Provisioning QoE and QoS Guarantee for Audio-Video Transmission over IEEE 802.11e HCCA Wireless LANs

Noh Zul Azri Bin Muhamad

2010-03-23

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Provisioning QoE and QoS
Guarantee for Audio-Video
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HCCA Wireless LANs

ZUL AZRI BIN MUHAMAD NOH
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Chapter 1

Introduction

Wireless networks have been an essential part of communication to make Internet service available to users from anywhere, at anytime, by anyone and anything. It is indispensable in establishing ubiquitous network society in Japan by 2010 as promoted by the Ministry of Internal Affairs and Communications [1]. In conjunction with that, wireless broadband network infrastructures have been deployed rapidly to reach 100% population coverage. As a result, the number of internet users in Japan reached 90.91 million people as of the end of 2008, with an Internet penetration rate of 75.3% [2].

The diffusion of Internet has shifted the network paradigm towards “Everything on Internet protocol (IP)” and “IP on Everything” ideas, where various contents and applications are developed on IP, and various infrastructures are deployed based on IP. Since then, a variety of multimedia applications such as voice over IP (VoIP), video conferencing, and live media streaming are developed on IP and becoming prevalent.

In many wireless network standards, IEEE 802.11 wireless local area network (WLAN) has become a major standard in wireless packet communications. From public hotspots to in-flight wireless networks, the IEEE 802.11 WLAN plays a prominent role in offering ubiquitous connectivity to the Internet. With an increasing demand for networked multimedia applications with strict delay constraint, the interest in WLANs supporting Quality of Service (QoS) has been growing rapidly. However, the IEEE 802.11 was originally designed for data transmission; it offers services on the best-effort basis. Therefore, multimedia applications which mostly require data to be transmitted in real time are exposed to networks delay and jitter, resulting in degradation in Quality of Experience (QoE) as perceived by the end users.
1. INTRODUCTION

In providing QoS guarantee service to IEEE 802.11 WLANs, IEEE 802.11 Task Group e has introduced the IEEE 802.11e, an enhancement of the IEEE 802.11 medium access control (MAC) protocol. The IEEE 802.11e MAC defines the hybrid coordination function (HCF), which has two access methods: enhanced distributed channel access (EDCA) and HCF controlled channel access (HCCA) [3]. The former is a contention-based protocol based on carrier sense multiple access with collision avoidance (CSMA/CA) that can support priority-based service differentiation. The latter is a polling-based protocol and can support guaranteed media access for real-time transmission. The EDCA, which is based on distributed control, is easy to be implemented, but under heavy load conditions, QoS cannot always be met. For this reason, the centrally controlled HCCA is preferred in providing high assurance of QoS guarantee. Therefore, this thesis will focus on the audio-video transmission with the HCCA.

In general, this thesis deals with QoE and QoS assessment of audio-video transmission by the IEEE 802.11e HCCA. This thesis investigates and clarifies the major factors that impede the QoS. Several packet scheduling schemes for audio-video transmission are proposed and their impact on application-level QoS and QoE is evaluated.

The rest of this section is organized as follows. Section 1.1 describes background of wireless IP communications. In Section 1.2, the concept of QoS in multimedia communications is introduced. Section 1.3 elaborates psychometric method in QoE assessment. Section 1.4 explains the IEEE 802.11 WLAN technology including IEEE 802.11e MAC protocols. Section 1.5 explains the packet scheduling concept of the HCCA. This is followed by Section 1.6, which discusses the media coding and compression scheme. Next, in Section 1.7, the media synchronization is discussed. Finally in Section 1.8, thesis objectives and structure are discussed.

1.1 Wireless IP Communications

Until now, voice and data transmissions are traditionally handled by separate network architectures. However, nowadays, the majority of incumbent network operators have started migrating their existing circuit-switched voice networks and packet-switched data networks, to a single packet-switched networks by the QoS guaranteed Next Generation Network (NGN) platform.
NGN is a packet-based network able to provide services including telecommunication services and able to make use of multiple broadband, QoS-enabled transport technologies and in which service-related functions are independent from underlying transport-related technologies [4, 5]. It offers unrestricted access by users to different service providers. It also supports generalized mobility which will allow consistent and ubiquitous provision of services to users.

In NGN, we would see the use of all IP communications where all communication devices will have unique IP addresses that act as phone number to connect with each others. In IP communications, communication devices are not locked into certain networks, but have the capability to access different access technologies such as IEEE 802.11 WLAN and worldwide interoperability for microwave access (WiMAX) more easily.

Wireless networks can be classified into three categories according to their network area coverage; wireless personal area network (WPAN), WLAN, and wireless metropolitan network (WMAN).

Bluetooth (ratified as IEEE 802.15.1 [6]) is widely used especially for WPAN as it offers short range wireless networks, generally within reach of a person. For example, it is used to connect a computer with its peripherals such as printers, mouse, keyboard and digital cameras. Bluetooth uses a globally available unlicensed 2.4 GHz Industrial, Scientific, and Medical (ISM) band for worldwide compatibility and can achieve a gross data rate of 1 Mb/s. In the latest Bluetooth specification, it can support data transfer speed of up to 24 Mb/s [7]. Besides Bluetooth, Infrared Data Association (IrDA), ultra-wideband (UWB), and ZigBee are also categorized as WPAN.

In WLAN, IEEE 802.11 WLAN has been widely deployed, coinciding with the proliferation of personal computers. Almost every computers manufactured today are equipped with IEEE 802.11 WLAN connectivity. The IEEE 802.11 WLAN offers mid-range wireless networks suitable for a small area such as a home, office, and school. IEEE 802.11b and IEEE 802.11g operate in the 2.4 GHz band and offer transmission speed of up to 11 Mb/s and 54 Mb/s, respectively [8, 9]. Meanwhile, IEEE 802.11a supports data rate of up to 54 Mb/s using the 5 GHz band [10]. The latest IEEE 802.11n uses both 2.4 GHz and 5 GHz band with maximum theoretical data rate of 600 Mb/s [11]. Additionally, in 2005, an approved amendment to the IEEE 802.11
1. INTRODUCTION

standard, known as IEEE 802.11e, was defined to provide QoS control for multimedia applications through MAC modifications [3].

In WMAN, WiMAX or IEEE 802.16 offers an alternative to cabled access networks because its wide range network coverage has the capacity to address broad geographic areas without the costly infrastructure development. This standard was designed to evolve as a set of air interfaces based on common IEEE 802.11 MAC protocol but with different physical layer specifications which works within 10 to 66 GHz range [12]. However, WiMAX penetration is very small compared to the IEEE 802.11 WLAN.

1.2 QoE and QoS in Multimedia Communications

In multimedia communications, unlike e-mail, file transfer protocol (FTP), and web traffic, multimedia traffic such as audio and video streams required stringent QoS guarantee requirements especially in error-prone wireless networks. Audio and video traffic contain temporal structures that may be ruined by transmission errors or delay jitter. Therefore, QoS guarantee is indispensable in transmitting multimedia traffic.

