits sent from the VD-side, the ACR set at the SES according to the received CI bits, and received cell rate on the VD-side. Note that Fig. 4-4 assumes that all traffic patterns are persistent (i.e., greedy models): the received cell rate on the VD-side is identical to the ACR. Defining the present time as t, the VD-side received cell rates between t and $t+T_{RT}$ are determined by the BRM-cell's CI bits sent from the VD-side between $t-T_{RT}$ and t (see Fig.4-4). In other words, the number of cells which are supposed to be received between t and $t+T_{RT}$ (the hatched area in Fig. 4-4) is uncontrollable at present time t. Because of these uncontrollable cells, the queue-length of the VD/VS coupling section can significantly increase depending on VS-side conditions in that period. Consequently, ABR performance is considerably deteriorated as shown in Sect. 5.



T_{RT} : WATM-RTT *BLD* : Backward link delay *FLD* : Forward link delay
I_n : Interval Between BRM cell transmissions

Fig. 4-4. Relationship between received cell rate on VD-side and CI bit of BRM cell.

4.4 Operating Principle of Proposed VD/VS Coupling Scheme

To cope with a long WATM-RTT, the proposed VD/VS coupling scheme controls the ACR of an SES in a feed-forward manner by predicting the queue length of the WATM-RTT ahead. Specifically, the proposed scheme calculates the number of forthcoming cells that will be received on the VD-side between t and t+T_{RT} (where t denotes the present time, see Fig. 4-4). For this calculation, the proposed scheme memorizes, over the past T_{RT} -duration, the CI-bit values of the BRM cells (sent to the SES) and the intervals between two consecutive BRM cells (CI_k, and I_k, k: an integer, in Fig. 4-4, respectively). Then, the future queue length of the VD/VS coupling section is predicted. Using the predicted queue length, the proposed VD/VS coupling scheme decides in a feed-forward manner whether or not the VC will become congested.

The queue-length prediction methodology is explained below in detail. First, the SES-expecting ACRs between t and t+T_{RT} are calculated at the VD/VS coupling section using the current cell rate (CCR) and the CI values (see Fig. 4-4). The CI values used for the calculation are included in the BRM cells that have already been sent between t-T_{RT} and t. Note that the CCR is obtained from one of the fields of the FRM cell which the VD/VS coupling section has received from the SES. Suppose that N BRM cells have been sent to the SES between t-T_{RT} and t, and the values of CI_0, CI_1, … , CI_{N-1} have been set to the zero-th, first, ..., and (N-1)-th BRM cells, respectively. The relationship between the ACR_k (i.e., the ACR after receiving the (k-1)-th BRM cell at the SES) and the ACR_{K+1} is given as

$$ACR_{k+1} = \begin{cases} \mathrm{Min}(ACR_k + PCR \times RIF, PCR) & \text{if } CI_k = 0 \\ \mathrm{Max}(ACR_k - ACR_k \times RDF, MCR) & \text{if } CI_k = 1 \end{cases} . \qquad (4.3)$$

Note that ACR_0 is obtained as the CCR. In general, a VD/VS coupling scheme knows the CCR from the CCR field of the last received FRM cell.

Using the relationship described above, the number of cells received on the VD-side between t and t+T_{RT}, N_C is calculated as

$$N_C = \Sigma ACR_k \times I_k , \qquad (4.4)$$

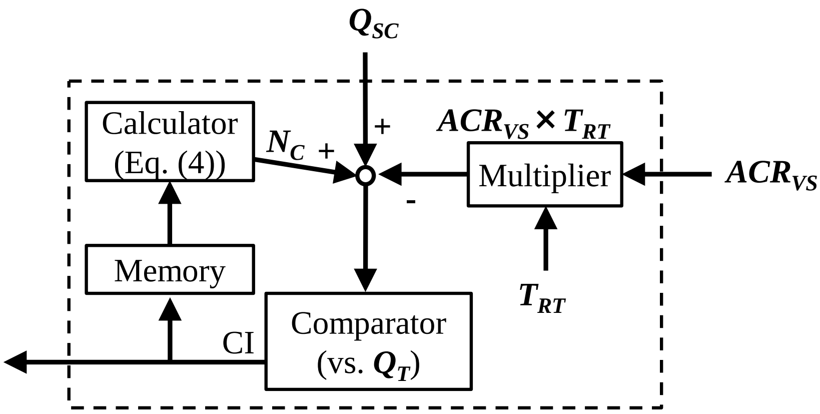
where I_0 is the interval between time t-T_{RT} and the transmitting time of the zero-th BRM (i.e., the initial BRM cell after time t-T_{RT}), I_k ($1 \leqq k \leqq N$-1) is the interval between the transmitting time of the

(k-1)-th BRM cell and that of the k-th BRM cell, and I_N is the interval between the transmitting time of the (N-1)-th BRM cell and present time t. N_C indicates the number of cells sent from the SES but not yet received on the VD-side, namely, the number of cells still in the radio link.

Using N_C, obtained by Eq. (4.4), and assuming that the ACR on the VS-side, ACR_{VS}, is not changed during the period of T_{RT} from now, the future queue length, Q_S, is estimated by

$$Q_S = Q_{SC} + N_C - ACR_{VS} \times T_{RT} , \qquad (4.5)$$

where Q_{SC} is the current queue length. The proposed scheme uses Q_S to detect the VC congestion. Figure 4-5 shows the processor structure of the proposed scheme. The processor comprises a calculator, memory, multiplier, and comparator. The CI values are stored in the memory. The calculator outputs N_C based on Eq. (4.4). The multiplier yields $ACR_{VS} \times T_{RT}$ using the ACR obtained from the VS-side. Then, the comparator compares Q_S obtained by Eq. (4.5) with threshold Q_T. If Q_S is larger than Q_T, the Cl is set to one; otherwise, it is set to zero. Note that this chapter deals with situations where the WATM-RTT is explicit or its approximate value can be obtained, as is often in the case of systems using a fixed frame structure [30].



Q_{SC}: current queue size
Q_T : queue threshold
ACR_{VS} : ACR at VS side
T_{RT}: WATM-RTT (pre-determined)
CI : congestion indication
N_C : number of cells sent from SES

Fig. 4-5. Processor structure (proposed scheme).

4.5 Performance Evaluation

4.5.1 Simulation Models and Parameters

The simulation model of the single control loop is shown in Fig. 4-6. The connections are established between SESl and DES1, between SES2 and DES2, and between SES3 and DES3. It is assumed that the WATM link which connects SESl and ATM switch SW1 has the WATM-RTT, T_{RT}, while the delays of other links are assumed to be negligibly small. The transmission rates of all links are l55.52 Mbps. The ATM switches conduct explicit forward congestion indication (EFCI) and ER marking [29]. In this simulation model, the enhanced proportional rate control algorithm (EPRCA) [38] is used for ER marking. The simulation model of the double control loop using a VD/VS is shown in Fig. 4-7. In this simulation, the radio link has a constant delay, which is the same as that for the single control loop simulation. The VD/VS loop segmentation is applied between SW1 and SES1 to divide the ABR control loop into two segments.

Primary ABR parameters used for the simulation are listed in Table 4-1. These are based on the ATM Forum default values [29]. Note that in the above simulation models, SES1 establishes a connection to DES1 and then begins to transmit after the two other connections, SES2-DES2 and SES3-DES3, have reached a stable state with an almost equivalent transmission rate. All traffic patterns are persistent (i.e., greedy models): the SESs always have the cells to send after connections are established.

Table 4-1. Primary parameters of ABR.

Parameters	Value
PCR	155Mbps
MCR	0
ICR	PCR
RDF	1/16
RIF	1/16
Nrm	32
Trm	100
CDF	1/16

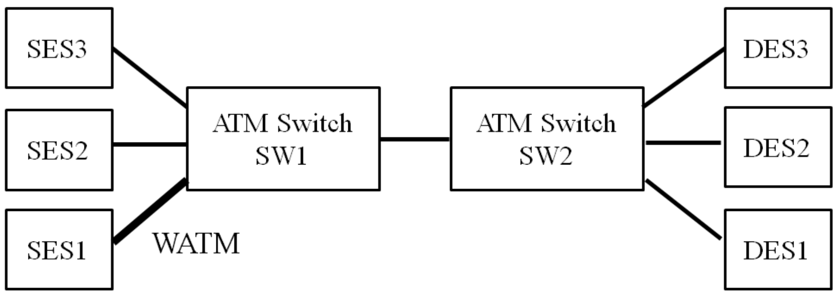


Fig. 4-6. Configuration of single control loop model.

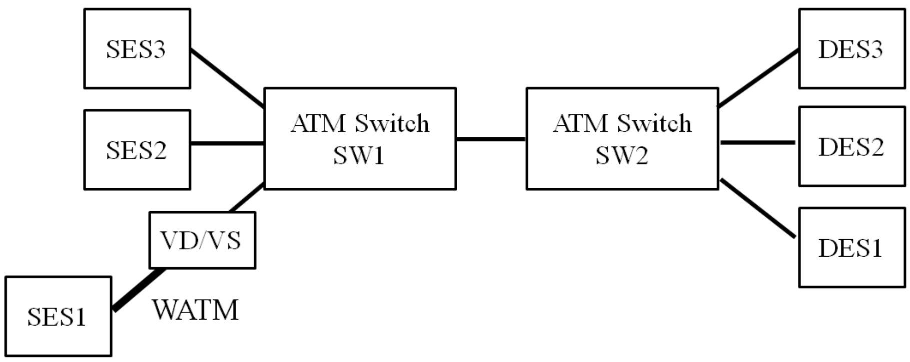


Fig. 4-7. Configuration of double control loop model.

4.5.2 Evaluation Results and Discussion

4.5.2.1 Fairness Index Evaluation

This section evaluates the ABR performance of the proposed scheme. Prior to the evaluation, the performance of the single control loop and the double control loop with the conventional simple coupling scheme is examined for comparison. Moreover, problems of the conventional feedback-based schemes are discussed. In this chapter, the ABR performance is evaluated using the fairness index at ATM switch SW1. Note that the fairness index is calculated based on the averaged CCR as suggested in [38].

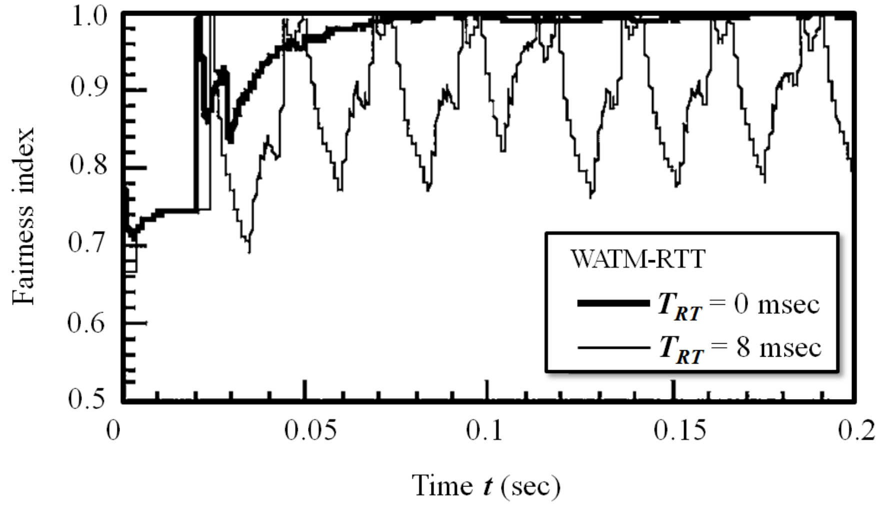


Fig. 4-8. Fairness index with single control loop model.

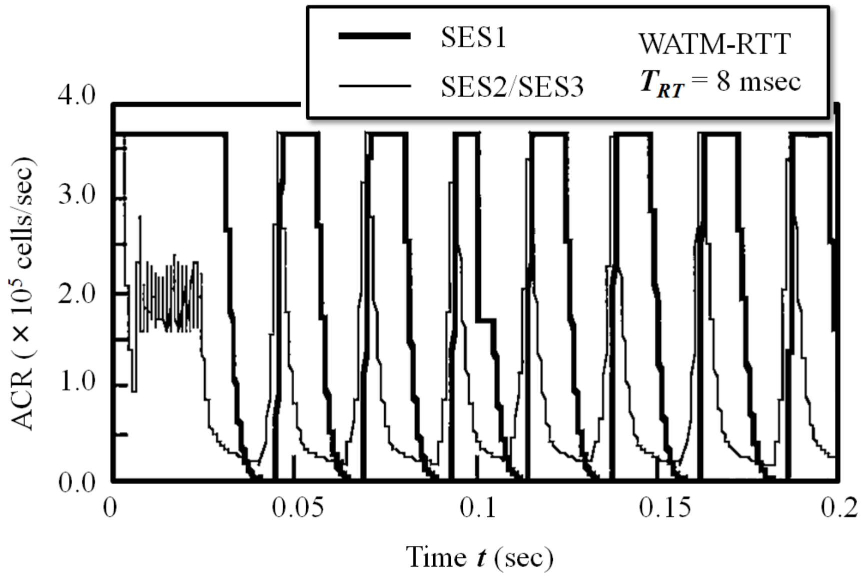


Fig. 4-9. ACR with single control loop model.

4.5.2.1.1 Single Control Loop (Without VD/VS Loop Segmentation)

The fairness index in the single control loop model is plotted in Fig. 4-8 as function of WATM-RTT T_{RT}. The ACRs of SES1 and SES2/SES3 are also depicted in Fig. 4-9. When T_{RT} =0, the fairness index rapidly converges to one, which represents the best condition with respect to fairness. When T_{RT} =8 msec, however, the fairness index oscillates continuously (see Fig. 4-8). This is because, When SW1 detects a congested state, SES2 and SES3 decrease their transmission rates immediately, but SES1 (i.e., a WATM terminal) continuously sends the cells at the PCR until at least 8 msec later. On the other hand, after the congestion has dissipated, all the ACRs reach the PCR almost at the same time. This is because, although SES1 starts to increase its ACR about 4 msec (one-Way trip time, i.e., T_{RT} /2) later than SES2/SES3 do, a number of RM cells queued at the buffer in SW1 contribute to the quick ACR increase at SESl which takes only sixIteen (i.e., a reciprocal of the RIP) BRM cells to reach the PCR from the bottom (see Fig. 4-9).

4.5.2.1.2 Double Control Loop (Conventional)

Furthermore, the fairness index obtained using the double control loop model with the conventional simple VD/VS coupling scheme is plotted in Fig. 4-10. In this simulation, the simple VD/VS coupling scheme described in Sect. 4.2 is used and the queue threshold, Q_T, is 1000 cells. Note that, when Q_T exceeds 1000 cells, the simple VD/VS coupling scheme performs well with the fairness index close to one. This result shows that the fairness index is more than 0.95 at t = 50 msec (see Fig. 4-10). From Fig. 4-10, the bandwidth of the link appears to be allocated equally to each of the SESs (compare Fig. 4-10 with Fig. 4-8). However, there is still a critical problem: the WATM link ends up having a very low transmission rate as shown in Fig. 4-11 which depicts the ACR and actual transmission rate in the direction to SW1 on the VS-side. In this case, as opposed to the ACR allocated by SW1, the actual transmission rate is often reduced to a very low rate. This is because the queue in the VD/VS coupling section (see Fig. 4-1) becomes empty during some periods as shown in Fig. 4-12. The queue-empty situations are caused by a long period of a very low ACR or zero at the SES1 as shown in Fig. 4-13. Such queue-empty situations can be a potential problem with a closed feedback loop with a long WATM-RTT .