<table>
<thead>
<tr>
<th>Network Layer</th>
<th>QoS Level</th>
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<td>User Layer</td>
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</tr>
<tr>
<td>Application Layer</td>
<td>Application-Level QoS</td>
</tr>
<tr>
<td>Transport Layer</td>
<td>End-to-End Level QoS</td>
</tr>
<tr>
<td>Network Layer</td>
<td>Network-Level QoS</td>
</tr>
<tr>
<td>Data Link Layer</td>
<td>Link-Level QoS</td>
</tr>
<tr>
<td>Physical Layer</td>
<td>Physical-Level QoS</td>
</tr>
</tbody>
</table>

QoS has different definitions depend on condition where it is applied. Reference [13] defines it as the degree of the way it should be. The Internet Engineering Task Force (IETF) Integrated Services (IntServ) working group defines the QoS as absolute performance that can be quantitatively measured. In RFC2216[14], QoS refers to the nature of the packet delivery service provided, as described by parameters such as achieved bandwidth, packet delay, and packet loss rates.
1.2 QoE and QoS in Multimedia Communications

Similar to layered network systems as in the Open System Interconnection (OSI) reference model, QoS can also be classified into several levels as shown in Table 1.1. Reference [15] mentioned that in IP networks, six kinds of QoS can be identified along the protocol stack: physical-level, link(node)-level, end-to-end(transport)-level, application-level, and user-level\(^1\).

Many researches on wireless multimedia communications focus on an assessment of QoS at the end-to-end and lower levels. However, provision of QoS guarantee to multimedia transmission should imply not only achieving high throughput and low delay but also high user satisfaction (user-level QoS), as the users are the ultimate recipients of multimedia application services. In multimedia communications, the end receivers are the users; it is important to focus on user-level QoS [16]. Here, the user-level QoS is the overall acceptability of an application or service, as perceived subjectively by the end-users and is the most important in multimedia transmission; this is also referred to as QoE in International Telecommunication Union-Telecommunication Standardization (ITU-T) [17]. Although we cannot directly control QoE, it is possible to control QoS at lower levels so that QoE can be kept high. Therefore, this thesis evaluates QoS at application-level and user-level (QoE).

In order to analyze the system performance in a network, QoS must be quantitatively measured. Since every level of QoS has its own QoS requirements, different type of parameters can be defined at each QoS level. Furthermore, at the application-level, QoS parameters can be specifically defined according to the type of traffic. Table 1.2 shows an example of QoS parameters.

Since this thesis deals on QoS assessment of audio-video transmission at application-level QoS and QoE, first, the definition of each application-level QoS parameters used in this thesis is given.

**Application-level QoS parameters**

This thesis considers the fidelity and the latency as the application-level QoS parameters. The fidelity indicates how exactly the temporal structure of media is preserved.

\(^1\)There is no user layer in layered networking, but while the application layer service entity’s service user is the user, we include the user layer in Table 1.1.
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<table>
<thead>
<tr>
<th>QoS Level</th>
<th>QoS Parameters</th>
</tr>
</thead>
<tbody>
<tr>
<td>QoE</td>
<td>Mean Opinion Score (MOS), Psychological Scale</td>
</tr>
<tr>
<td>Application-Level QoS</td>
<td>PESQ, R-Value, MU Rate, SNR, Throughput, Delay</td>
</tr>
<tr>
<td>End-to-End Level QoS</td>
<td>Throughput</td>
</tr>
<tr>
<td>Network-Level QoS</td>
<td>Packet Delay, Packet Delay Jitter</td>
</tr>
<tr>
<td>Link-Level QoS</td>
<td>Packet Loss Ratio</td>
</tr>
<tr>
<td>Physical-Level QoS</td>
<td>Transmission Speed, Bit Error Rate, SNR</td>
</tr>
</tbody>
</table>

For this reason, the fidelity also relates to media synchronization quality. This thesis will treat only two types of media: audio and video.

In order to represent the media synchronization quality and the latency, this thesis uses six application-level QoS parameters. At the application-level QoS, the transmission unit of data is known as *media unit* (*MU*). A video MU is defined as a video frame, and an audio MU consists of 1000 audio samples\(^1\).

**Average MU delay** (audio: \(D_a\), video: \(D_v\)) The average time from the moment an MU is generated at the source station until the moment the MU is output at the receiver.

**MU loss ratio** (audio: \(L_a\), video: \(L_v\)) The ratio of the number of MUs not output at the receiver to the number of MUs generated by the source station.

**Average MU rate** (audio: \(R_a\), video: \(R_v\)) The average number of MU output in a second at the receiver. Since the MU loss ratio highly correlates with the average MU rate, this thesis does not utilize the average MU rate in this thesis.

**Coefficient of variation of output interval** (audio: \(C_a\), video: \(C_v\)) The ratio of the standard deviation of the MU output interval to its average. This QoS parameter represents the smoothness of the output flow.

\(^1\)We use an audio flow of ITU-T G.711 µ-law sampled at 8 kHz, with 8 bits per sample.
1.3 Psychometric Method in QoE Assessment

Mean square error of intra-stream synchronization (audio: $E_a$, video: $E_v$) The average square of difference between the output interval of MU at the receiver and the generation one at the source station. This thesis does not treat mean square error of intra-stream synchronization as it highly correlates with the coefficient of variation of output interval.

Mean square error of inter-stream synchronization ($E_{int}$) An indicator of "lip-sync" and is the average square of the difference between the output time of each video MU and its derived output time obtained from the output time of the corresponding audio MU. The derived output time of a video MU means the output time of the corresponding audio MU plus the difference between the timestamps of the two MUs.

1.3 Psychometric Method in QoE Assessment

In order to investigate user satisfaction, the QoE must be assessed quantitatively, which can be done by subjective experiment. Regarding the subjective experiment in QoE assessment, mean opinion score (MOS) [18] has been a popular QoE parameter especially in technical papers and ITU-T/ITU-R recommendations. MOS is obtained by the rating-scale-method [19], where an observer classifies objects or events into a certain number of categories each assigned an integer. Basically, the MOS value is obtained by averaging the subjective scores under an implicit assumption that the difference in integer between any two successive categories means the same magnitude of the assessor’s sensation to a sensory attribute of a sample; assessor’s sensation between category 5 and category 4, for example, has the same magnitude as that between category 3 and category 2. However, this is not necessarily the case. Thus, the MOS is an ordinal scale\(^1\); the integers assigned to the categories only have a greater-than-less-than relation between them.

As an exact measure, we are required to use at least an interval scale, where the intervals between the scale value represent difference or distance between amounts of

\(^1\)We can define four basic types of the measurement scale according to the mathematical operations that can be performed legitimately on the numbers obtained by the measurement; from lower to higher levels, we have nominal, ordinal, interval, and ratio scales.
1. INTRODUCTION

the sensory attribute measured [20, 21]. Moreover, almost all the statistical procedures can be applied to the interval scale. In this thesis, the interval scale is utilized as the measurement scale.

Since QoE is directly related to human perception, this thesis utilizes a psychometric method [22] referred to as the method of successive categories [20]. In the next subsection, the method of successive categories and QoS mapping is explained.

1.3.1 Method of successive categories

In the method of successive categories, a subjective score is measured by the rating-scale method. In the method, assessors are asked to classify the test samples into a certain number of categories each assigned an integer. In this thesis, we use five categories of impairment of the rating-scale method as shown in Table 1.3: "imperceptible" assigned integer 5, "perceptible, but not annoying" 4, "slightly annoying" 3, "annoying" 2, and "very annoying" 1. However, the numbers assigned to the categories is nothing but an ordinal scale. Therefore, we need to transform the obtained ordinal scale into the interval scale.

<table>
<thead>
<tr>
<th>Category</th>
<th>Criterion</th>
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<tbody>
<tr>
<td>5</td>
<td>Imperceptible</td>
</tr>
<tr>
<td>4</td>
<td>Perceptible, but not annoying</td>
</tr>
<tr>
<td>3</td>
<td>Slightly annoying</td>
</tr>
<tr>
<td>2</td>
<td>Annoying</td>
</tr>
<tr>
<td>1</td>
<td>Very annoying</td>
</tr>
</tbody>
</table>

According to the method of successive categories, the results obtained from the rating-scale method can be calculated into the interval scale with the law of categorical judgment [19].

1.3.2 The law of categorical judgment

In the law of categorical judgment, the following assumptions are made. Let the number of categories be \( m + 1 \). When test samples \( j(j = 1, \ldots, n) \) is presented to an assessor, a
1.3 Psychometric Method in QoE Assessment

A psychological value designated by $s_j$ occurs on a *psychological continuum*, which is an interval scale, in the assessor. For the $m + 1$ categories, their boundaries have value on the interval scale. We denote the upper boundary of category $g (g = 1, \ldots, m + 1)$ by $c_g$ and define $c_0 \triangleq -\infty$ and $c_{m+1} \triangleq +\infty$. The assessor classifies $n$ sample into the $m + 1$ categories ($n > m + 1$) by comparing $s_j$ with $c_g$. If $c_{g-1} \leq s_j < c_g$, then sample $j$ is classified into category $g$. The categories can be arranged in a rank order, in the sense that each sample in category $g$ is judged to have psychological value which is “less than” the one for any sample in category $g + 1$. This statement holds for all values of $g$ from 1 to $m$. The variable $c_g$ is normally distributed with mean $t_g$ and standard deviation $d_g$. Moreover, the variable $s_j$ is normally distributed with mean $R_j$ and standard deviation $\sigma_j$. Then, we can consider $R_j$ as an interval scale.

Since the law of categorical judgment is a suite of assumption, we must test goodness of fit between the obtained interval scale and the measurement result. In order to verify the obtained interval scale, we have to perform Mosteller’s test [23] using Thurstone’s law of comparative judgment [24], which is one of psychometric methods. Then, we refer to the interval scale as the *psychological scale* for the QoE parameters [20].

1.3.3 QoS mapping

As the QoE is a perceptual quality metrics, we cannot directly control it. However, it is feasible to control QoS at lower levels so that the QoE can be kept high. In order to do so, we must find the correspondence of QoE to QoS at lower levels. The correspondence is referred to as *QoS mapping*. In this thesis, QoS mapping from application-level to user-level is performed. As a QoS mapping method, this thesis uses *multiple regression analysis* [25, 26].

In the QoS mapping, this thesis considers the QoE parameter as the criterion variable (dependent variable), and the application-level QoS parameters as predictor variables (independent variables). An estimate $\hat{S}$ of the QoE parameter can be expressed with $n$ application-level QoS parameters $A_1, A_2, \ldots, A_n$ by

$$\hat{S} = \beta_0 + \beta_1 A_1 + \ldots + \beta_n A_n$$

(1.1)

where $\beta_i (1 \leq i \leq n)$ is the partial regression coefficient of the $i$-th application-level QoS parameters, and $\beta_0$ is the intercept.
1. INTRODUCTION

Before performing the multiple regression analysis, we must classify the application-level QoS parameters by the principal component analysis. As a result of the principal component analysis, we obtain the cumulative contribution rate for each principal component. Then, we calculate the principal component loading of each principal component for all the application-level QoS parameters. With these values, the application-level QoS parameters can be classified into some groups. To avoid the effect of multi-collinearity, one application-level QoS parameter is selected from each group as predictor variables. After that, all the obtained combinations of the application-level QoS parameters are treated as predictor variables and perform multiple regression analysis. We select the combination which indicates the highest contribution rate adjusted for degree of freedom. Finally, multiple regression analysis with selected predictor variables is carried out again. By the statistical test, predictor variables which give little contribution to the multiple regression line are removed.

1.4 IEEE 802.11 Wireless LAN

In 1997, the Institute of Electrical and Electronics Engineers (IEEE) approved the first WLAN standard, known as IEEE 802.11 [27]. This standard covers distributed and centralized control of data link level, and modulation schemes and frequency of physical layer. The main characteristics of the IEEE 802.11 standard are simplicity and robustness against failures. At the beginning, the IEEE 802.11 WLAN only offers a maximum transmission speed of 2 Mb/s for data transmission purpose. But, in 1999, operating in the ISM band at 2.4 GHz, IEEE 802.11b was introduced to provide data rates of up to 11 Mb/s, applying complementary code keying (CCK) and direct sequence spread spectrum (DSSS). In the same year later, IEEE 802.11a that operates in the unlicensed 5 GHz band was introduced. Using the multicarrier technique orthogonal frequency division multiplexing (OFDM), the IEEE 802.11a can offer data rates up to 54 Mb/s. Four years later, in 2003, IEEE 802.11g standard was approved with the same data rates as the IEEE 802.11a but operates in the 2.4 GHz band. Recently, in October 2009, the IEEE 802.11 working group (WG) has finalized IEEE 802.11n that promises higher transmission speed and wider transmission range. Table 1.4 shows the major standards in the IEEE 802.11 WLAN.
Table 1.4: The IEEE 802.11 WLAN standards

<table>
<thead>
<tr>
<th>Standard</th>
<th>Approval Year</th>
<th>Max. transmission speed</th>
<th>Frequency</th>
<th>Modulation scheme</th>
</tr>
</thead>
<tbody>
<tr>
<td>IEEE 802.11</td>
<td>1997</td>
<td>2 Mbps</td>
<td>2.4 GHz</td>
<td>DSSS/FHSS</td>
</tr>
<tr>
<td>IEEE 802.11b</td>
<td>1999</td>
<td>11 Mbps</td>
<td>2.4 GHz</td>
<td>DSSS</td>
</tr>
<tr>
<td>IEEE 802.11a</td>
<td>1999</td>
<td>54 Mbps</td>
<td>5 GHz</td>
<td>OFDM</td>
</tr>
<tr>
<td>IEEE 802.11g</td>
<td>2003</td>
<td>54 Mbps</td>
<td>2.4 GHz</td>
<td>DSSS-OFDM</td>
</tr>
<tr>
<td>IEEE 802.11n</td>
<td>2009</td>
<td>600 Mbps</td>
<td>2.4/5 GHz</td>
<td>OFDM</td>
</tr>
</tbody>
</table>

The IEEE 802.11 WLAN has two different topologies: *infrastructure mode*, and *ad hoc mode*. The former requires an *access point (AP)* that usually connected to wired network. An AP with several wireless stations is known as *basic service set (BSS)*. On the other hand, the latter requires no AP as each wireless station communicates directly to other wireless stations. This is well known as *independent BSS (IBSS)*. An area where the communications take place is called the *basic service area (BSA)*.

### 1.4.1 IEEE 802.11 MAC protocols

The IEEE 802.11 WLAN standard defines two MAC protocols: *distributed coordination function (DCF)* and *point coordination function (PCF)*. The DCF is the fundamental access method based on *carrier sense multiple access with collision avoidance (CSMA/CA)* and can be implemented in both infrastructure based BSS and IBSS. The PCF on the contrary is an optional access method that can only be used in infrastructure based BSS. This method uses a central coordinator, the *point coordinator (PC)*, collocated with an AP, which assigns the permission to transmit by a round robin polling procedure. The details about these two MAC protocols will be described below.