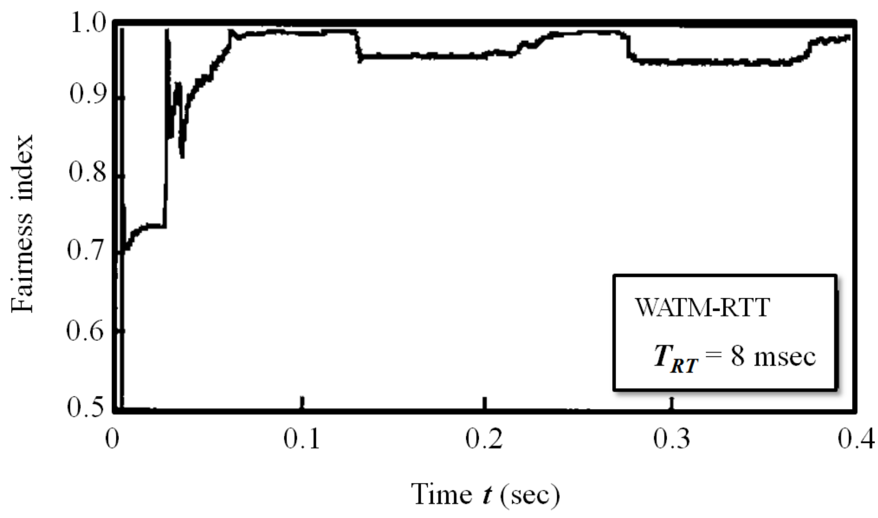


Fig. 4-10. Fairness index with simple VD/V S coupling scheme.

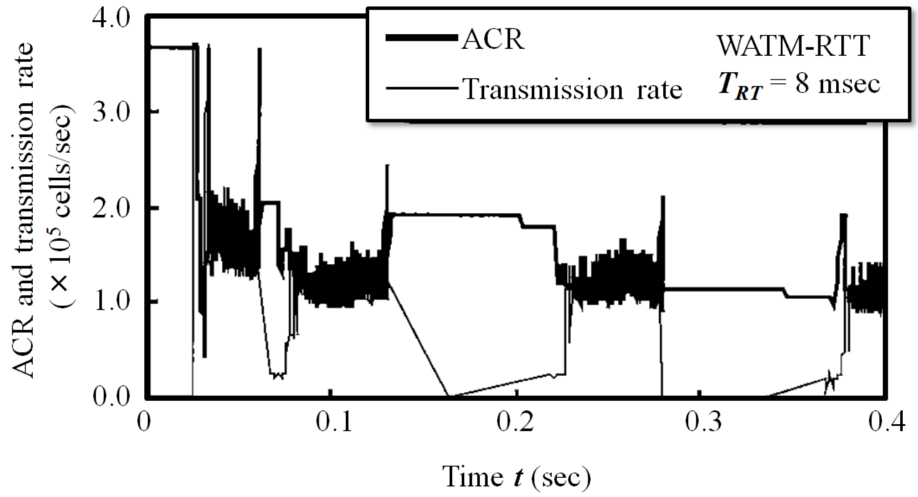


Fig. 4-11. ACR and transmission rate on VS-side With simple coupling scheme.

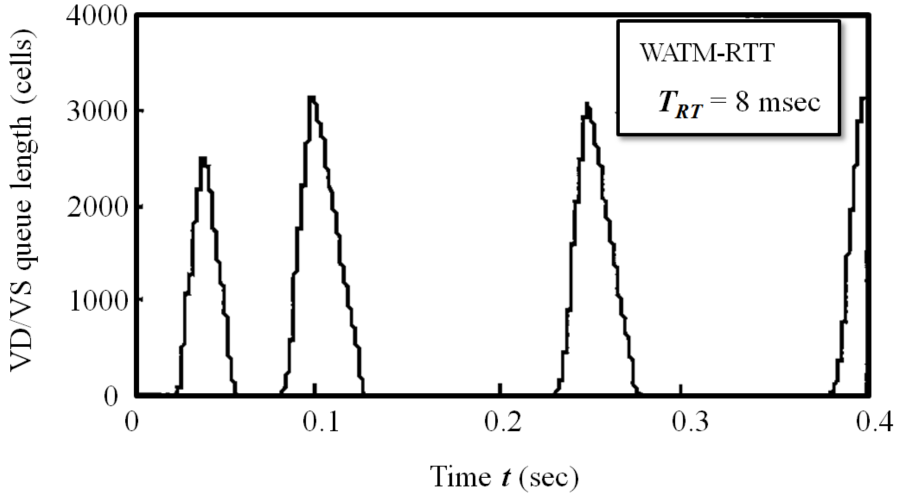


Fig. 4-12. VD/VS queue length with simple coupling scheme.

4.5.2.1.3 Double Control Loop (Proposed)

From the above observation, we realized that, to transmit the data following the ACR allowed by SW1 on the VS-side, the queue-empty situations should be avoided. In contrast to the conventional feedback based VD/VS coupling schemes, the proposed feed-forward based scheme avoids such WATMRTT-contributing problematic situations by predicting the VD/VS-coupling-section queue length at the time the WATM-associated-round-trip ahead. The fairness index obtained using the proposed scheme is shown in Fig. 4-14. The index completely converges to one without fluctuation in the region where time is greater than 0.1 sec, which indicates ultimate fairness. In addition, deviation from complete fairness is reduced by the proposed scheme. This is because the proposed scheme enables the transmission rate to completely follow the allowed ACR as shown in Fig. 4-15. The queue length for this case is depicted in Fig. 4-16. From this figure, it is clear that no queue-empty situation occurs. By comparing Fig. 4-16 with Fig. 4-12, it is obvious that the maximum VD/VS queue length of the proposed scheme is approximately 40% less than that of the conventional scheme. Namely, the proposed scheme contributes to reducing the required buffer size in the VD/VS coupling section.

The ACR and transmission rate on the VS-side with the proposed scheme is shown in Fig. 4-15 for T_{RT} = 8 msec. As shown in Fig. 4-17 where T_{RT} = 16 msec, the transmission rate coincides with the ACR similarly to the case of T_{RT} = 8 msec. It can be said that the proposed scheme is effective for a rather long WATM-RTT.

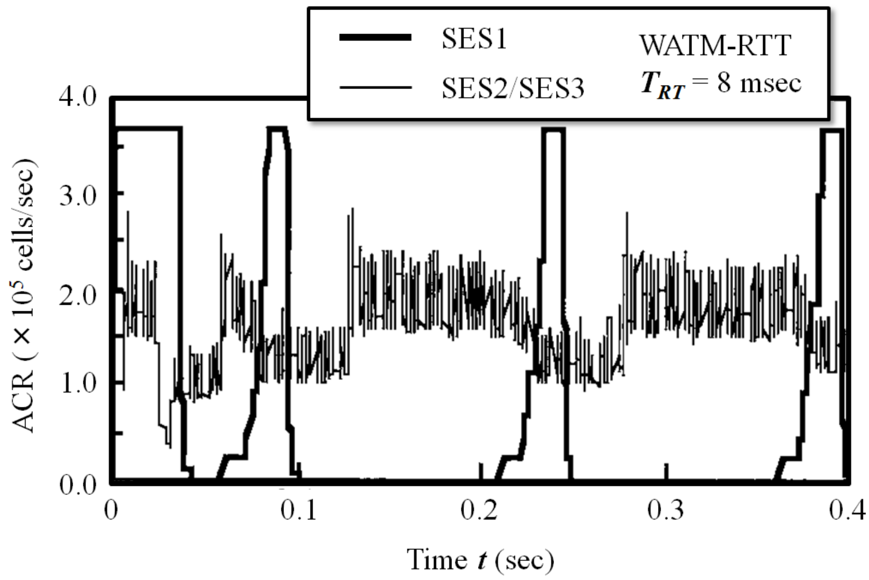


Fig. 4-13. ACR with simple coupling scheme.

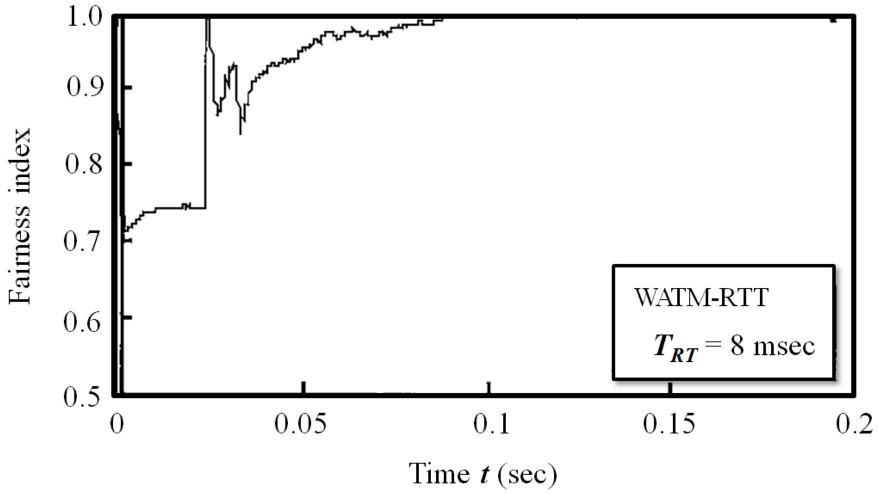


Fig. 4-14. Fairness index with proposed scheme.

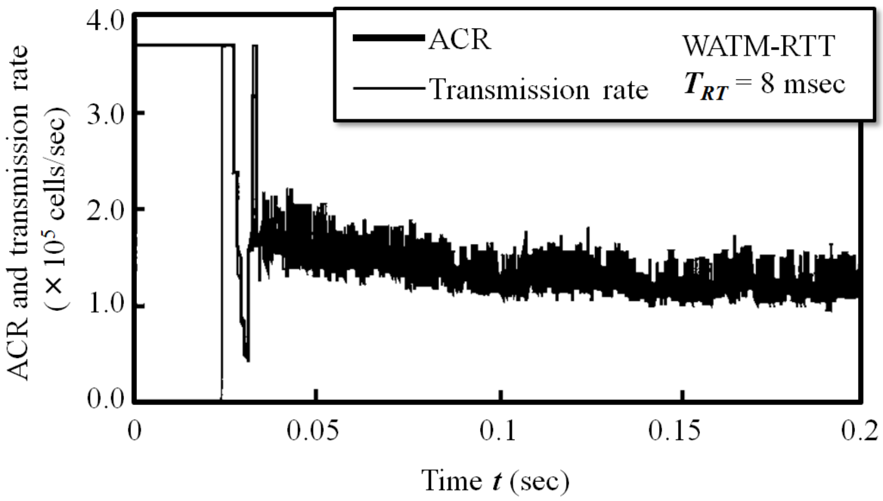


Fig. 4-15. ACR and transmission rate on VS-side with proposed scheme.

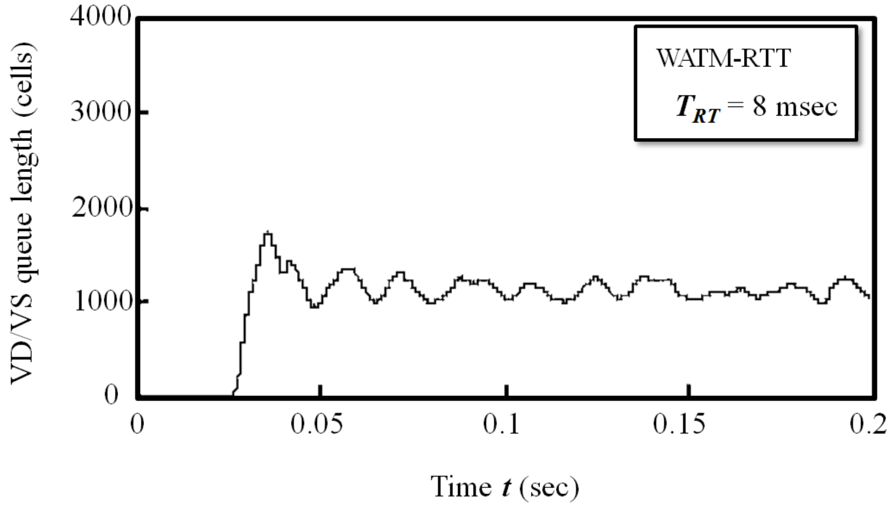


Fig. 4-16. VD/VS queue length with proposed scheme.

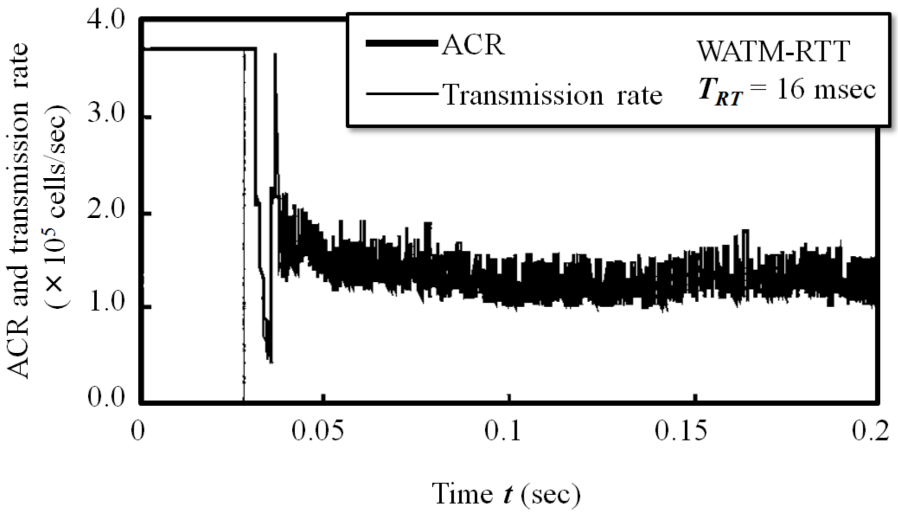


Fig. 4-17. ACR and transmission rate on VS-side with proposed scheme.

4.5.2.2 Stability Against WATM-RTT Deviation

Finally, we discuss the stability against the WATM-RTT deviation, D_{RT}, which is defined as the difference between the predetermined WATM-RTT, T_{RT}, and the actual one. In the proposed scheme, the predetermined WATM-RTT is used to predict the number of cells in the WATM loop based on the assumption described in Sect. 4.3. However, in practice, some WATM-RTT deviation is unavoidable. As an example, the resolution of a time stamp used in CSR and/or processing delays in the operating circuits can be a cause. Hence, we evaluate the stability against WATM-RTT deviation. For this evaluation, this chapter uses the root mean square (rms) value of the difference between the current queue size and queue threshold of Q_T: 1000 cells. The rms value indicates a degree of fluctuation around the queue threshold (thus, let us call this value the fluctuation indicator). Namely, larger fluctuation more likely incurs a queue-ernpty situation; then, it results in an alrnost-zero-transinission-rate, even when the given ACR is large.

Figure 4-18 shows the fluctuation indicator defined above, where T_{RT} = 8 and 16 msec. Through further simulation, it has been confirmed that, when the fluctuation indicator is less than 250 cells, no queue-empty situation occurs. In other words, the proposed scheme performs well in the region of -1.5 < D_{RT} < 2 msec: this holds true for either case when T_{RT} = 8 msec or 16 msec. This is because a factor obtained by multiplying the actual transmission rate with the WATM-RTT deviation, D_{RT}, determines the degree of fluctuation. Thus, it can be said that, even when there is a rather large WATM-RTT deviation, the proposed scheme can perform satisfactorily with tolerance.

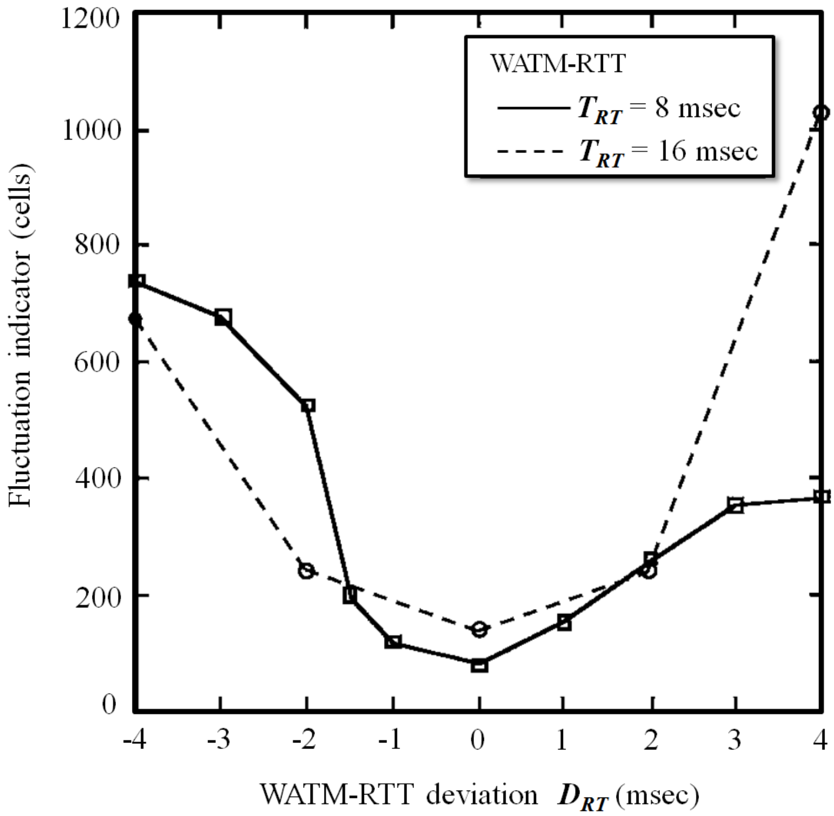


Fig. 4-18. Fluctuation indicator with proposed scheme.