**DCF**

DCF is a distributed system that is based on CSMA/CA. Basically this system works like push-to-talk radio where only one person can talk at a time. In a wireless channel, if more than one station tries to send data simultaneously, a collision will occur.
Therefore, stations have to detect that there is no other transmission in progress before starting a transmission. However, if two stations detect the channel as idle at the same time, a collision occurs if both stations initiate a transmission. The collision avoidance (CA) in DCF is defined to reduce the probability of such collisions. Stations have to wait for a random time called backoff timer after detecting the channel as being idle for a minimum duration called *DCF interframe space (DIFS)* as shown in Fig. 1.1. Only if the channel remains idle for this additional backoff timer, a station is allowed to initiate its transmission. Here, the backoff timer is obtained from Eq. (1.2).

\[
\text{BackoffTimer} = [0, CW] \times \text{SlotTime} \quad CW_{\text{min}} \leq CW \leq CW_{\text{max}} \tag{1.2}
\]

From Eq. (1.2), backoff timer is obtained from product of *SlotTime* and a randomly picked value range of 0 through *contention window (CW)*. The initial value of CW is \( CW_{\text{min}} \) and bounded by \( CW_{\text{max}} \).

![Figure 1.1: DCF channel access](image)

After waiting for DIFS, each station decreases its backoff timer by *SlotTime*. Stations stop the decrement of their backoff timer if the channel is busy again, and continue decreasing the remaining backoff timer again when idle channel is detected. The first station whose backoff timer reach *zero* has a chance to transmit its frame. In DCF, only one frame or *MAC service data unit (MSDU)* can be transmitted at every transmission opportunity got by a station.

When a frame is received successfully, the recipient will transmit an *acknowledgment frame (ACK)* right away after a *short interframe space (SIFS)*. After each successful transmission, another random backoff which is called post-backoff is performed by the sender station even if no other pending frames is delivered. The CW of transmitting
station increases when a transmission fails. After any unsuccessful transmission attempt, a new backoff procedure is performed with double-sized CW, up to a maximum value defined by $CW_{\text{max}}$. For the $n$-th retransmission, the CW is calculated as Eq. (1.3) with a maximum value of $CW_{\text{max}}$.

$$CW \leftarrow \min\left[2^n(CW_{\text{min}} + 1) - 1, CW_{\text{max}}\right] \quad (n: \text{number of retransmission}) \quad (1.3)$$

**PCF**

![Figure 1.2: An example of transmission in PCF](image)

Unlike DCF, PCF is a centralized polling based scheme, coordinated by the PC, which usually collocated with an AP. The PC polls stations in a round robin fashion. PCF has higher priority than DCF because it may start transmissions after a shorter period than DIFS; the period is known as *PCF interframe space (PIFS)*. Figure 1.2 shows an example of frames transmission in PCF.

In PCF, a *contention free period (CFP)* and a *contention period (CP)* alternate periodically over time, where a concatenation of a CFP and a CP forms one superframe. The PCF is used only during the CFP, whereas DCF is used during the CP.

During the CFP, there is no contention among stations; instead, stations are polled. The PC polls each station according to its polling list by sending CF-Poll frame to a station. In a time, only the polled station could transmit its frames. Any polled station is allowed to transmit only one frame or MSDU. If AP does not receive any response from the polled station after waiting for a PIFS, it polls the next station according to its polling list. Therefore, no idle period longer than a PIFS occurs during a CFP. Since only one station can transmit a frame at a time, no collision occurs. The PC continues with polling other stations until the CFP expires, or there are no stations with pending frames.
1. INTRODUCTION

1.4.2 IEEE 802.11e

The IEEE 802.11 WLAN is first designed for data transmission to offer best-effort service. However, the interest in wireless network supporting QoS has recently grown. But, there are a number of problems with the current MAC protocol. Among many others, those include the unpredictable beacon delays and unknown transmission duration of the polled stations of PCF. This may severely affect the QoS as this problems lead to unpredictable time delay in each CFP. Furthermore, in the case of DCF, all stations use the same value for $CW_{\text{min}}$, but select their random backoff timer individually. This result in all stations has the same medium access priority in the DCF making it difficult to adopt a mechanism to differentiate between stations and their traffic [28]. For that, to introduce QoS support in IEEE 802.11, the IEEE Task Group e has introduced an enhancement of IEEE 802.11 MAC protocols called IEEE 802.11e.

The IEEE 802.11e introduces the hybrid coordination function (HCF) for QoS support. The HCF consists of two medium access methods: enhanced distributed channel access (EDCA) and HCF controlled channel access (HCCA) [3]. The former is a contention-based protocol based on CSMA/CA and can support relative priority services for multimedia transmission. The latter is a polling-based protocol and can support guaranteed media access for real-time transmission.

EDCA

EDCA is designed to provide prioritized QoS by enhancing the contention based DCF using eight different user priorities (UP). The EDCA mechanism defines four access categories (AC) that provide support for the delivery of traffic with UP at the stations [29, 30]. Here, the eight UPs are mapped into four ACs according to Table 1.5. The ACs are labeled according to their target application like $AC_{\text{VO}}$ (voice), $AC_{\text{VI}}$ (video), $AC_{\text{BE}}$ (best effort), and $AC_{\text{BK}}$ (background).

Packets that arrive from the upper layer are mapped into four ACs and served into independent queue entities. Each queue entity has its own parameters to determine backoff timer. Unlike DCF, EDCA uses arbitration inter frame space (AIFS) instead of DIFS for the contention processes shown in Fig.1.3. The duration of AIFS[AC], which defines as AIFS for particular AC, is determined by the following equation:
Table 1.5: The mapping between the UP and the AC

<table>
<thead>
<tr>
<th>Priority</th>
<th>AC</th>
<th>Packet type</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>AC_BK</td>
<td>Background</td>
</tr>
<tr>
<td>2</td>
<td>AC_BK</td>
<td>Background</td>
</tr>
<tr>
<td>0</td>
<td>AC_BE</td>
<td>Best Effort</td>
</tr>
<tr>
<td>3</td>
<td>AC_VI</td>
<td>Video</td>
</tr>
<tr>
<td>4</td>
<td>AC_VI</td>
<td>Video</td>
</tr>
<tr>
<td>5</td>
<td>AC_VI</td>
<td>Video</td>
</tr>
<tr>
<td>6</td>
<td>AC_VO</td>
<td>Voice</td>
</tr>
<tr>
<td>7</td>
<td>AC_VO</td>
<td>Voice</td>
</tr>
</tbody>
</table>

Figure 1.3: EDCA channel access

\[
AIFS[AC] = SIFS + AIFSN[AC] \times SlotTime \quad (AIFS[AC] > 0) \quad (1.4)
\]

where AIFSN[AC] is arbitration interframe space number (AIFSN) for particular AC. Here, the AIFS[AC] has a minimum value equivalent to DIFS. In addition, the value for \( CW_{\text{min}}[AC] \) and \( CW_{\text{max}}[AC] \) are selected per AC. The idea of using different contention parameters for different ACs is to give low priority packets a longer waiting time than high priority packets; so the high priority packets are likely to access the wireless medium earlier than the low priority packets. If packets collision from different ACs within a station occurs since both AC backoff timer reach zero simultaneously, a vir-
1. INTRODUCTION

tual scheduler inside every station allows only the highest priority packets to transmit frames.

HCCA

In HCCA, QoS support is achieved through packet scheduling and admission control. HCCA provides parameterized QoS support and is controlled by the hybrid coordinator (HC), which is collocated with an AP. In HCCA, the HC is allowed to start contention free transmission or known as controlled access period (CAP) at any time. For this purpose, the HC just has to wait for channel being idle for PIFS. Since PIFS is shorter than DCF and AIFS, the HC is given greater opportunity to initiate HCCA than EDCA. Figure 1.4 shows an example of HCCA channel access within a beacon period where the HC grants TXOPs in both CFP and CP.

![HCCA Channel Access](image)

Figure 1.4: An Example of HCCA mechanism

Unlike PCF, stations are allowed to transmit a burst of MSDU within a period of time called transmission opportunity (TXOP). Stations must not initiate transmission unless the remaining TXOP duration is enough to finish that transmission. Figure 1.5 shows an example of TXOP allocation sequences which formed a CAP in an service interval (SI). In order to be included into the HC polling list, stations must send a QoS reservation request to the HC by sending traffic specification (TSPEC) parameters [31]. The HC then calculates the TXOP duration utilizing the TSPEC. The HC only allocates TXOP to stations which have been admitted by admission control. The TSPEC parameters are listed as below.

1. Mean Data Rate($\rho$)
2. Delay Bound(D)
3. Maximum Service Interval (MSI)

4. Nominal MSDU Size (L)

5. User Priority (UP)

6. Maximum MSDU Size (M)

7. Maximum Burst Size (MBS)

8. Minimum PHY Rate (R)

9. Peak Data Rate (PR)

![Contention Free Burst (CAP)](image)

**Figure 1.5: An Example of TXOP allocation**

During the CFP or CAP, the HC allocates TXOP to a station by sending *QoS CF-Poll* frame. The success of each MSDU transmission must be acknowledged immediately after the SIFS interval from the last transmission. If the acknowledgment frame is not received, the sender retransmits the corresponding MSDU, after PIFS from the end of the last transmission. An MSDU is subject to retry limit as well as MSDU lifetime parameter and is discarded when either of them is exceeded.

The methods of TXOP calculation and admission control in HCCA are elaborated in Section 1.5.
1. INTRODUCTION

1.5 The TGe Scheme

For reference, the IEEE 802.11e standard has presented an example packet scheduler, which is referred to as the Task Group e (TGe) scheme in this thesis. In the TGe scheme, the TXOP duration for a station is calculated by the HC on the basis of the TSPEC information received from the station. The TSPEC consists of a set of parameters that include the mean data rate, delay bound, nominal MSDU size, and MSI.

For simplifying the description of the TGe scheme, let us focus on flow $i$ where $\rho_i$ is the mean data rate in bits per second, $L_i$ is the nominal MSDU size in bits, and $R_i$ is the minimum physical transmission rate of flow $i$ in bits per second.

In the TGe scheme, the HC first calculates the number of MSDUs of flow $i$ that arrives at the mean data rate during an SI as

$$N_i = \left\lfloor \frac{SI \times \rho_i}{L_i} \right\rfloor$$

where $SI$ is the duration of the service interval in seconds. The SI must be shorter than the minimum value of all MSI’s for admitted flows and must be a submultiple of the beacon interval. If a new flow is admitted with a MSI smaller than the current SI, the scheduler needs to change the current SI to a smaller value than the MSI of the newly admitted flow. In Eq. (1.5), the ceiling function is used; the ceiling function of $x$ is defined as the smallest integer greater than or equal to $x$.

Then, the TXOP duration for flow $i$ is computed as

$$TXOP_i^{TGe} = \max \left( \frac{N_i \times L_i}{R_i} + O, \frac{M}{R_i} + O \right)$$

where $M$ is the maximum size of an MSDU, and $O$ is the overhead in time units due to the physical header, MAC header, inter-frame space (IFS), acknowledgment frames, and poll frames. The superscript $TGe$ means the TGe scheme.

Owing to the limitation of the capacity during an SI, the number of stations to which TXOP can be allocated is limited. Therefore, the TGe scheduler implements admission control to ensure that all admitted flows have adequate TXOPs for their QoS requirements. Admission control decides which flow should be admitted and which flow
should be dropped from the polling list. When flow $k+1$ issues a QoS reservation, the HC will first check whether the available capacity of the medium exists or not by the following equation:

\[
\frac{TXOP_{TGe}^{k+1}}{SI} + \sum_{i=1}^{k} \frac{TXOP_{TGe}^{i}}{SI} \leq \frac{T - T_{CP}}{T}
\]  

where $T$ is the beacon interval, and $T_{CP}$ is the time for the EDCA or CP. If Eq. (1.7) is satisfied, the HC admits flow $k+1$ into its polling list and allocates TXOP to the flow.

The TGe scheme uses the mean data rate to compute the TXOP duration. This implies that the TGe scheduler allocates fixed TXOP duration in every SI; therefore this scheme is suitable for constant bit rate (CBR) traffic. Moreover, the use of ceiling function as in the Eq. (1.5), which always returns a greater integer value also contributes to problems such as allocation of excessive TXOP duration and errors in determining the incoming number of MSDUs.

1.6 Media Coding and Compression Schemes

In mobile multimedia communications, media coding and compression standards play an important role to assure that the transmitted audio or video contents are compatible with recipient system environments. This thesis utilizes G.711 $\mu$-law[32] for audio, and MPEG-1[33] and H.264[34] for video.

G.711 $\mu$-law coding scheme

G.711 is the international standard recommended by ITU-T [35] for encoding telephone audio on 64 kb/s channel. It is a pulse code modulation (PCM) scheme. Basically, the analog telephone signal with maximum frequency of 3.4 kHz is sampled at about twice its highest frequency component, which is 8 kHz, with 8 bits per sample. This is because Nyquist theorem [36] states that exact reconstruction of a continuous-time baseband signal from its sample is possible if the sampling frequency is greater than twice the signal bandwidth.
1. INTRODUCTION

As human being is more sensitive to small change in low amplitude signal than in high amplitude signal, G.711 use this characteristic by encoding low amplitude signals using more bits than high amplitude signals. This non-linear quantization ensures that low amplitude signals will be well represented, while maintaining enough range to encode high amplitude signals.

G.711 consists of two versions: µ-law and A-law. The µ-law is used in North America and Japan, while the A-law is used in the rest of the world generally. The difference is in the method of the analog signal being sampled. In this thesis, G.711 µ-law is utilized as audio encoding scheme.

MPEG coding and compression scheme

MPEG stands for “Moving Picture Experts Group” [37], which works on standards for the coding of moving pictures and associated audio. For example, MPEG-1, MPEG-2 and MPEG-4 are major digital media standards that have been widely used today. MPEG-1 was designed back in 1993 for digital storage media format with bit rates up to about 1.5 Mb/s. MPEG-1 is used for the video CD (VCD) format and is the most compatible format in the MPEG standard; it is playable in almost all media player.

The basic idea behind MPEG video compression is to remove spatial redundancy within a video frame and a temporal redundancy between video frames. Discrete cosine transform (DCT) based compression is used for spatial redundancy while motion-compensation is used to exploit temporal redundancy.

Basically, MPEG-1 uses three types of picture: I, P, and B. These types of picture are arranged in specific order, and a group of these pictures is defined as the group of picture (GOP). The I-picture or intra picture is simply a frame coded as a still image, not using any past history. I-picture is independent of other picture types and each GOP begins with this type of picture. P-picture or predicted picture is predicted from the most recently reconstructed I or P picture. Lastly, B-picture or bidirectional picture is predicted from the closest preceding and following I or P pictures.