4.6 Conclusions

This chapter explored VD/VS-based ABR flow control performance targeting application to WATM in which large link-delay can be inherent. It was revealed that the conventional feedback-based VD/VS scheme has difficulty in sustaining satisfactory ABR performance, when it is applied for long-delay-causing WATM. To ensure the ABR performance in WATM, We proposed a new VD/VS scheme that controls the ACR of an SES in a feed-forward manner. The proposed scheme calculates the number of cells in the WATM-associated loop and predicts the queue length of the WATM-RTT. The proposed scheme exhibits excellent ABR performance even when the WATM-RTT is rather long. In addition, the proposed scheme possesses robustness against uncertainty and/or time-variation regarding the predetermined WATM-RTT.

At a moderate cost of implementation complexity for the queue length prediction, memorizing the BRM cell transmitting time and associated CI bits, the proposed scheme enables long-RTT-causing WATM to well support ABR service in practice. When systems have extensive WATM-RTT variation or difficulty in identifying the WATM-RTT, technical modification may be needed. A study on a scheme that requires no WATM-RTT information is under way.

Chapter 5
Quality-of-Service Control Scheme in Distributed Access Control System

This chapter proposes a new quality of service (QoS) control scheme that alters the Inter Frame Space (IFS) length of stations (STAs) according to the traffic circumstances of the wireless local area network (WLAN). It controls the amount of data transmitted in order to guarantee the throughput of each user and improve transmission delay performance. In this chapter, we show that the problem of “priority inversion” occurs and that the QoS is degraded when conventional schemes are employed in the medium access control (MAC) layer. The effectiveness of the proposed scheme is validated by comparison against conventional schemes. The results reveal that the proposed scheme can guarantee QoS in terms of throughput even if there is lot of best effort traffic.

5.1 Overview

In recent years, the applications of wireless systems have expanded rapidly. The most popular network in the wireless domain is the IEEE 802.11 wireless local area network (WLAN)[39]. The reasons for its popularity are said to be its interoperability, mobility, flexibility and cost-effective deployment. The rapid adoption of WLANs and the rising popularity of multimedia applications have caused increasing demand for Quality of Service (QoS) enabled wireless communication. Due to the limited radio spectrum, however, the demand for wireless resources is likely to surpass the available wireless resource in the future. Therefore, it is inevitable that each access point (AP) will need to provide some kind of QoS in terms of throughput and transmission delay to users while efficiently accommodating the heavy traffic generated by stations (STAs).

The IEEE 802.11e standard [40] aims to add QoS to the IEEE 802.11 WLAN by employing Enhanced Distributed Channel Access (EDCA). In EDCA, STA behavior depends on the access parameters, namely Contention Window minimum (CWmin), Contention Window maximum (CWmax) and Arbitration Inter Frame Space (AIFS). CWmin and CWmax are used to determine a contention window (CW) size and limit the range of back-off time; AIFS determines the minimum length of channel monitoring period. EDCA defines four access categories (ACs) to provide traffic accommodation with different priorities. Each AC uses different access parameters so as to provide QoS differentiated services to users. That is, each AC offers a different level of QoS. In general, small parameter values raise transmission priority.

In EDCA, an STA wins the right to transmit a data frame over the radio channel according to its CW size, which is a probabilistic factor. So, it can be said that the transmission delay of an STA with high priority AC (high priority (HP) class, hereafter) is probabilistically smaller than the delay of an STA with low priority AC (this is referred to as a low priority (LP) class, hereafter). The same can be said with respect to throughput. We note that EDCA can provide QoS only when all the STAs transmit the data frame of the same size using the same physical (PHY) transmission rate. In the worst case, the throughput of a high priority STA is throttled by the traffic of the LP class, and the throughput of the LP class may exceed the throughput of the HP class. This situation is called "priority inversion."

In order to avoid priority inversion, much effort [41]-[43] has been made to modify the functionality of EDCA. Adaptive EDCA (AEDCA)[44][45] is an adaptive CW size control mechanism that changes CW size according to the collision rate of transmitted data frames. In this scheme, all STAs calculate their own frame collision rate from the number of successful and failed data frame transmissions. When it changes CW size after a successful frame, the STA calculates the new CW size using a multiplicative factor for each AC that is determined from its collision rate. More specifically, AEDCA sets a larger CW size than that of EDCA when the collision rate is higher than the predefined threshold so as to suppress the collision rate and guarantee QoS. Thus, AEDCA has the functionality of adjusting CW size according to the traffic condition and lessen the frequency of collision. Therefore, compared to EDCA, AEDCA offers an improved ability to differentiate QoS provisioning between ACs, while maintaining overall bandwidth efficiency. However, the scheme can provide adequate grade of QoS only when the collision rate is sufficiently representative of the traffic condition. For instance, when priority inversion is caused by the transmission of long data frames rather than the frequent transmission of short data frames, AEDCA may not detect the inversion because the collision rate is only slightly changed by the transmission of long data frames.

To solve the priority inversion problem, which frequently happens in the multi PHY transmission rate circumstance where STAs communicate using different PHY transmission rates, we propose a new dynamic IFS adaptation scheme that uses closed loop control and dynamically changes the IFS length so as to maintain the QoS of the HP class STAs. In the proposed scheme, the AP measures the throughput of HP class STAs by means of the leaky bucket method, judges whether congestion occurs or not, and informs the LP class STAs of the occurrence of congestion by sending one bit information in the ACK frame; this alters the IFS length of LP class STA's.

The remainder of this chapter consists of five sections. Section 2 describes EDCA and its problems. Section 3 explains the conventional schemes. In Sect. 4, we propose a novel IFS adaptation scheme. Section 5 details a performance evaluation of the proposed scheme. Finally, Sect. 6 concludes this section.

5.2 EDCA prioritization mechanism and technical issues

In the IEEE 802.11 MAC layer, the basic access scheme is based on the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) protocol with binary exponential back-off algorithm. In EDCA, an STA with a new packet to transmit monitors channel activity before transmission. The STA continues to monitor until the idle period equals AIFS, which is the period between frames and is different for each AC. After detecting the AIFS-length idle period, the STA enters the back-off process in order to reduce the probability of collision with data frames transmitted by the other STAs. In this process, the STA calculates a number that is randomly chosen in the range of zero to the CW size, and sets the number to the back-off counter. In EDCA, the fixed-length time period in which an STA will always be capable of determining if another STA has accessed the channel is defined as a slot time. The counter is decremented in terms of slots as long as the channel is sensed idle. The counter stops when the channel is busy due to other STA's transmissions and resumes when the channel is sensed as idle again for more than AIFS. The STA transmits its data frame when the counter reaches zero.

EDCA employs an exponential back-off scheme in order to avoid collisions as does the MAC protocol of IEEE 802.11. In the scheme, the CW size depends on the number of failed frames. At the first transmission attempt, CW size is set equal to CWmin. After each unsuccessful transmission, CW size is doubled, up to CWmax. In the IEEE 802.11 specification, CW size and inter frame space (IFS) are set the same for all STAs regardless of their traffic flows. Such a mechanism is not effective with regard to QoS support. Thus, IEEE 802.11e introduces service differentiation in EDCA by assigning different CW sizes and IFS values to the ACs. Figure 5-1 shows the transmission sequences for the different priorities. AIFS length represents data frame priority. Data frames with the highest priority have the shortest AIFS. For further differentiation, the CWmin can be set differently for different priority classes; classes with smaller CWmin have higher priority.

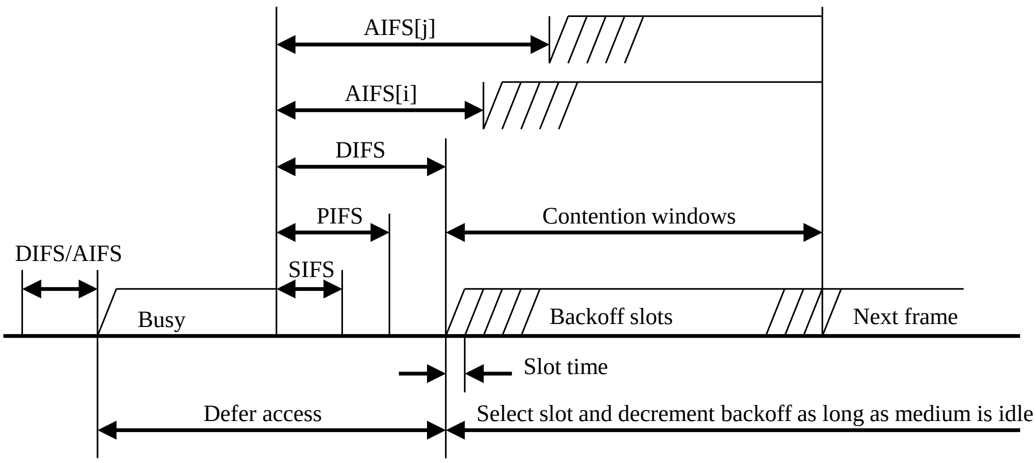


Fig. 5-1. EDCA back off sequence with different priorities.

However, the above is true only if all STAs have the same transmission condition. That is, EDCA can provide QoS only when all STAs transmit data frames of the same size using the same PHY transmission rate. Otherwise, the transmission delay of HP class may become larger than the transmission delay of LP class. For example, if HP class STA has a PHY transmission rate of 54 Mbit/s, and an LP class STA has a PHY transmission rate of 6 Mbit/s, the throughput of the HP class STA is impacted by LP class traffic and the throughput of the LP class may exceed the throughput of the HP class. This situation is called "priority inversion". There are two reasons for this problem. The first is that EDCA has no mechanism to alter access parameters according to the traffic condition. The other is that the scheme originally controls only the probability of gaining access rights to the channel and does not manage the actual traffic.

5.3 Conventional scheme

To solve the above-mentioned problem, many efforts have been made to enhance EDCA functionality [41]-[45]. Most schemes take the adaptive CW size control approach, where the state of the radio channel is represented by metrics such as throughput, transmission delay, and collision probability. AEDCA is one of the most effective schemes in terms of QoS performance [44][45]. The only difference of AEDCA from EDCA is how CW size is altered. AEDCA uses a dynamic procedure to change CW size according to its estimated collision rate.

In AEDCA, an STA holds its collision rate, R_a, which is calculated from the number of successful and failed transmissions [44]. AEDCA updates its CW size when it completes data frame transmission correctly or detects transmission failure. After each successful frame, CW size is updated by the following equation,

$$CW \Leftarrow \mathrm{Max}(\ CWmin,\ CW \times \mathrm{Min}(R_a, 0.8)\ ). \tag{5.1}$$

This formula means that, unlike EDCA, CW size is not made too small when the collision probability is high. When data frame transmission fails, AEDCA doubles the CW size, as does EDCA. That is

$$CW \Leftarrow \mathrm{Min}(\ CWmax,\ CW \times 2\ ). \tag{5.2}$$

EDCA reduces CW size more than necessary and this reduction can degrade the throughput due to the increase in collision probability; AEDCA avoids this throughput degradation and so achieves QoS guarantees even if collision is frequent. However, AEDCA fails to consider that STAs will transmit data frames at different rates. This weakness is serious if the transmission rate of the lower priority frames is low because they take longer to send.

5.4 Proposed dynamic IFS adaptation

Priority inversion can be suppressed by increasing the difference between LP class AIFS and HP class AIFS. Priority control schemes using AIFS are very effective because they eliminate the probabilistic factor from medium access. However, a long IFS leads to long channel monitoring periods which decreases system throughput because of the increase in overhead [46]-[48].

We avoid priority inversion by proposing a QoS control scheme that measures the throughput of prioritized STAs and alters IFS length of STAs to suit their circumstance so as to guarantee QoS. In the proposed scheme, the AP observes the traffic of each STA and orders LP class STAs to lower their frame transmission rates if the AP detects traffic congestion. When there is no congestion, the AP and STAs operate in the same way as in IEEE 802.11. This system operates uniquely only when traffic congestion is detected. Figures 5-2, 5-3, and 5-4 clarify this mechanism.

5.4.1 AIFS control loop

Figure 5-2 depicts the proposal's AIFS control mechanism. Unlike the conventional scheme, which infers the occurrence of congestion by using a distributed control method, the proposal precisely detects congestion by using a centralized control method. In the proposed scheme, the AP has a traffic observer that measures the traffic of each STA; it detects congestion by comparing the throughput of each HP class STA with the agreed transmission rate, which is negotiated when the STA joins the AP. After detecting congestion, the AP notifies STAs of the congestion by inserting congestion information into an acknowledgement (ACK) frame and sending it. Each STA has a transmission (TX) timing controller which adjusts AIFS to a suitable length according to the congestion information notified by the AP. Details of congestion detection and AIFS alteration are described below.

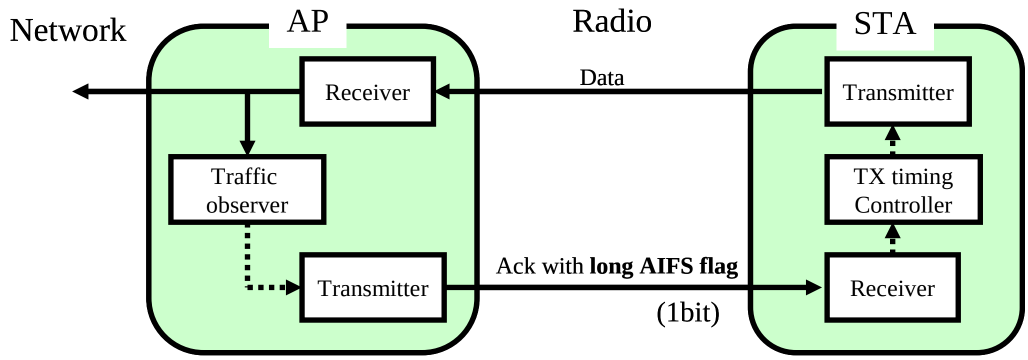


Fig. 5-2. Structure of AIFS control mechanism.

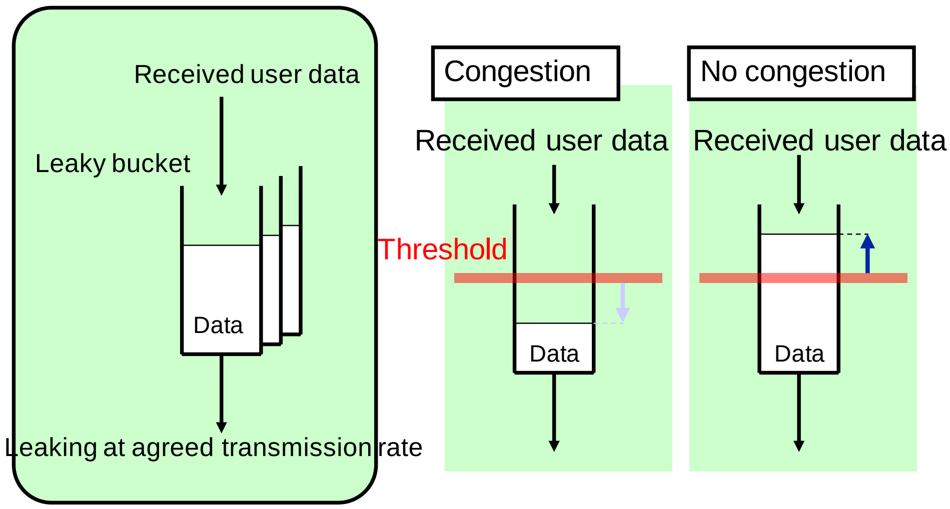


Fig. 5-3. Congestion detection mechanism.

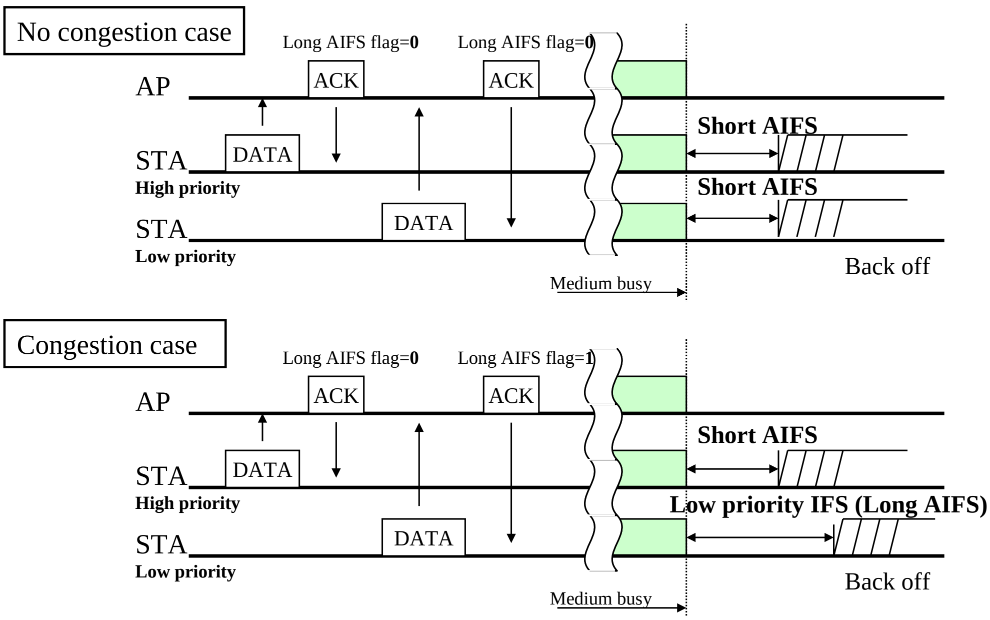


Fig. 5-4. Transmission sequence of AIFS control mechanism.

5.4.2 Congestion detection

Figure 5-3 shows the congestion detection mechanism of the proposed scheme. In order to detect congestion on the radio channel precisely, the AP employs the leaky bucket algorithm, which is often used in traffic shaping methods. The traffic observer of the AP has a leaky bucket for each HP class STA. When a data frame from a HP class STA is received at the AP, the user data conveyed by the data frame is put into the bucket of the sending STA. That is, the amount of the user data stored in the bucket (called the depth of the bucket hereafter) increases by the amount of the user data received. The bucket continually leaks the stored data at the agreed transmission rate of the STA. When the AP receives a data frame, the depth of the bucket, L_d, is updated by the following equation,

$$L_d = L_{d\_prev} + L_r - R_g * (T_{now} - T_{prev}), \tag{5.3}$$

where L_{d_prev} is the depth of the bucket before updating, L_r is the received user data size, R_g is the agreed rate, T_{prev} is the time at which the previous data frame was received, and T_{now} is the current time. When there is no congestion, STAs have sufficient radio resources and the bucket depth increases. On the contrary, bucket depth decreases if congestion is present. Therefore, the traffic observer can detect congestion accurately by comparing bucket depth against a pre-defined threshold. While the conventional scheme can only guess the congestion state, the proposed scheme can precisely identify it.

5.4.3 AIFS adaptation

The congestion information detected by the AP is carried by an ACK frame using the long AIFS flag, which is 1-bit long. Figure 5-4 shows the transmission sequence. If the AP receives a data frame in the congestion-free case, the AP sends an ACK frame with the long AIFS flag of 0. The STA that receives the ACK uses a shorter AIFS to access the radio channel the next time it tries to send a data frame. In the congestion case, the long AIFS flag sent to the LP class STA is set to 1. If an LP class STA receives this ACK, its uses the LP class AIFS, which is longer than the HP class AIFS, to transmit data frames. As a result, the LP class STA dynamically changes its AIFS length and throttles its transmission rate to suppress congestion. LP class STAs don’t use longer AIFS in the ordinary condition and priority control is achieved without degrading system throughput. STA operation is outlined below.

- STA checks the long AIFS flag when it receives an ACK frame.

- If the long AIFS flag is 0, the STA uses a short AIFS at the next data frame transmission.
- Otherwise, the STA uses a long AIFS.

Unlike the conventional schemes, the proposal realizes explicit rate control of STAs by AP congestion indication.

5.5 Performance evaluation

5.5.1 Simulation models

We used two simulation models to clarify the effectiveness of the proposal. The first simulation model realized an n-STA simulation, Model A; see Fig. 5-5. In Model A, the k-th STA communicates the k-th NODE in the wired network through one AP. Each STA transmits user data using User Datagram Protocol (UDP). As shown in Table 5-1, STA1 is HP class and other STAs are LP class. STA1 traffic is constant at 20 Mbit/s and the traffic of other STAs is 0.8 Mbit/s. The number of LP class STAs was varied in order to clarify the influence of LP class traffic on HP class throughput. Model A shows that priority inversion occurs even if AEDCE is employed when the LP class traffic increases with the increase in the number of LP class STAs. The second simulation model, Model B, assumes six STAs, see Figure 5-6. Table 5-2 shows the QoS class and traffic of each STA. Each HP class STA generates traffic at a different rate. The traffic of STA6 was changed in order to observe the throughput of HP class STAs.

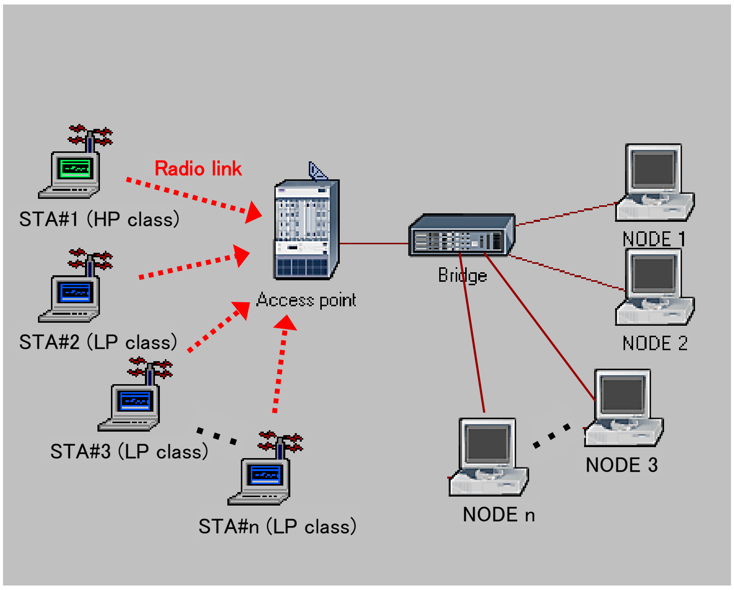


Fig. 5-5. n-STA simulation model (Model A).

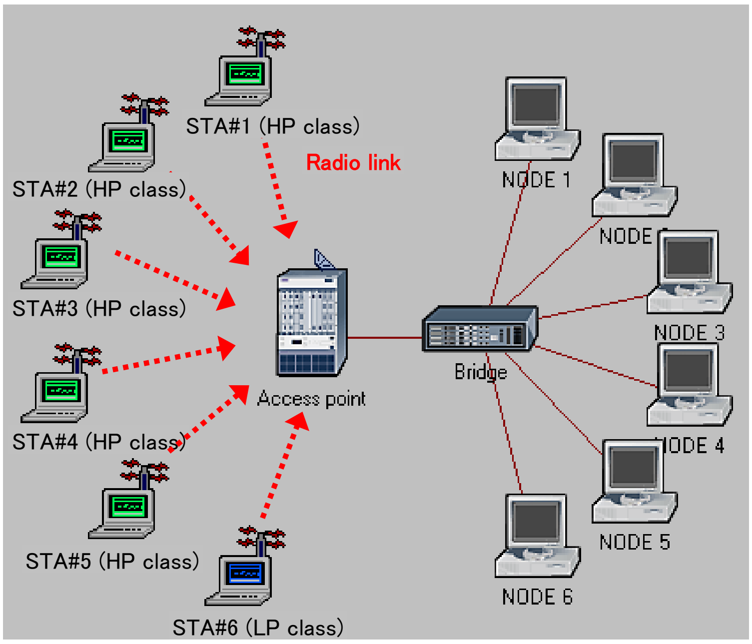


Fig. 5-6. Six-STA simulation model (Model B).

Table 5-1. QoS classes and traffic in n-STA simulation.

	QoS class	Traffic (Mbit/s)
STA1	High priority	20
Others	Low priority	0.8

Table 5-2. QoS classes and traffic in six-STA simulation.

	QoS class	Traffic (Mbit/s)
STA1	High priority	12
STA2	High priority	8
STA3	High priority	4
STA4	High priority	2
STA5	High priority	1
STA6	Low priority	0.1 - 1.0

We used Model B to show that the proposal achieves good performance even when several HP class STAs are active and each sends data frames at a different rate.

In these two models, HP class STAs transmit data frames using 16QAM(R=3/4), that is, the PHY transmission rate of the HP class is 54 Mbit/s. On the other hand, the PHY transmission rate of the LP class is 6 Mbit/s (BPSK(R=1/2)). The greater the difference is between HP class PHY transmission rate and LP class PHY transmission rate, the more readily priority inversion will occur. 16QAM(R=3/4) is the highest rate of IEEE802.11a and IEEE802.11g and BPSK(R=1/2) is the lowest rate. Therefore, these rate settings maximize the likelihood of priority inversion.

We compare the performances of the proposed scheme against those of the conventional scheme, namely AEDCA. STA prioritization was achieved by setting different CW sizes. The access parameters of AEDCA, shown in Table 5-3, were taken from Reference [44]. Table 5-4 shows the parameters of the proposed scheme. In the congestion-free case, all STAs use the short AIFS, which is the same length as the IFS of IEEE 802.11. In the congestion case, only LP class STAs use the long AIFS.

Table 5-3. Simulation parameters of AEDCA.

QoS class	IFS	CW min	CWmax
High priority	SIFS+2slots	7	255
Low priority	SIFS+2slots	127	1023

Table 5-4. Simulation parameters of proposed scheme.

QoS class	IFS	CW min	CWmax
High priority	SIFS+2slots	15	1023
Low priority	SIFS+2slots (flag=0) SIFS+15slots (flag=1)	15	1023

5.5.2 Performance evaluation: Model A

Figure 5-7 plots the total throughput of the HP class and the LP class versus the number of LP class STAs. The plots reveal that HP class throughput is negatively impacted by LP class traffic when AEDCA is employed and the AP accommodates more than four LP class STAs. On the other hand, the proposal maintains HP class throughput STA regardless of the traffic of the LP class STA. This

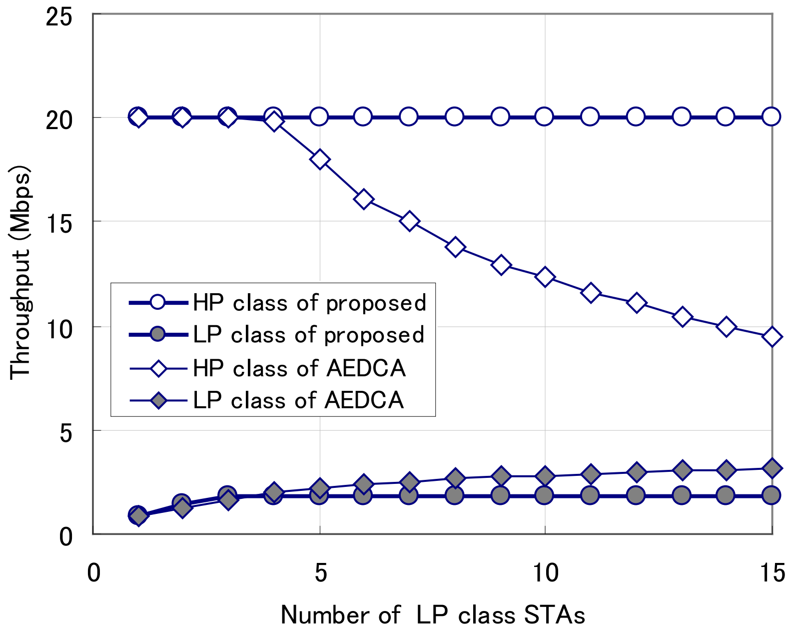


Fig. 5-7. Throughput vs. number of LP class STAs.

shows that the proposed scheme has the ability to grasp the traffic state of the radio channel correctly and to fully control the traffic of LP class STAs. This result confirms that the proposed scheme well provides QoS guarantees to STAs unlike AEDCA.

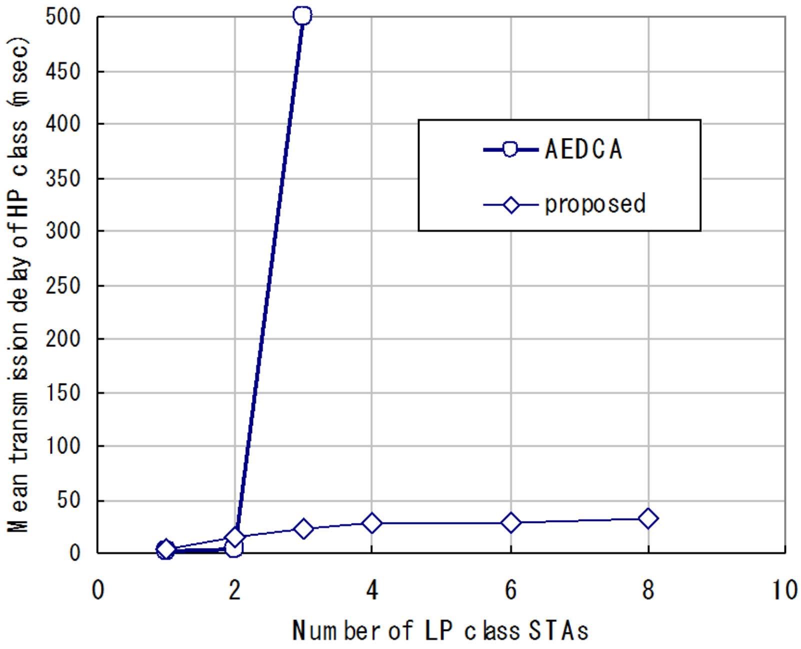


Fig. 5-8. Transmission delay of HP class vs. number of LP class STAs.

Even when there is no difference in the throughput characteristic, a difference may appear in the transmission delay characteristic of two schemes. In such senses, the transmission delay characteristic expresses the superiority of a scheme more precisely rather than the throughput characteristic. Figure 5-8 shows the transmission delay of the HP class versus the number of the LP class STAs.

In AEDCA, the transmission delay is very low when the number of LP class STAs is less than 3. However, the delay increases in haste when the number of the LP class STAs is more then 2. This is because AEDCA does not need an exchange of a control signal and there are few overheads, but its capability to inhibit the influence of the traffic of LP class is scarce. Therefore, AEDCA demonstrates good performance when the LP class STAs are a few. But its performance degrades when the LP class traffic increases. In the meantime, the transmission delay of the proposed scheme is very small over the whole, and that indicates the effectiveness of the proposal well.

5.5.3 Performance evaluation: Model B

The simulation was performed in order to show the advantage of the proposed scheme in more detail. The throughput performance of AEDCA and the proposal are plotted in Fig. 5-9 and Fig. 5-10, respectively.