H.264

H.264 is the latest video standardization project of the ITU-T Video Coding Experts Group (VCEG) and the ISO/IEC MPEG. The main goals of this standardization effort
are to develop a simple and straightforward video coding design, with enhanced compression performance, and to provide a “network-friendly” video representation. H.264 has achieved a significant improvement in the rate-distortion efficiency – providing, typically, a factor of two in bit-rate savings when compared with existing standards such as MPEG-2 video. H.264 is popular for its use on Blu-ray Disc, HD DVD, and multimedia set-top box (STB).

The H.264 design covers a video coding layer (VCL), which efficiently represents the video content, and a network abstraction layer (NAL), which formats the VCL representation of the video and provides header information in a manner appropriate for conveyance by particular transport layers or storage media. The VCL of H.264 is similar to other standards such as MPEG-2 Video. It consists of a hybrid of temporal and spatial prediction, in conjunction with transform coding.

### 1.7 Media Synchronization

Audio and video have strict temporal structure to be preserved. In multimedia communications, audio and video streams transmitted through wireless networks will suffer from delay jitter, which ruins the audio-video temporal structure. This could leads into corrupted media synchronization.

Media synchronization is classified into intra-stream synchronization and inter-stream synchronization [38, 39]. The intra-stream synchronization controls the output time of MUs at the recipient to preserve temporal relationships between MUs within a single audio or video stream (Fig. 1.6). Here, MU stands for a “media unit”, which indicates the information unit for media synchronization at the application layer. The inter-stream synchronization is to synchronize between a audio MU and corresponding video MU (Fig. 1.7). In inter-stream synchronization, media can be classified into master media and slave media. Basically, the master media will first performs intra-stream synchronization and followed by slave media to sync itself to the corresponding master media performing inter-stream synchronization.

For example, let us denote the audio stream and video stream as media1 and media2, respectively. Additionally, the generation time of the n-th MU of media\(_i\) \((i=1,2)\) is \(T_n^{(i)}\), and that of output time is \(D_n^{(i)}\). Here, the intra-stream synchronization is to maintain the MU output interval \(D_{n+1}^{(i)} - D_n^{(i)}\) equal to the MU generation interval
1. INTRODUCTION

Figure 1.6: Intra-stream synchronization

Figure 1.7: Inter-stream synchronization

\[ T_{n+1}^{(i)} - T_{n}^{(i)} \]. On the other hand, inter-stream synchronization is to keep the MU output interval \( D_{n}^{(1)} - D_{m}^{(2)} \) the same as its MU generation interval \( T_{n}^{(1)} - T_{m}^{(2)} \).

1.8 Objectives and Thesis Structure

The performance of the HCCA in provisioning QoS guarantee for audio-video transmission has already been studied by many researchers. However, all these researches focus only on MAC-level QoS; that is, the QoS is assessed in terms of the throughput and delay of MAC frames. In multimedia communications, QoS at each level of the
1.8 Objectives and Thesis Structure

Protocol stack in the WLANs should also be considered. Reference [16] identifies six levels of QoS in IP networks: physical-level, link-level, network-level, end-to-end level, application-level, and user-level. In multimedia applications, application-level QoS and user-level QoS (QoE) should be examined since the ultimate goal of a multimedia service is to provide high user-level (perceptual) QoS for end-users, and application-level QoS is closely related to it.

In multimedia transmission with the HCCA, one of key issues for providing high QoS is the packet scheduling problem. The IEEE 802.11e standard has presented a reference design for an example scheduler, which is referred to as the TGe scheme as in [40], and an admission control scheme. The TGe scheme is suitable for CBR traffic only. Moreover, the TGe scheme has three disadvantages, which are mentioned below.

First, the TGe scheme cannot guarantee QoS for variable bit rate (VBR) traffic because it does not take the data rate and packet size fluctuation into account. Secondly, in the TGe scheme, the TXOP duration for each station is calculated on the assumption that the channel is error-free. However, transmission errors can occur in WLANs owing to shadowing, multipath fading, and interference. In this case, the TXOP duration becomes insufficient because channel capacity for retransmission traffic is not allocated. Thirdly, the derived number of arriving MSDUs calculated in the TGe scheme becomes smaller than the real number; this leads to insufficient channel allocation in the TGe scheme [41].

In this thesis, several packet scheduling schemes for audio-video transmission with the IEEE 802.11e HCCA are proposed to solve the three problems of the TGe scheme. The rest of the thesis is organized as follows.

Chapter 2 elaborates the static scheduling (SS) and multimedia priority dynamic scheduling (MPDS) schemes and compares them with the TGe scheme in terms of application-level QoS and QoE in an error-free transmissions. In the assessment of the SS scheme for CBR traffic, the minimum TXOP duration is examined under the condition that the application-level QoS and QoE are kept high. The target is to maximize the number of stations which can be admitted in a BSA. For VBR traffic, the improvement of the application-level QoS and QoE brought by the MPDS scheme is shown. This thesis evaluates the effect of the number of multimedia stations on the QoS because the QoS for the MPDS scheme depends on the remaining channel capacity for the additional TXOP duration.
1. INTRODUCTION

Chapter 3 discusses a packet scheduling scheme for transmission of CBR traffic over error-prone channel. This thesis tries to solve the second and third problems of the TGe scheme described earlier to support QoS of audio and video transmission by the IEEE 802.11e HCCA. To cope with the second problem, this thesis designs a simple strategy to allocate surplus bandwidth for MAC protocol data unit (MPDU) retransmission. In the proposed scheme, the AP allocates additional TXOP duration in an SI on the basis of the number of corrupted MPDUs in the previous SI. Furthermore, the proposed scheme calculates the number of MSDUs generated by a station in an SI, using the generation interval of audio samples and that of video frames at the application-layer to overcome the third problem.

Chapter 4 studies a cross-layer packet scheduling scheme to solve all the three problems of the TGe scheme under lossy channel conditions. In the proposed scheme, the HC first calculates basic TXOP for each station in an SI using the inter-arrival time of audio samples and that of video frames at the application layer to overcome the third problem. The AP then allocates additional TXOP in the SI to each station on the basis of the queue length of its source buffer. This TXOP allocation is useful to cope with the first and second problems. In addition, video frame skipping at the MAC-level of a source station is also proposed.

In Chapter 2 through Chapter 4, QoS mapping between application-level and user-level is performed by multiple regression analysis to estimate QoE from the application-level QoS parameters.

Finally, Chapter 5 concludes this thesis.
Chapter 2

Multimedia Priority Dynamic Scheduling Scheme

2.1 Introduction

The packet scheduling scheme for the HCCA has already been studied by many researchers [40, 42, 43, 44, 46, 47, 48, 49]. In [40], Grilo et al. propose the scheduling scheme based on estimated transmission time-earliest due date (SETT-EDD). In the SETT-EDD scheme, the HC allocates the TXOP duration to stations taking into account the deadline of each packet. In [42], Ansel et al. propose a scheduling scheme called FHCF, where the HC allocates additional TXOP duration after it calculates the TXOP duration based on the mean data rate. Reference [43] examines a scheduling scheme where the HC polls all stations and then it performs additional polling if there are some stations that require further channel allocation. An application-aware adaptive scheduling scheme has been examined in [44]. The scheduler of the scheme adapts SIs, polling order, and TXOPs depending on the traffic characteristics and instantaneous network conditions. References [45] and [46] propose polling schemes to reduce polling overhead by adopting variable SIs. In [47], a coordination function called isochronous coordination function (ICF) is proposed for efficient voice transmission with the HCCA. Admission control algorithms to support QoS requirement are studied in [45] and [48]. In [49], theoretical analysis of the HCCA is carried out.