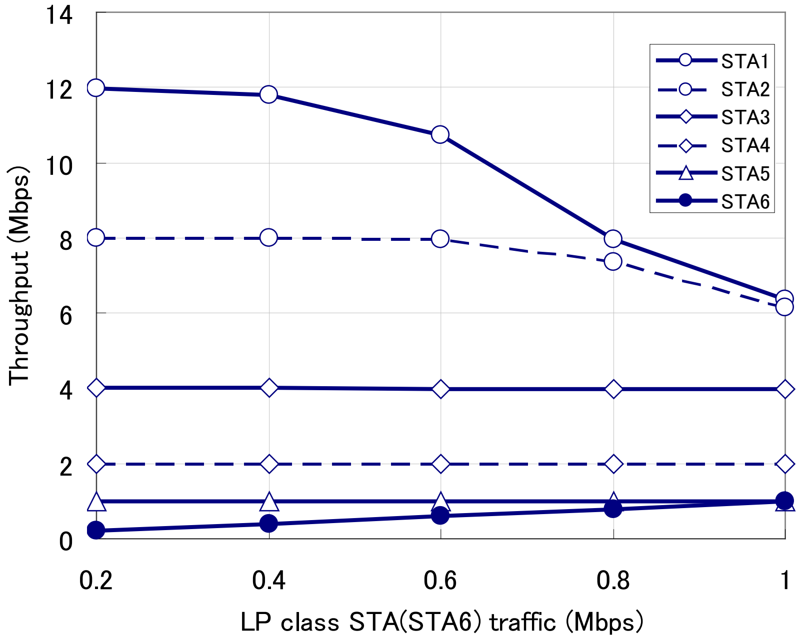


Fig. 5-9. Throughput vs. LP class STA traffic.

(AEDCA, six-STA simulation model)

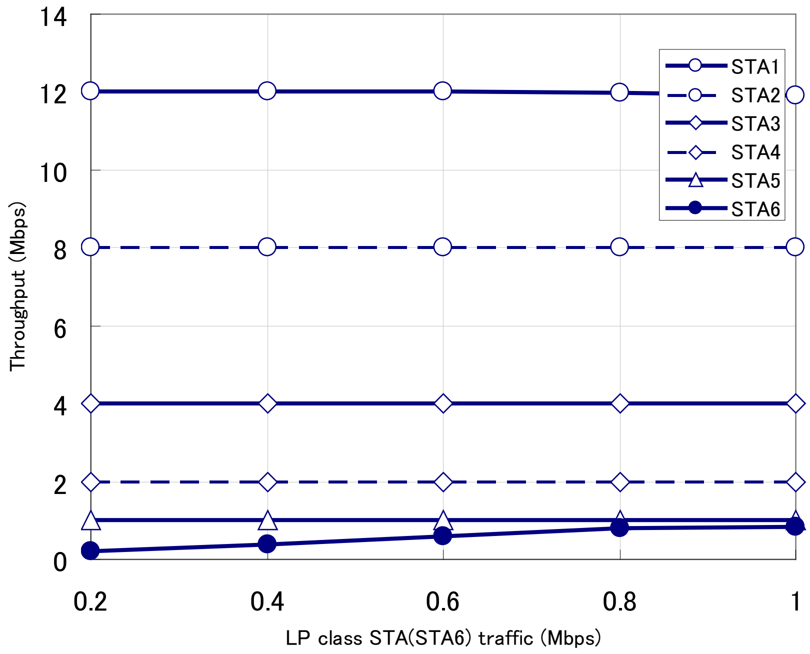


Fig. 5-10. Throughput vs. LP class STA traffic.

(Proposed scheme, six-STA simulation model)

As shown in Fig. 5-9, STA1 and STA2 (both HP STAs) experience throttling as the traffic of the LP class STA increases. Figure 5-10 shows the performance of the proposed scheme. The throughputs of all HP class STAs are kept regardless of LP class STA traffic. These results verify that the proposed scheme well provides QoS guarantees to STAs unlike AEDCA even when the AP accommodates several HP class STAs transmitting at different rates.

5.6 Conclusions

This chapter has introduced a QoS control scheme that alters the IFS length of STAs to suit their traffic circumstance in WLANs. It controls the transmission rate of data frames so as to guarantee throughput for each users. We showed that just four(or more)LP class STAs trigger priority inversion when the conventional scheme is employed and thus it can not provide QoS assurances if LP class STAs transmit large numbers of data frames at low speed. We have also showed that the proposed QoS control scheme guarantees QoS in terms of throughput regardless of the LP class traffic, and that it offers excellent performance, even in this worst-case situation.

Chapter 6
QoS Control Scheme using Dynamic Window Size Control for Wide Area Ubiquitous Wireless Networks

This chapter describes a QoS control scheme for a wide area ubiquitous wireless network which is designed to accommodate many wireless terminals (WTs), such as sensors and actuators, in a large cell area. The purpose of this chpater is to establish a QoS control scheme in a media access control (MAC) layer that can hold the transmission delay time of high priority (HP) class traffic within a predefined value regardless of how much low priority (LP) class traffic there is. Several QoS control schemes for wireless communication use have been proposed. However, in the wide area ubiquitous wireless network, an access point (AP) accommodates many WTs and the AP traffic volume often drastically changes. Therefore, conventional schemes sometimes cannot control the QoS of HP traffic. To solve this problem, we propose a QoS control scheme that calculates a suitable initial back-off window size (IWS) of random access (RA) for each QoS class by using equations derived from a Markov chain behavior model. The proposed scheme adjusts the window size so as to prevent increased transmission delay of HP traffic. The scheme's performance is clarified by computer simulation.

6.1 Overview

In recent years, there has been growing demand for machine-to-machine (M2M) networks such as Smart Grid to support systems for energy management, environment monitoring, medical management, traffic assessment, logistics control, human health care, agricultural management, and security [49]-[51]. One of the characteristics distinguishing M2M networks from other networks is that they support automatic data exchanges without human intervention. In M2M networks, small communication devices or appliances begin to transmit data at timings determined by machines, and the data destinations are also determined by machines, without human action.

Interest in ubiquitous networks and wireless communication devices has been increasing due to the popularity of cellular phone systems and wireless LANs [39], [40]. Ubiquitous wireless

The contents of this chapter are based on [71].

networks must accommodate many kinds of wireless terminals [52]-[56]. One of the most promising applications of these networks is to collect small bits of information from sensors or simple wireless terminals that are omnipresent over a very wide area [53]. In the wide area ubiquitous wireless network that is currently being developed [57] and is considered in this chapter the radio link transmission rates are very low in order to enlarge the service area, and each access point (AP) will accommodate many wireless terminals (WTs) because the sensors will send comparatively short data packets at long transmission intervals. Moreover, the network needs to provide users with a wide variety of services. Therefore, it needs to have several QoS levels.

In the wide area ubiquitous wireless network, a demand assignment (DA) scheme is employed in the medium access control (MAC) layer. The DA schemes can be classified into two types, according to the way they request radio resources. They are the random access (RA) requesting method [58], [59] and the polling requesting method [62], [63]. The wide area ubiquitous wireless network employs a DA scheme with an RA requesting method so that they can accommodate many WTs and provide high throughput performance [62]. Therefore, to ensure that our wide area wireless network can guarantee QoS, we need to develop a QoS scheme that supports RA.

In this chapter, we assume that there are only two QoS classes: high priority (HP) and low priority (LP). The transmission dely of the HP class WTs is kept within a specific value. On the other hand, the transmission delay of the LP class WTs is not guaranteed. That is, the LP class is a best effort service class. Our goal is to establish a QoS control scheme that can hold the transmission delay time of the HP class within a predefined value regardless of how much LP class traffic there is. A well-known QoS scheme that uses RA with different back-off window sizes is implemented in the enhanced distributed channel access (EDCA) of IEEE 802.11e [40], [63], [64]. In this scheme, different window sizes are assigned to all QoS classes, which are called access categories (ACs) in the IEEE802.11e standard. The scheme is easy to implement and provides good performance when it is used in home networks or small office networks, where the number of WTs is small. However, the wide area ubiquitous wireless network considered in this chapter has difficulty in using EDCA because one AP accommodates a great many WTs (e.g.,. more than 10,000 WTs) and EDCA fails to guarantee QoS when the number of WTs changes significantly. To address this problem, a QoS control scheme that controls back-off window size dynamically according to the measured transmission delay time was proposed [65]. Unfortunately, the scheme achieves good performance only if the number of WTs does not change drastically in short time.

In order to solve the problem of the conventional scheme characteristics tend to be affected by changes in the number of WTs, we have studied applying dynamic window size control to random access and proposed a basic approach [69]. In the proposal, traffic of each QoS class is observed at the AP and, according to the mean traffic of each QoS class, the AP calculates the appropriate back-off window sizes by using a Markov behavior model and passes them to the WTs.

In this paper, we clarify the problem of the conventional scheme and explain a new QoS control scheme that uses a dynamic window size control mechanism in detail. In particular, random access operations that avoid collisions effectively and equations that calculate appropriate window size are described precisely. We also provide simulation results that demonstrate the general effectiveness of our proposal.

The remainder of this chapter consists of five sections. Section 6.2 describes the structure of the wide area ubiquitous wireless network considered in this chapter. Section 6.3 clarifies the problem of the conventional scheme. Our proposed scheme's operating principle is given in Section 6.4 and its performance is evaluated in Section 6.5. Section 6.6 concludes the section with a summary of key points.

6.2 Wide area ubiquitous wireless access system

Figure 6-1 shows the configuration of the wide area wireless network considered in this chapter and Table 6-1 lists the system parameters [55]. In the network, the APs, which are connected to a wired network, accommodate many WTs, which are scattered over a very wide area that has a radius of several kilometers. We assume the system uses a time division multiple access (TDMA)/time division duplex (TDD) scheme in the MAC layers [55], [56].

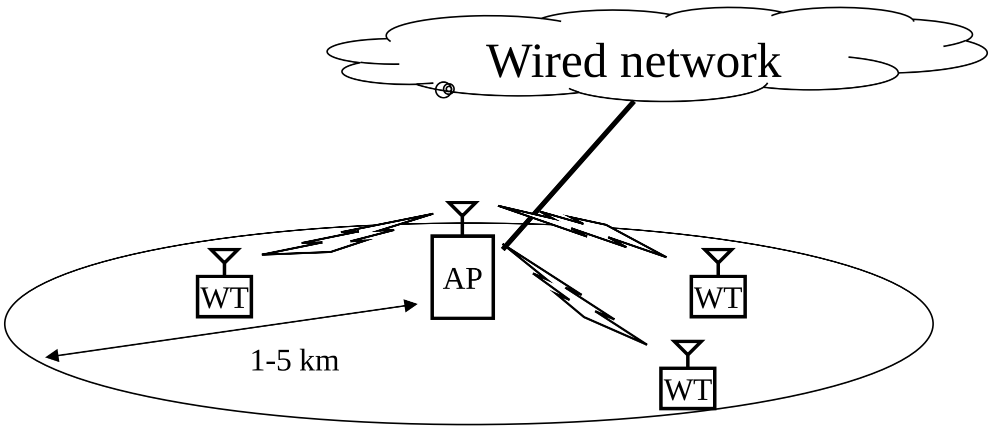


Fig. 6-1. Configuration of wide area wireless network.

Table 6-1. System parameters.

Parameters	Value
Access method	TDMA/TDD
MAC frame length	1 sec
Transmission rate	9600 bps
Error collection	ARQ (stop and wait)
Resource assignment	Dynamic Slot Assignment
Collision avoidance	Exponential back-off
Number of QoS classes	2
Priority control	RA window control priority packet scheduling
Number of accommodated WTs	~ 65,535

6.2.1 MAC frame structure

Figure 6-2 shows the assumed MAC frame structure and Table 6-2 lists its logical channels. In this chapter, we assume that a dynamic slot assignment (DSA) scheme is employed in order to use radio resources efficiently by assigning resources to meet WT needs. A MAC frame consists of a broadcast channel (BCCH), a frame control channel (FCCH), an RA feed-back channel (RFCH), a user data channel (UDCH), a logical control channel (LCCH), and an RA channel (RACH). BCCHs are used to provide data about the AP attributes and the data is sent at the top of each MAC frame. We assume that the MAC frames have a constant length, so the BCCHs send the information at regular intervals in time. FCCHs indicate the MAC frame structure (i.e., the position and the length of other channels following the FCCH). DSA is realized by changing the FCCH contents. RFCHs are used to send the information associated with RA (i.e., the RA results in the previous MAC frame, the back-off window size of each QoS class, the position of RA slots, and the number of RA slots in the current MAC frame). UDCHs are used to send user data and LCCHs are used to send MAC control information (i.e., automatic repeat requests (ARQs), acknowledgements (ACKs), and so on). The RA area, which consists of one or more RA slots, is arranged in the later part of the uplink. The position of the RA area may change for every MAC frame, so RFCHs notify to WTs as mentioned above.

6.2.2 Transmission sequence

In DSA, radio resources are assigned at the request of each user. The AP easily finds the user requests in the downlink without any communication through the radio link. However, the AP must find the user requests in the uplink by using one of two methods: one sends requests by polling and

Table 6-2. Logical channels.

Channels	Uses
BCCH	Broadcasts the common system information at the top of MAC frames
FCCH	Broadcasts the frame structure of MAC frame
RFCH	Informs terminals of the information concerned with random access
RBCH	Broadcasts the link control information to all users
LCCH	Used in transmission of MAC control information
UDCH	Used in transmission of user data
RACH	Used in transmission of random access

the other sends them by RA. The RA request method is suitable when the number of accommodated terminals is large because radio access occurs only when a request is generated. Accordingly, we assume that requests are sent by RA.

Figure 6-3 shows the user data transmission sequence using RA and DA. In this scheme, the AP sends a BCCH protocol data unit (PDU), an FCCH PDU, and an RFCH PDU in that order at the top of each MAC frame. Whenever a WT has user data to transmit, it automatically performs

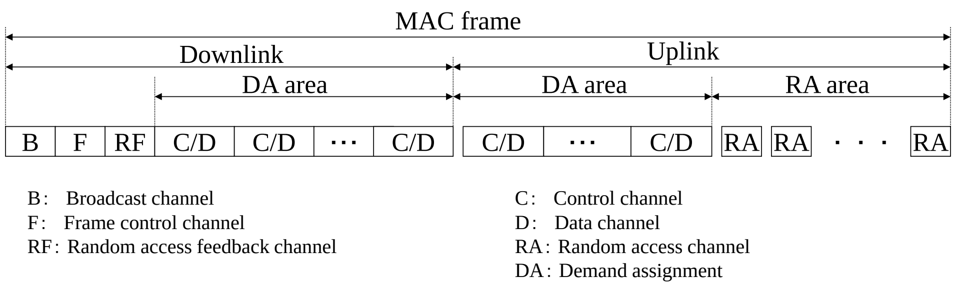


Fig. 6-2. MAC frame structure.

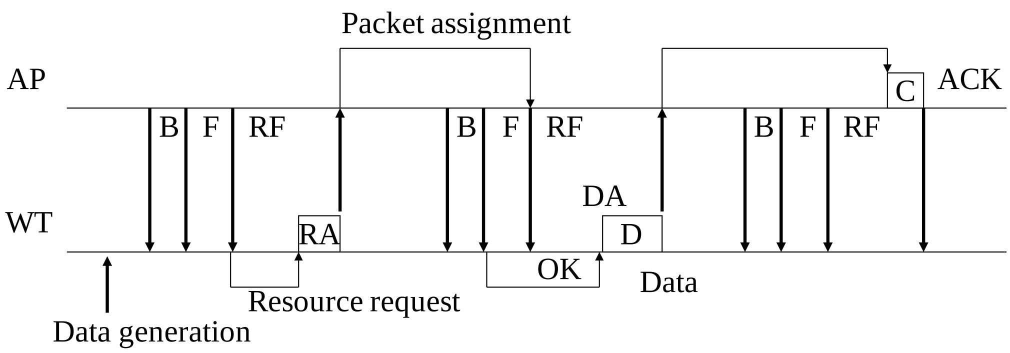


Fig. 6-3. Data transmission sequence.

back-off before the transmission to avoid collision with other WT's RACHs. The WT determines the start position of the back-off procedure according to the data generation timing. When the user data generation occurs on the outside of the RA area, the back-off procedure starts at the top of the RA area as shown in Fig. 6-4 (a). When it occurs within the RA area, the procedure starts at the RA slot immediately after the data generation as shown in Fig. 6-4 (b). Unlike the MAC protocol of the IEEE802.11, the assumed scheme performs the back-off procedure even when sufficient time has passed since the last transmission. This is because the collision probability of the first RA slot in the RA area of each MAC frame may significantly increase if the back-off procedure is not performed. Figure 6-5 shows the transmission sequence without back-off. From this figure, it is obvious that collision takes place when two or more WTs have user data on the outside of the RA area if the WTs do not perform back-off.

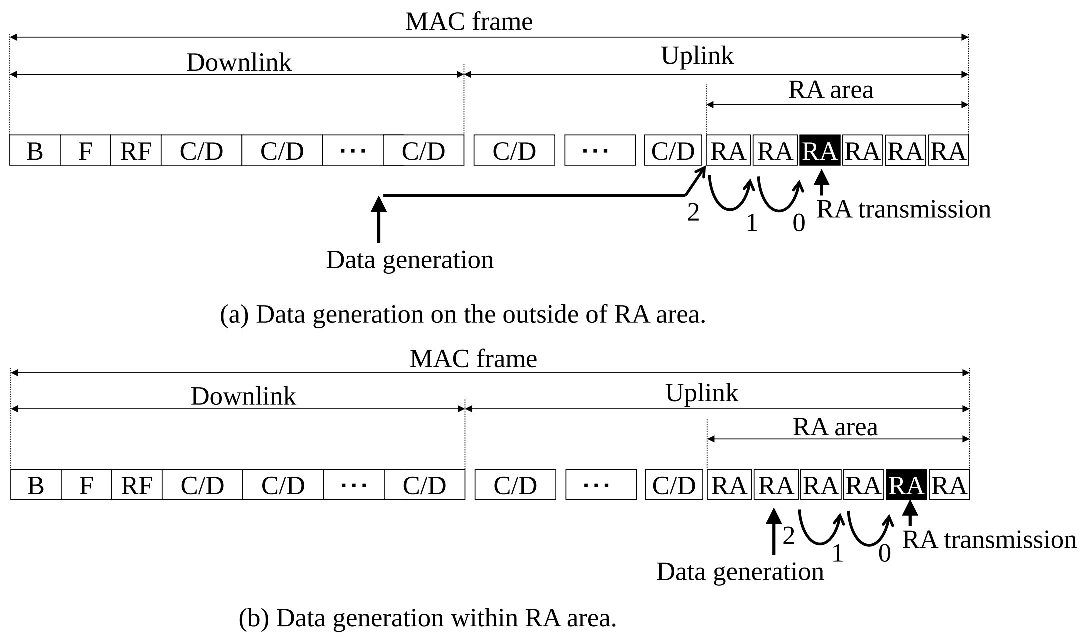


Fig.6-4. RA back-off procedure.

The number of back-off slots in the RA is randomly selected by the WT as an integer value within the range of zero to window size (WS). If the number of back-off slots the WT selected is larger than the number of RA slots in the RA area of the current MAC frame, the former decreases as much as the latter and then the WT performs back-off in the next MAC frame (Fig. 6-6). This back-off procedure continues until the number of back-off slots reaches zero. At the first RA attempt, the WS is set to the value that is informed by the AP as the initial back-off window size (IWS) through the RFCH. After performing back-off to avoid collision, the WT sends an RACH to request the assignment of UDCH in which to transmit its user data.

After transmitting an RACH, the WT receives an RFCH again in the next MAC frame. If the RA is successful in the previous MAC frame, the AP sends back an ACK to the WT through the RFCH. Otherwise, it sends nothing. When the WT fails to receive an ACK, it doubles the WS and recalculates the number of back-off slots using the latest WS. After back-off, the WT tries to resend an RACH. The attempts to resend it continue until it is successfully resent or the sending count reaches the predefined retry limit.

After receiving an ACK from the AP, the WT waits for the UDCH assignment. When the AP receives a UDCH assignment request from a WT, it assigns UDCH transmission area to the WT in the later MAC frame and informs the WT of the assignment area information through the FCCH. When the WT is assigned a UDCH, it transmits the UDCH, which now contains the user data. When the AP receives the UDCH, it sends back an LCCH PDU with an ARQ-ACK if it receives the UDCH without any error. Otherwise, it sends back an LCCH PDU with an ARQ negative acknowledgement (ARQ-NAK), which informs the WT that a UDCH transmission failure has occurred.

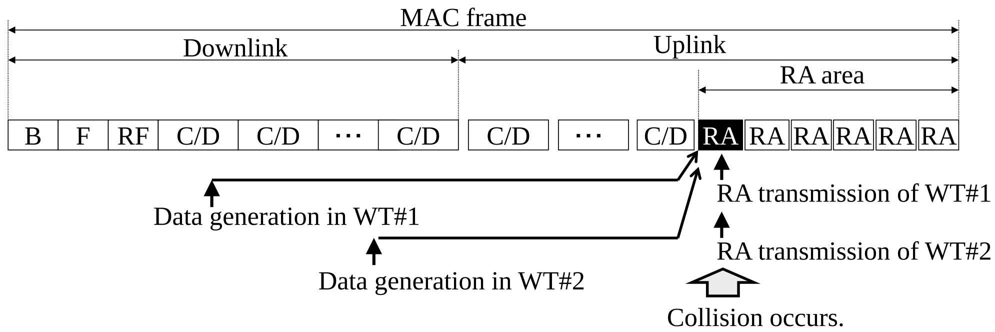


Fig. 6-5. RA transmission without back-off.

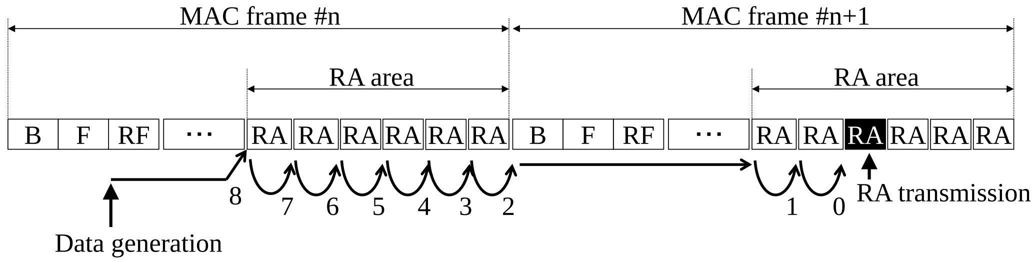


Fig. 6-6. Back-off in two MAC frame.

6.3. Conventional scheme and its problem

6.3.1 Conventional scheme

The conventional QoS control scheme uses dynamic back-off window size control to set the HP class's transmission delay (Fig. 6-7) [65]. In this scheme, the AP measures the HP class's mean delay and changes the LP class's IWS every predefined interval. The AP compares the observed time with the predefined threshold, which is the target delay that must be satisfied for HP class service. If the AP detects that the observed time exceeds the threshold, it increases the LP class's IWS value; when the LP WTs receive the new value they lower their transmission rate. On the other hand, the AP lowers the LP class's IWS value if the observed time is less than the threshold.

Figure 6-8 shows the flow chart of the scheme's back-off window size control mechanism. The scheme uses three predefined thresholds: the target delay of the HP class that is to be satisfied, the upper threshold, and the lower threshold. The second and third thresholds are used to select either window size change mode, linear change mode, or exponential change mode. When the mean delay is between the upper and lower thresholds, the AP changes the LP IWS linearly. Otherwise, it changes it exponentially. That is, if the mean delay is larger than or equal to the upper threshold, the AP doubles the LP IWS every predefined interval and uses twice the value of the HP persistent factor (PF) for the LP one, which drastically suppresses the LP traffic. If the mean delay is less than the upper threshold and more than the target delay, the AP increases the LP IWS, which throttles the

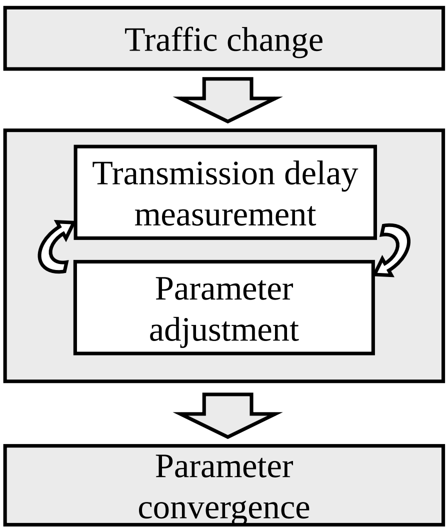


Fig. 6-7. Diagram of conventional scheme.

LP traffic gradually. If the mean delay is more than the lower threshold and equal to or under the target delay, the AP decreases the LP IWS by 2. If the calculated mean delay is less than the lower threshold, the AP halves the LP IWS.

6.3.2 Problem of conventional scheme

When traffic changes comparatively slowly, the conventional scheme can adjust the window size to a suitable value according to the traffic the LP class WTs generate and maintain the HP class's QoS level. On the other hand, the scheme can't keep the QoS level when the traffic rapidly changes.

According to the results obtained in comparing the transmission delay with a pre-defined threshold, the conventional scheme determines whether to increase or decrease the WS of LP class. However, it cannot determine the degree to which it must increase or decrease the WS and only changes it by a pre-defined amount. It is expected that this operation will change the WS value and

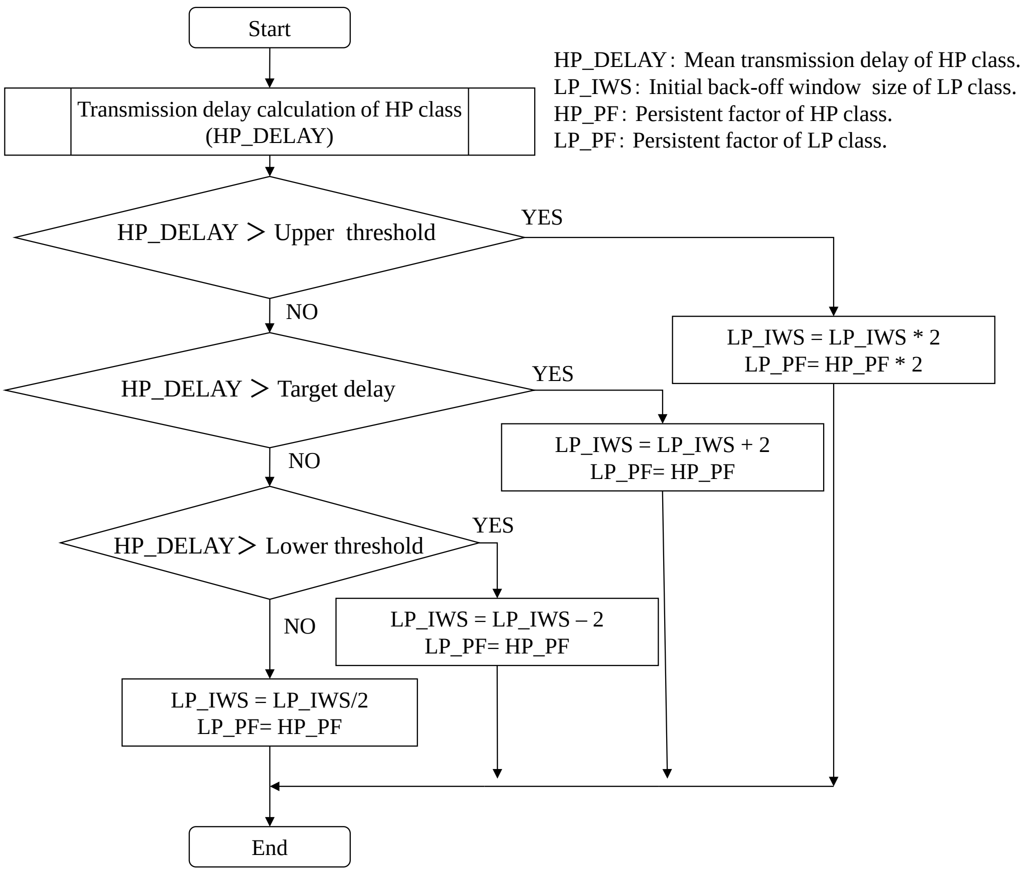


Fig. 6-8. Flow chart of conventional window control.

bring it closer to the optimum value, but the value does not become the optimum value at once. In order to converge the WS to the optimum value, it is necessary to repeat the operation many times. Therefore, a control delay occurs and causes large transmission delay of the HP class when the traffic volume changes substantially in a short time.

6.4 Proposed window size control

To diminish the increase in the HP class's transmission delay as much as possible when traffic drastically changes and provide QoS in the wide area ubiquitous wireless network, we propose a new QoS control scheme that uses equations derived from a Markov chain behavior model to calculate the appropriate IWS and controls the LP class traffic precisely to satisfy the HP class's transmission delay. Moreover, the proposed scheme maximizes the LP class throughput on the condition that the HP class's transmission delay is satisfied. Figure 6-9 shows a diagram of the proposed scheme. In the scheme, the AP observes the traffic and calculates the data generation probability of the HP and LP class WTs from each QoS class's mean traffic. In the parameter calculation, the scheme obtains an appropriate IWS using the data generation probability and the RA failure probability.

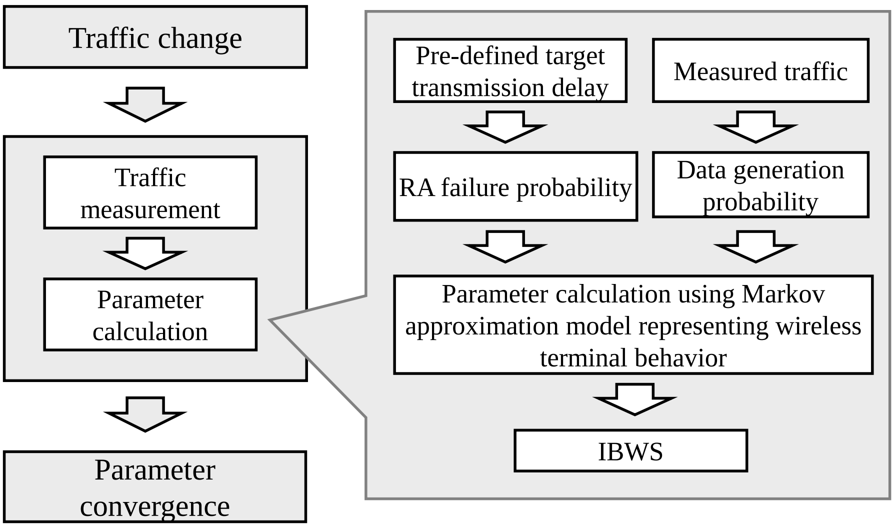


Fig. 6-9. Diagram of proposed scheme.

6.4.1 Markov chain behavior model

The transmission delay can be divided into two portions: the RA delay time and the radio resource assignment (packet assignment) delay. The former is, literally, the time taken by RA. The latter is the period from receiving a radio resource assignment request until the resource is assigned. In the HP class, the latter is much smaller than the former because HP class assignment takes precedence over LP class assignment and its assignment time is shorter than the MAC frame length. Therefore, to control the transmission delay, it is necessary to control the RA delay.

In the proposed scheme, the RA delay is calculated by using equations derived from Markov chain behavior models of RA. Several Markov chain behavior models, which represent RA behavior, have been proposed [66]-[68]. In [68], a non-saturated (non-greedy) model was proposed and its validity was confirmed for IEEE 802.11 wireless LAN. In this section, a Markov model that is a simplified variant of the non-saturated model in [68] is used to derive the equations that represent the relation between the offered load probability p_g, the transmission failure probability p, and the transmission probability τ. The Markov behavior model is depicted in Fig. 6-10. Since this is a steady-state model, each WT's behavior in the same class is expressed by the same model. Thus, a model exists for each class, not for each WT. The offered load probability p_g is the data frame arrival rate at which the MAC layer receives data frames from the upper layer in an effective RA slot for a given WT. This probability is easily predicted by calculating the reciprocal of the WT's data receiving interval at the AP, which is measured by the traffic observer in the AP. When the probability is given, the Markov chain behavior model determines an appropriate IWS for the WT.