From the above researches, no researches regarding QoE of audio-video transmission with the HCCA can be found. As a first step toward this type of study, this thesis
focus mainly on assessment of QoE; it also try to control QoE through achieving high application-level QoS. In this chapter, the static scheduling (SS) scheme and the multimedia priority dynamic scheduling (MPDS) scheme, which are designed for CBR traffic and VBR traffic, respectively are proposed [50, 51, 52, 53, 54]. The SS scheme is suitable for CBR traffic as it allocates fixed TXOP duration within an SI. Furthermore, the TXOP duration derived according to the SS scheme can be shorter than that of the TGe scheme; thus, the SS scheme increases the number of stations granted with TXOP.

In some cases where multimedia applications like videoconferencing produce VBR traffic, the SS scheme alone might not be able to keep its QoS at high level. The VBR traffic might generate data at higher rates than usual; it requires additional TXOP duration. Therefore, on the basis of the SS scheme, this thesis also proposes the MPDS scheme, which allocates additional TXOP duration only when the remaining channel capacity is available.

The effect of the number of multimedia stations and TXOP duration of the SS and MPDS schemes on application-level QoS is examined through simulation. The QoE of the two schemes is also evaluated by a subjective experiment. This chapter evaluates the application-level QoS and QoE by utilizing MPEG-1 video and encode it into two types of picture pattern; I and IPPPPP, each of which has three different bit rates. The former represents the CBR traffic while the latter represents the VBR traffic. In the assessment of the SS scheme for CBR traffic, we examine the minimum TXOP duration under the condition that the application-level QoS and QoE are kept high. We aim to maximize the number of stations which can be admitted in a BSA. For VBR traffic, we show the improvement of the application-level QoS and QoE brought by the MPDS scheme. We evaluate the effect of the number of multimedia stations on the QoS because the QoS for the MPDS scheme depends on the remaining channel capacity for the additional TXOP duration.

In the QoE assessment, we consider a video stream and the corresponding audio stream together since cross-modal influences between audio and video affects the overall perceptual quality [15, 20]. Since QoE is directly related to human perception, we utilize a psychometric method referred to as the method of successive categories [20].

The rest of the chapter is organized as follows. Section 2.2 explains the proposed packet scheduling schemes. Section 2.3 specifies experimental methodology.
2.2 Proposed Scheduling Schemes

In this section, we introduce the two proposed packet scheduling for HCCA: SS, and MPDS. We consider multimedia stations and data stations in a BSS. A multimedia station transmits a pair of audio and video flows to the AP. A data station transmits a single flow of random data to the AP to interfere with the transmission of the multimedia traffic.

2.2.1 SS scheme

In order to determine the minimum TXOP duration required to keep the QoS high, we propose the SS scheme which suitable for CBR traffic.

![Diagram of TXOP allocation in the SS scheme]

Figure 2.1: Procedure of TXOP allocation in the SS scheme

Similar to the TGe scheme, the SS scheme also statically allocates the duration of TXOP to each station. However, unlike the TGe scheme, the TXOP duration in the SS scheme is calculated on the basis of the product of the mean data rate and a parameter $\alpha$; therefore, in the SS scheme, the HC can allocate longer or shorter TXOP duration by setting the value of $\alpha$ accordingly. Figure 2.1 illustrates the procedure of TXOP allocation in the SS scheme. The TXOP duration for flow $i$ is calculated as $TXOP_{i}^{SS} = \alpha \times TXOP_{i}^{TGe}$ where the value of $\alpha$ smaller than 1 gives shorter TXOP duration than that in the TGe scheme, while the value of $\alpha$ larger than 1 gives longer.
2. MULTIMEDIA PRIORITY DYNAMIC SCHEDULING SCHEME

TXOP duration. The superscript $SS$ means the SS scheme. When $\alpha = 1$, the SS scheme is equal to the TG scheme. By setting the $\alpha$ value smaller than 1, we can study whether allocating TXOP duration less than that of the TG scheme can degrade the QoS level. In the SS scheme, the HC can perform admission control utilizing an equation obtained by replacing $TXOP^{TG}$ with $TXOP^{SS}$ in Eq. (1.7) in Chapter 1.

2.2.2 MPDS scheme

The MPDS scheme is proposed to handle traffic with VBR characteristics because the SS scheme alone cannot absorb the burstiness of this traffic.

In the MPDS scheme, the HC scheduler allocates additional TXOP duration to stations for transmission of only audio and video packets after it calculates the TXOP duration based on the product of the mean data rate and $\alpha$. That is, priority is given to audio-video transmission over data transmission. The procedure of TXOP allocation in the MPDS scheme is shown in Fig. 2.2.

Let us describe the algorithm of the MPDS scheme. First, as usual, the HC computes the basic TXOP duration of flow $i$, which is denoted by $basicTXOP_i^{MPDS}$, on the basis of TSPEC information sent by the station. The superscript $MPDS$ denotes the MPDS scheme. The calculation is performed in the same way as that in the SS scheme.

Secondly, the HC uses the QoS control field of the IEEE 802.11e MAC header to record the queue length of the audio buffer and that of the video buffer of the station at the end of the TXOP. This queue length means the number of MPDUs which could not be transmitted during the current SI because of the insufficient TXOP duration. This case usually happens when VBR traffic is transmitted. Note that the scheduler records the queue length only for audio and video traffic because these kinds of traffic require strict QoS guarantee. In this scheme, the queue length for data traffic is not estimated. The HC uses the traffic identifier (TID) of the QoS Control field in the MAC header to distinguish the flow types in the queue.

After obtaining the queue length record during the previous SI and then computing the basic TXOP duration, the HC computes the additional TXOP duration required for transmission of MPDUs left in the queue as follows:

$$addTXOP_i^{MPDS} = \frac{queue_i \times L_i}{R_i}$$

(2.1)
Figure 2.2: Procedure of TXOP allocation in the MPDS scheme
where $queue_i$ is the number of MPDUs in the (audio or video) queue of flow $i$. Then, the HC adds the additional TXOP duration to the basic TXOP:

$$TXOP_i^{MPDS} = basicTXOP_i^{MPDS} + addTXOP_i^{MPDS}$$

(2.2)

Note that there is no additional TXOP for data stations, which obtain the basic TXOP duration only.

If the sum of the TXOP duration for multimedia and data stations is smaller than the duration of an SI, the TXOP duration for multimedia stations can be increased proportionally so as to use up the remaining time. Otherwise, the HC reduces the TXOP duration of each station proportionally until the sum of all TXOP duration is equal to or lower than the SI. However, the reduced TXOP duration must not be lower than the basic TXOP duration; that is each station is guaranteed with basic TXOP duration. In contrast to the SS scheme, where the TXOP duration can be reduced to accommodate additional stations, the MPDS scheme takes an advantage of available bandwidth to accommodate VBR traffic when the number of stations is small.

In the MPDS scheme, admission control can be carried out with an equation obtained by replacing $TXOP_i^{TG}$ with $basicTXOP_i^{MPDS}$ in Eq. (1.7).

From the explanation above, the SS scheme is easy to be implemented, but it cannot handle the burstiness of VBR traffic. Therefore, the SS scheme is suitable for CBR traffic and can maximize the number of admitted stations by reducing the TXOP duration of each station. In contrast, the MPDS scheme is more complex to implement, but it performs well for VBR traffic transmission during low traffic condition as additional TXOP can be allocated to multimedia stations.