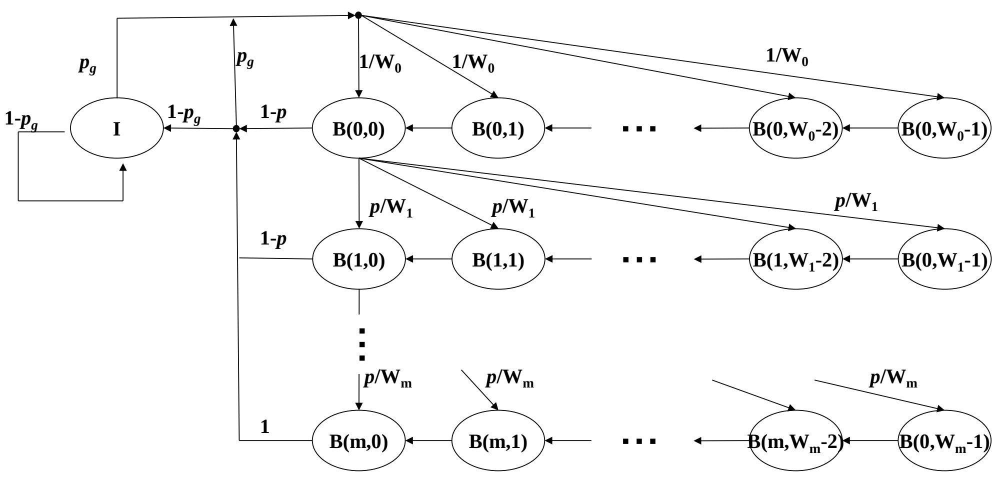


Fig. 6-10. Markov chain behavior model.

The relations between p_g, p, τ, and IWS, which is written W_0 in Fig. 6-10, are expressed as

$$\frac{1-p_{gh}}{p_{gh}}+\frac{W_{0h}}{2}\times\frac{1-(2p_h)^{m+1}}{1-2p_h}+\frac{1}{2}\times\frac{1-p_h{}^{m+1}}{1-p_h}=\frac{1}{\tau_h}\times\frac{1-p_h{}^{m+1}}{1-p_h} \quad , \tag{6.1}$$

$$\frac{1-p_{gl}}{p_{gl}}+\frac{W_{0l}}{2}\times\frac{1-(2p_l)^{m+1}}{1-2p_l}+\frac{1}{2}\times\frac{1-p_l{}^{m+1}}{1-p_l}=\frac{1}{\tau_l}\times\frac{1-p_l{}^{m+1}}{1-p_l} \quad . \tag{6.2}$$

where p_h, p_{gh}, τ_h, and W_{0h} are the RA failure probability, data generation probability, transmission probability, and IWS of the HP class, and p_l, p_{gl}, τ_l, and W_{0l} are those for the LP class. AS described above an equation of the relationship exists for each class.

These equations are derived from the Markov chain behavior models of each QoS class. Appendix A shows in detail how the equations are derived. In the equations, m represents the pre-defined retry limit. The relationship between p and τ is expressed as

$$p_h = 1-(1-\tau_l)^{n_l}\times(1-\tau_h)^{n_h-1} \quad , \tag{6.3}$$

$$p_l = 1-(1-\tau_l)^{n_l-1}\times(1-\tau_h)^{n_h} \quad , \tag{6.4}$$

where n_h is the number of HP class WTs and n_l is that of LP class WTs. From the relation between the transmission delay and the initial back-off window size W_{0h} and the transmission failure probability p_h of the HP class, the HP class’ s mean delay is expressed as

$$Delay = T_{erss}\times\frac{W_{0h}}{2}\times\left(2\times\frac{1-(2p_h)^{m+1}}{1-2p_h}\times\frac{1-p_h}{1-p_h{}^{m+1}}-1\right) \quad , \tag{6.5}$$

where T_{erss} represents the effective RA slot length. This slot length means the averaged elapsed time per back-off slot. Figure 6-11 shows the relation between the MAC frame, the RA slots, and the effective RA slot length. The latter is the time yielded by dividing the MAC frame length by the number of RA slots in the MAC frame. The derivation details are shown in Appendix B. When the target value of the HP class transmission delay is expressed as *Target_delay*, the initial back-off window size W_{0h} and the transmission failure probability p_h of the HP class should satisfy the following condition,

$$Target\_delay = T_{erss} \times \frac{W_{0h}}{2} \times \left(2 \times \frac{1-(2p_h)^{m+1}}{1-2p_h} \times \frac{1-p_h}{1-{p_h}^{m+1}} - 1 \right). \tag{6.6}$$

While guaranteeing the HP class transmission delay, the proposed scheme maximizes the LP class throughput. The proposed scheme changes transmission probabilities τ_h, τ_l, to suitable values by controlling IWSs and then maximizes the LP throughput while the transmission delay of the HP class is guaranteed. The maximizing conditions give the following equation

$$\frac{\partial S}{\partial \tau_l} = \frac{\partial \left(n_l \cdot \tau_l \cdot (1-\tau_l)^{n_l - 1} \cdot (1-\tau_h)^{n_h} \right)}{\partial \tau_l} = 0 \quad , \tag{6.7}$$

where S is the LP class throughput. By calculating partial differential of Eq. (6.7), we can obtain the relationship between τ_l and n_l as follows

$$\tau_l = \frac{1}{n_l} \quad . \tag{6.8}$$

In the proposed scheme, the transmission delay of the HP class is kept lower than or equal to the target delay. Therefore, from the above descriptions, there are six constraint equations (Eq. (6.1)~(6.4), (6.6) and (6.8)) in order to calculate suitable initial back-off window sizes. These six equations include thirteen variables: p_{gh}, p_{gl}, p_h, p_l, τ_h, τ_l, W_{0h}, W_{0l}, n_h, n_l, T_{erss}, m, and *Target_delay*. The variable *Target_delay* is set to a target delay of the HP class. Three of these variables, T_{erss}, m, and *Target_delay*, are system parameters and four of them, p_{gh}, p_{gl}, n_h, and n_l, have values that the AP measures and calculates. Thus, there are six remaining variables and six equations. Therefore, the values of the variables are determined by solving the six equations.

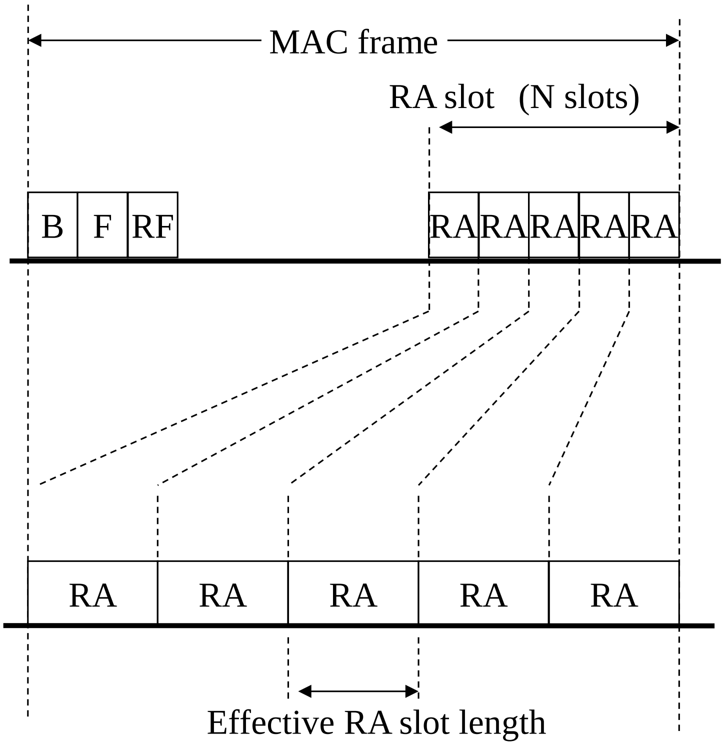


Fig. 6-11. Effective random access slot length.

The numbers of active WTs, n_h, n_l, are needed to calculate access parameters. In the proposal scheme, the numbers of each class WTs are computed by counting the number of the identifiers which identify a terminal and are send in the random access channel PDU.

We can obtain appropriate values of the initial back-off window sizes that satisfy the transmission delay requirement by solving the above six equations. However, the calculated values of access parameters may not be appropriate under certain conditions. If one or more calculated values are not in the appropriate range, the proposed scheme considers them as inadequate. When the values calculated by the six equations cannot be used, it is necessary to throttle the LP class throughput to zero and to newly calculate appropriate values by solving the three HP class related equations (Eq. (6.1), (6.3), and (6.6)) under the no LP class throughput condition, where the transmission probability of the LP class, τ_l, is zero.

The proposed scheme assumes that all WTs are in steady state when the equations are derived because it is difficult to model WT's behavior and to derive equations in transient state. The proposed scheme supposes the number of WTs does not change during one MAC frame, solves the equations and obtains parameter values. Therefore, a difference may arise between the actual and theoretical delay in transient state. However, as described later, the proposed scheme accomplishes better performance than the conventional scheme even if the difference occurs.

6.4.2 Dynamic IWS control

In the proposed scheme, the AP calculates an appropriate IWS for each QoS class as mentioned above. The AP informs WTs of these IWSs by sending RFCHs periodically; they include the IWS values of each QoS class. Every time the WTs receive an RFCH, they update the IWS value they hold. When a WT begins to transmit or retransmit a data frame, it calculates the number of back-off slots according to its IWS.

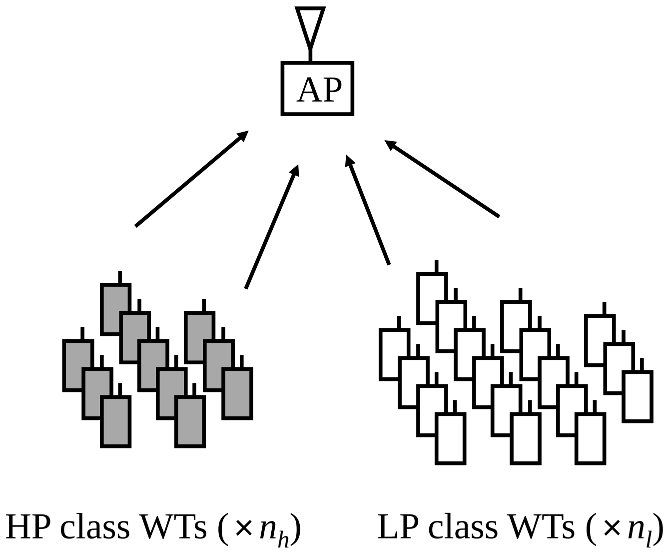


Fig. 6-12. Simulation model.

6.5 Performance evaluation

6.5.1 Simulation model

We conducted computer simulations to clarify the conventional scheme's problem and the proposed scheme's effectiveness. Figure 6-12 shows the simulation model and Table 6-3 lists the simulation parameters. In this model, the AP accommodates many HP class WTs and LP class WTs. Both of them use the same radio frequency channel to send data frames. The numbers of WTs were set according to the scenarios listed in Table 6-4. Since the purpose of these scenarios is to reveal QoS control scheme performance when the LP class traffic volume changes, the number of LP class active WTs increases during a simulation. There are two kinds of WT states, active and sleep states.

In the active state, a WT generates traffic at predefined rate. In the sleep state, a WT does not send any data.

Seven simulations were performed for each scenario. In the simulations, the number of WTs was the same and all the HP class WTs are active for the same scenario. The number of initial LP class active WTs was also the same. However, the final number of LP class active WTs was different in each simulation. For example, in the base scenario (scenario 1), the number of HP class WTs (1,000) did not change and all of them were active during the simulation. The initial number of LP class active WTs was set to be 3,000 while the final number became 3,000, 4,000, 5,000, 7,000, 8,000, or 9,000. In each scenario, all WTs in the active state generated the same traffic volume but the total LP class traffic volume varied with the number of LP class active WTs. The reason for having assumed these simulation models is because the traffic of WTs may increase simultaneously if ignited by broad-based disasters such as an earthquake.

Table 6-3. Simulation parameters.

Parameters	Value
Transmission rate	9600 bps
MAC frame length	9600 bits (=1sec)
QoS class	2 QoS classes HP class and LP class
Distribution of traffic interval	Poisson
User data size	40 bytes
Target delay of HP class.	60 sec

In these simulations, the target mean delay of the HP class to be satisfied was set to sixty seconds as a required value. In gas (or other utility) metering systems, a meter mainly has two functions. One is to send collected information about the amounts of utilities consumed in individual homes to center servers. The other is to open or close its corks according to remote commands from the center servers. In remote cork control, several command transmissions are performed in one control sequence, and success or failure of the control sequence is determined by whether the sequence is finished before a timer expires. Since the usual timer value is several minutes, the time available for transmitting one command transmission is about one minute. Therefore, we assume the target delay of the HP class to be satisfied is one minute.

Table 6-5 shows the conventional scheme's parameters; the parameter values are defined in consideration of the target mean delay.

6.5.2 Conventional Scheme's Performance

Figure 6-13 plots the conventional scheme's transmission delay performance in the base scenario (scenario 1). It is clear the scheme has good performance and keeps the HP class's mean transmission delay at sixty seconds before the number of LP class WTs changes. Unfortunately, the delay increases rapidly after the number changes to 9,000 at 10 hr. simulation time. This is because the scheme fails to control the LP class's IWS at an appropriate value due to the controlling delay its algorithm imposes. The main delay factors are the time required to measure the HP class traffic's transmission delay and that required for the IWS to settle on a suitable value.

Table 6-4. Simulation scenarios.

Scenarios	Parameters	Value
1(base)	Number of HP class WTs	1000
	Number of LP class WTs	9000
	Initial number of LP class active WTs	3000
	Final number of LP class active WTs	3000,4000,5000, 6000,7000,8000,9000
	WT traffic	0.96 bps/WT
2	Number of HP class WTs	500
	Number of LP class WTs	4500
	Initial number of LP class active WTs	1500
	Final number of LP class active WTs	1500,2000,2500, 3000,3500,4000,4500
	WT traffic	1.92 bps/WT
3	Number of HP class WTs	1500
	Number of LP class WTs	13500
	Initial number of LP class active WTs	4500
	Final number of LP class active WTs	4500,6000,7500, 9000,10500,12000,13500
	WT traffic	0.64 bps/WT
4	Number of HP class WTs	2000
	Number of LP class WTs	18000
	Initial number of LP class active WTs	6000
	Final number of LP class active WTs	6000,8000,10000, 12000,1400016000,18000
	WT traffic	0. 48 bps/WT

Table 6-5. Conventional scheme parameters.

Parameters	Value
Upper threshold	66 sec
Lower threshold	54 sec

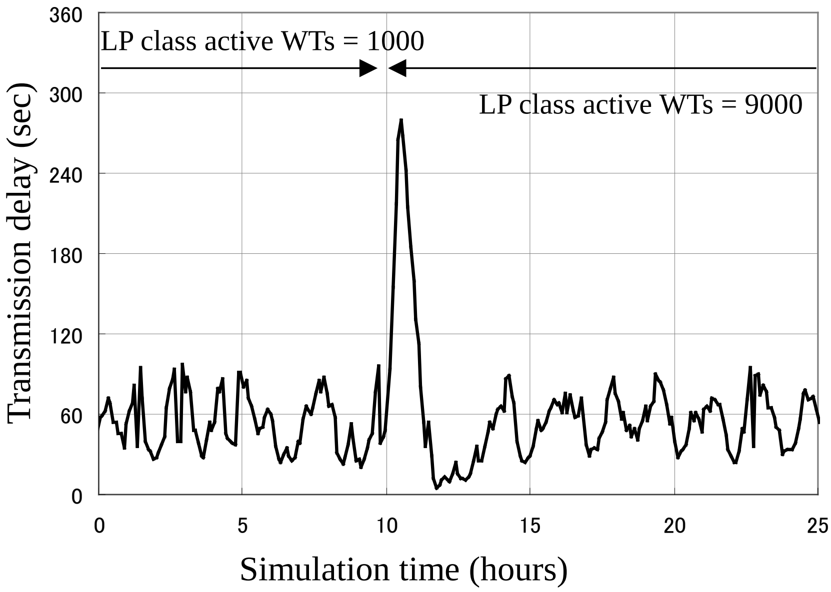


Fig. 6-13. Transmission delay time performance of conventional scheme.

These results make it obvious the conventional scheme has a critical problem. The problem is that the HP class traffic's transmission delay can become excessive when the LP traffic increases.

6.5.3 Proposed scheme's performance

Figure 6-14 plots our proposed scheme's transmission delay performance. It is clear our scheme also shows good performance when the number of LP class WTs doesn't change. Furthermore, the increase in delay when the number of LP class WTs does increase is much smaller than in the conventional scheme. This is because our scheme can very rapidly set suitable values for the IWSs. The proposed scheme determines how much initial back-off window sizes should change, while the conventional scheme determines only whether to increase or decrease the sizes. Consequently, the time needed to change the sizes to suitable value is shorter than that of the conventional scheme. The proposed scheme can make the difference between the HP class delay and target delay smaller even in the transient state where the number of the LP class active WTs changes.

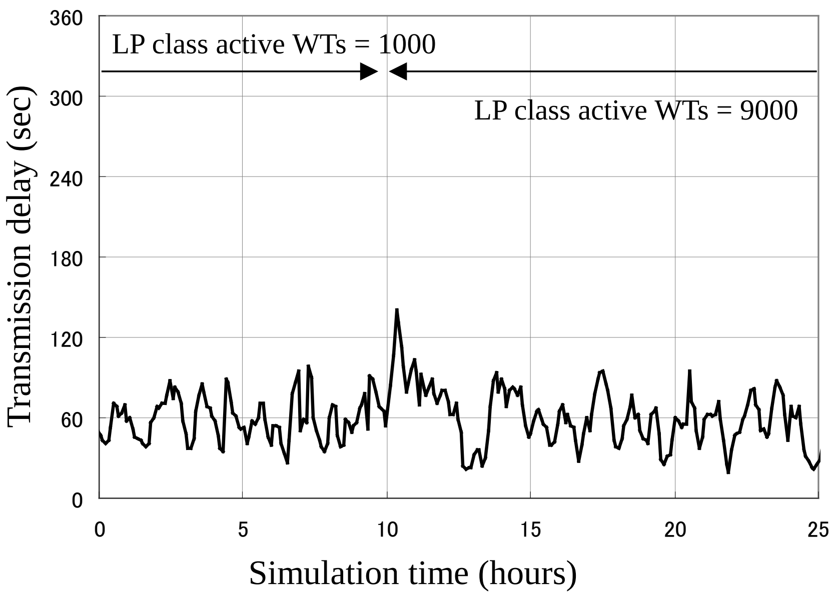


Fig. 6-14. Transmission delay time performance of proposed scheme.

6.5.4 Maximum transmission delay

We define the maximum transmission delay as the largest delay that occurs after the number of LP class WTs changes. The LP class WTs' increase ratio is the ratio of the number of LP class WTs at ten hours after the simulation starts to the initial number. That is, the ratio is denoted by Ni/Ns, where Ns is the number of LP class WTs at the simulation start time and Ni is the number ten hours later. Figure 6-15 shows both schemes' maximum transmission delay; that of the proposed scheme is almost constant regardless of the increase ratio of the number of LP class WTs, while that of the conventional scheme gradually increases with the increase ratio. This indicates the proposed scheme can make the IWSs rapidly converge on adequate values regardless of the number of LP class WTs.

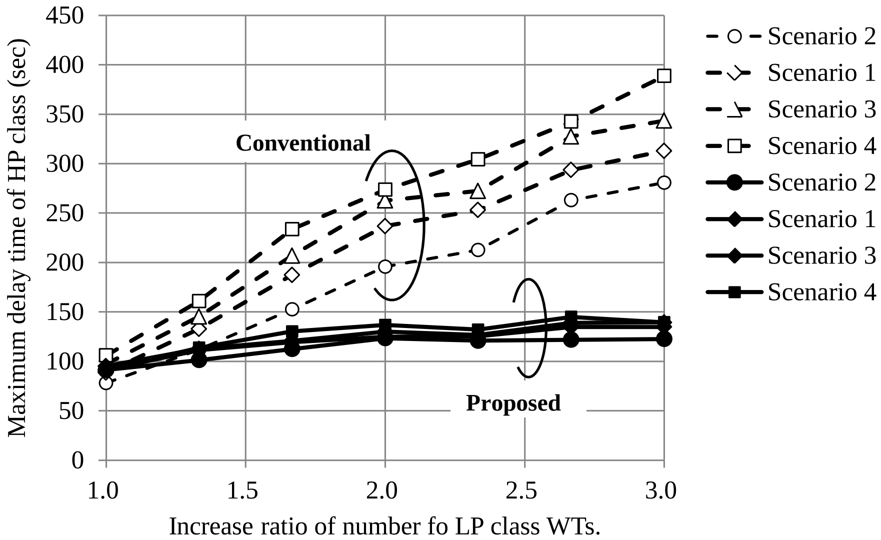


Fig. 6-15. Maximum transmission delay performance.

6.5.5 Threshold over time

In order to evaluate the length of time over which the HP class QoS is not satisfied, we define the threshold exceeded duration as the time over which the HP class's transmission delay exceeds triple σ, where σ is the standard deviation of the HP class's transmission delay distribution for a constant number of LP class WTs. Figure 6-16 plots both schemes' threshold exceeded duration performance. The proposed scheme is clearly superior in that it holds the threshold exceeded duration to zero for most cases examined.

To maintain QoS for a large increase in the number of WTs, the IWSs must be changed drastically. Since the conventional scheme can only change them gradually, it needs to change them repeatedly. In contrast, our proposed scheme can adjust them to the proper values almost at once. These results make it clear that the proposed scheme controls the IWSs properly and can provide the QoS guarantees even if the number of LP class WTs increases dynamically.

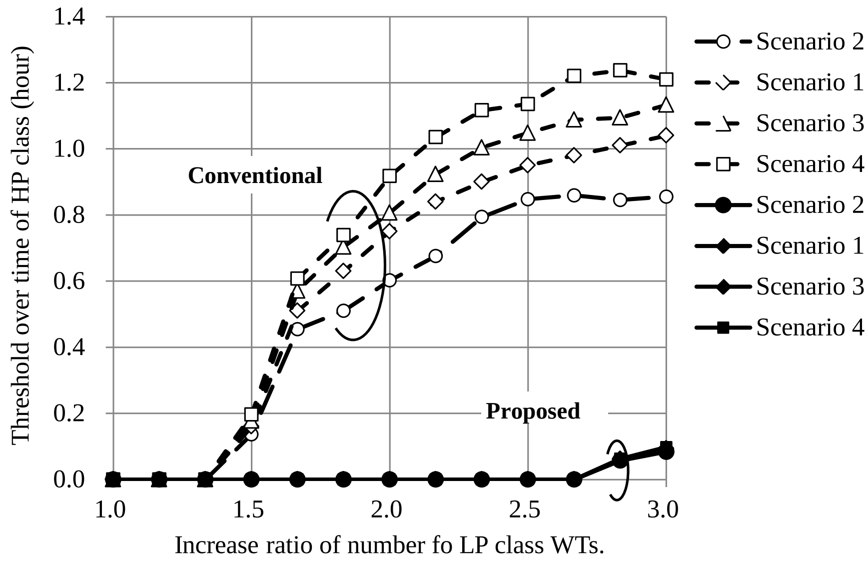


Fig. 6-16. Threshold exceeded duration of HP class.

6.6 Conclusion

This chapter proposes a QoS control scheme for a wide area ubiquitous wireless network; it keeps the transmission delay of high priority (HP) class traffic within a specified limit by using dynamic back-off window size control of each QoS class. The proposed scheme calculates suitable values of initial back-off window sizes (IWSs) using an equation derived from the Markov chain behavior models of wireless terminals (WTs) and the measured traffic. In this section, we showed that the conventional scheme cannot provide QoS assurances when the number of low priority (LP) class WTs increases drastically; however, the proposed scheme exhibits excellent performance even in this worst-case situation.

Appendix A
Derivations of Eqs. (6.1) and (6.2)

In the Markov chain behavior model shown in Fig.6-10, $B(i,j)$ is a probability that a given WT is in the back off stage i and the number of the WT's remaining back off slot is j. From the relation between $B(i,j)$ and $B(i\text{-}1,0)$ where $i \in (1, m)$ in the model, we easily obtain the following equation

$$B(i,j) = B(0,0)\frac{W_i - j}{W_i}p^i, \tag{A.1}$$

where the p is the transmission failure probability in RA.
A WT transits the state I only from the state I and the state $B(i,0)$ where $i \in (0, m)$. This yields

$$p_{idle} = (1-p_g)p_{idle} + (1-p_g)\left\{(1-p)\sum_{i=0}^{m-1}B(i,0) + B(m,0)\right\}, \tag{A.2}$$

where the p_{idle} is a probability that a given WT is in the state I of Fig. 7.
From Eq. (A1), we obtain $B(i,0) = B(0,0)p^i$. Hence, Eq. (A2) rewrites as

$$\begin{aligned} p_{idle} &= (1-p_g)p_{idle} + (1-p_g)\left\{(1-p)\sum_{i=0}^{m-1}B(0,0)p^i + B(m,0)\right\} \\ &= (1-p_g)p_{idle} + (1-p_g)B(0,0) \\ p_{idle} &= \frac{1-p_g}{p_g}B(0,0) \quad . \end{aligned} \tag{A.3}$$

From the normalization condition, the summation of the total probability is one. Therefore, we obtain

$$p_{idle} + \sum_{i=0}^{m-1}\sum_{j=0}^{W_i-1} B(i,j) = 1\,. \tag{A.4}$$

Using Eq. (A1) and (A3), we can rewrite Eq. (A4) as

$$B(0,0)\left[\frac{1-p_g}{P_g}+\frac{W_0+1}{2}+\frac{1}{2}\left\{W_0\cdot 2p\frac{1-(2p)^m}{1-2p}+p\frac{1-p^m}{1-p}\right\}\right]=1\,. \tag{A.5}$$

Making use of the fact that data frame transmission occurs only when a WT is in the state of $B(i,0)$ where $i \in (0, m)$, τ is expressed as

$$\tau=\sum_{i=0}^{m}B(i,0)$$

$$\tau=\frac{1-p^{m+1}}{1-p}B(0,0)\ . \qquad \text{(A.6).}$$

Finally, we obtain the relation between p_g, p, τ, and W_0 as follows by substituting Eq. (A6) for Eq. (A5)

$$\frac{1-p_g}{P_g}+\frac{W_0}{2}\times\frac{1-(2p)^{m+1}}{1-2p}+\frac{1}{2}\times\frac{1-p^{m+1}}{1-p}=\frac{1}{\tau}\times\frac{1-p^{m+1}}{1-p}\,. \tag{A.7}$$

Appendix B
Derivations of Eqs. (6.5)

In the case that the *i*-th transmission is succeed, the mean transmission delay time $D(i)$ is expressed as follows,

$$\begin{aligned} D(i) &= T_{erss} \cdot \frac{1}{2} \cdot \sum_{j=1}^{i} W_{j-1} = T_{erss} \cdot \frac{1}{2} \cdot \sum_{j=1}^{i} W_0 \cdot 2^{j-1} \\ &= T_{erss} \cdot \frac{(2^i - 1)}{2} \cdot W_0 \quad . \end{aligned} \tag{B.1}$$

The occurrence probability of successful *i*-th transmission $P(i)$ is expressed as follows,

$$P(i) = \frac{(1-p)\, p^{i-1}}{\sum_{j=1}^{m+1} (1-p)\, p^{j-1}} = \frac{(1-p)\, p^{i-1}}{1-p^{m+1}} \quad . \tag{B.2}$$

Therefore, the mean delay time of all successful transmission is calculated from Eq. (B3),

$$\begin{aligned} Delay &= \sum_{i=1}^{m+1} D(i) \cdot P(i) = \sum_{i=1}^{m+1} \left\{ T_{erss} \cdot \frac{(2^i - 1)}{2} \cdot W_0 \cdot \frac{(1-p)\, p^{i-1}}{1-p^{m+1}} \right\} \\ &= T_{erss} \cdot \frac{W_0}{2} \cdot \left(2 \cdot \frac{1-(2p)^{m+1}}{1-2p} \cdot \frac{1-p}{1-p^{m+1}} - 1 \right) \quad . \end{aligned} \tag{B.3}$$

Chapter 7
Conclusion

In this thesis, Quality-of-Service (QoS) control schemes in the medium access control (MAC) layer of wireless local area networks (WLANs) were investigated in order to guarantee some kinds of QoS parameters such as throughput, throughput fairness, transmission delay and delay variation. Resource management scheme of MAC layer in wireless communication has mainly two types: centralized access control scheme and distributed access control scheme. This thesis revealed problems in both types of management schemes so as to control QoS level according user requests and proposed new and effective approaches to overcome the problems. Performances of the proposals were confirmed by computer simulations.

The chapter 3 of this thesis proposed a new polling-based dynamic slot assignment (DSA) scheme for the broadband wireless access systems that allow the access point (AP) to assign slots. It was revealed that the conventional constant length assignment scheme can have difficulty in achieving satisfactory throughput performance and short delay performance at the same time, especially when the traffic of one or more mobile terminals (MTs) is excessive. Unlike to the conventional scheme, the proposed scheme exhibits excellent performance; it offers fair slot assignment while maintaining high throughput and short delay performance even in unbalanced-traffic circumstances, mainly by employing the exponential decreasing and proportional increasing rate control scheme. We also proposed a new resource request scheme that combines polling and random access and clarified its performance. The proposed scheme uses polling access when traffic is not bursty and random access otherwise. Consequently, it minimizes the transmission delay by matching the MT's traffic characteristics. Combining the proposed slot assignment scheme and the resource request scheme allows us to use the given radio bandwidth efficiently while guaranteeing low delay with robustness against other MT's traffic. Since, in practice, some MTs may yield significantly large traffic compared to the other MTs, the proposed scheme offers significant benefit for wireless data access systems that aim to provide stable services.

The chapter 4 explored virtual destination (VD)/ virtual source (VS)-based available bit rate (ABR) flow control performance targeting application to wireless asynchronous transfer mode (WATM) in which large link-delay can be inherent. It was revealed that the conventional feedback-based VD/VS scheme has difficulty in sustaining satisfactory ABR performance, when it is applied for long-delay-causing WATM. To ensure the ABR performance in WATM, we proposed a new VD/VS scheme that controls the allowed cell rate (ACR) of a source end system (SES) in a feed-forward manner. The proposed scheme calculates the number of cells in the WATM-associated loop and predicts the queue length of the WATM-round trip time (RTT). The proposed scheme exhibits excellent ABR performance even when the WATM-RTT is rather long. In addition, the proposed scheme possesses robustness against uncertainty and/or time-variation regarding the predetermined WATM-RTT. At a moderate cost of implementation complexity for the queue length prediction, memorizing the backward resource management (BRM) cell transmitting time and associated congestion indication (CI) bits, the proposed scheme enables long-RTT-causing WATM to well support ABR service in practice. When systems have extensive WATM-RTT variation or difficulty in identifying the WATM-RTT, technical modification may be needed.

The chapter 5 has introduced a QoS control scheme that alters the inter frame space (IFS) length of stations (STAs) to suit their traffic conditions in WLANs. It controls the transmission rate of data frames so as to guarantee throughput for each users. We showed that just four low priority (LP) class STAs trigger priority inversion when the conventional scheme is employed and thus it can not provide QoS assurances if LP class STAs transmit large numbers of data frames at low speed. We have also showed that the proposed QoS control scheme guarantees QoS in terms of throughput regardless of the LP class traffic, and that it offers excellent performance, even in this worst-case situation.

The chapter 6 proposed a QoS control scheme for a wide area ubiquitous wireless network; it keeps the transmission delay time of high priority (HP) class traffic within a specified limit by using dynamic back-off window size control of each QoS class. The proposed scheme calculates suitable values of initial back-off window sizes (IWSs) using an equation derived from the Markov chain behavior models of wireless terminals (WTs) and the measured traffic. In this chapter, we showed that the conventional scheme cannot provide QoS assurances when the number of LP class active WTs increases by 50 percent; however, the proposed scheme exhibits excellent performance even in this worst-case situation.

In this thesis, QoS control schemes of WLANs were studied. Wireless communication networks are been developed and near future networks will be more complex as a result of combining WLANs with other wireless systems such as long term evolution (LTE), worldwide interoperability for microwave access (WiMAX). In near future complex networks, cross layer QoS control schemes, which manage some functions in several layers, will achieve important roles to guarantee QoS. Therefore, future researches should be carried out on study to cross layer QoS controls in order to adapt network complexity. The demands for supporting a great variety of applications will increase. Future QoS control schemes will ask for providing users with more QoS classes. However, the number of QoS classes the proposals provide is two or three in this thesis. Therefore, future researches should achieve to establish QoS control schemes that provide more kinds of QoS classes.

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