### 2.3 Experimental Methodology

In this chapter, the application-level QoS is assessed by simulation with ns-2 (network simulator version 2) [55, 56, 57]. Then, the QoE is assessed by subjective experiment where we actually output the audio and video flows according to the output time-stamps obtained from the simulation. In this section, we first present the simulation conditions used for the application-level QoS assessment. We then elaborate on the methods of subjective experiment.
2.3.1 Simulation conditions

Figure 2.3 illustrates the system configuration used in the simulation. We focus on a single BSS which includes an AP, four data stations, and a various number of multimedia stations. We assume that there is no movement of wireless stations during the simulation. We use the IEEE 802.11b physical layer based on DSSS with a channel data rate of 11 Mb/s. In the simulation, we assume error-free transmission, and three types of traffic are considered: audio, video, and random data. Each multimedia station sends a pair of audio and video flows to the HC as two separate transport streams using UDP/IP. Each data station generates UDP datagrams of 1472 bytes each in its payload at exponentially distributed intervals and sends them to the HC. We also assume that the average load per data station is 1 Mb/s.

![System configuration diagram]

Table 2.1 summarizes media specifications of audio-video flows used in the simulation. We use an audio flow of ITU-T G.711 μ-law and an MPEG-1 video flow. A video MU is defined as a video frame and is transferred as one or more UDP datagrams. An audio MU consists of 1000 audio samples, which corresponds to a single UDP datagram.

Three types of the contents are used in the simulation: Music video, Sport, and Movie. The Music video shows scenes of a Japanese female singer dancing with background dancers. For Sport, scenes of an F1 race with a commentator’s voice have been
Table 2.1: Specifications of audio and video

<table>
<thead>
<tr>
<th></th>
<th>Audio</th>
<th>Video</th>
</tr>
</thead>
<tbody>
<tr>
<td>coding scheme</td>
<td>G.711 μ-law</td>
<td>MPEG-1</td>
</tr>
<tr>
<td>image size [pixel]</td>
<td>–</td>
<td>320 × 240</td>
</tr>
<tr>
<td>picture pattern</td>
<td>–</td>
<td>I</td>
</tr>
<tr>
<td></td>
<td></td>
<td>IPPPPP</td>
</tr>
<tr>
<td>average MU rate [MU/s]</td>
<td>8</td>
<td>20</td>
</tr>
<tr>
<td>average inter-MU time [ms]</td>
<td>125</td>
<td>50</td>
</tr>
<tr>
<td>average bit rate [kb/s]</td>
<td>64</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>800</td>
</tr>
<tr>
<td></td>
<td></td>
<td>1000</td>
</tr>
<tr>
<td>measurement time [s]</td>
<td></td>
<td>20</td>
</tr>
</tbody>
</table>

chosen. The Movie is a Japanese film with scenes of a woman having discussion with her friends. Here, we have encoded each video flow into two picture patterns of I and IPPPPP, each at three bit rates of 600 kb/s, 800 kb/s, and 1000 kb/s. Here, the video with picture pattern of I represents the CBR traffic while the video with picture pattern of IPPPPP represents the VBR traffic in our research. The difference in quality among these video flows is hardly noticeable. Note that we use only one type of video flow in each simulation.

In the simulation, we set the beacon interval to 500 ms. For the TSPEC parameters, we set the MSI’s for all flows to 50 ms, and the nominal MSDU size for audio, that for video, and that for data are set to 1000 bytes, 1500 bytes, and 1500 bytes, respectively. We assume that each source buffer at the MAC layer in a station or an AP can accommodate a maximum of 50 MPDUs for each flow and that a newly generated MPDU is discarded unless enough space to accommodate it is available.

2.3.2 Methods for subjective experiment

The QoE assessment performed in our study is done according to part of the methodology recommended in [58] and [59]. The subjective assessment was conducted by 30 students; their ages were 20s. In each simulation, we have six samples by changing
the value of $\alpha$. The number of samples each assessor assessed is 216, since for each value of $\alpha$ we used two packet scheduling schemes, two picture patterns, three video contents, and three bit rates. We fix the number of multimedia stations to 4. Note that we use only the SS and the MPDS schemes in the assessment because the results for the TGe scheme is equivalent to the result of the SS scheme when the value of $\alpha$ is equal to 1.00. For the assessment, we use 17-inch liquid crystal displays (LCD), and the distance between the assessors and the display is about 50-70 cm. The assessors listen to the audio output using a headphone. The assessors were asked to classify the test samples into a certain number of categories each assigned an integer. Here, we use five categories of impairment of the rating-scale method: "imperceptible" assigned integer 5, "perceptible, but not annoying" 4, "slightly annoying" 3, "annoying" 2, and "very annoying" 1.

2.4 Application-level QoS Assessment

In this section, we show simulation results of application-level QoS assessment of audio-video transmission with the TGe, the SS, and the MPDS schemes.

First, we compare the three schemes by changing the value of $\alpha$ used in the calculation of TXOP duration for video flows. We study whether allocating TXOP duration in the SS scheme less than that of the TGe scheme can degrade the QoS. From this simulation, we can find the minimum TXOP duration needed by the video flow to keep its QoS high. We also evaluate the performance of the MPDS scheme.

Second, we compare the TGe scheme and the MPDS scheme by changing the number of multimedia stations. This is because we should examine how much improvement the MPDS scheme can give in the QoS of audio and video flows by means of the allocation of additional TXOP. Note that the MPDS scheme uses the remaining channel capacity to allocate additional TXOP and that as the number of multimedia stations increases, the remaining channel capacity becomes smaller.

2.4.1 The effect of static TXOP duration for video

Here, we compare the three schemes by changing the value of $\alpha$ used in the calculation of TXOP duration for video flows. Note that in the SS scheme $\alpha$ less than 1.00 derives
shorter TXOP duration than that of the TGe scheme. We study whether allocating TXOP duration (basic TXOP duration in the case of the MPDS scheme) less than that of the TGe scheme can degrade the application-level QoS. We also look into the effects of the video picture pattern, bit rate, and content type on the QoS. This is because different video specifications produce different bit rate distributions and also require different TXOP duration. In the following simulation results, we set the number of multimedia stations to 4.

Figures 2.4 through 2.12 show simulation results of application-level QoS assessment of audio-video transmission with the TGe, the SS, and the MPDS schemes. In this figure, notation “SS-I-600”, for instance, refers to the result for the SS scheme in the case of the I picture pattern at the bit rate of 600 kb/s. Each figure consists of simulation results of the two schemes for all combinations of the picture patterns and bit rates. We use the video with the I and IPPPPP picture patterns to imitate CBR traffic and VBR traffic, respectively. The value of $\alpha$ used in the simulations is set to one ranging from 0.75 to 1.05. Note that the results for the TGe scheme are equivalent to those of the SS scheme with $\alpha = 1.00$.

In the case of audio flows, we set the value of $\alpha$ to 1.00 for the calculation of the TXOP duration. For this reason, the simulation results of the audio flows do not show significant difference between all the three schemes. Therefore, we do not show the simulation results for the audio flows in this chapter.

2.4.1.1 The SS scheme for CBR traffic

First, we discuss the performance of the SS scheme in the case of CBR traffic, namely, the I picture pattern.

Figures 2.4 through 2.6 show the average MU delay for Music video, Sport and Movie as a function of $\alpha$, respectively. From these figures, we can observe that the average video MU delay deteriorates with the decrement of the $\alpha$. For video content of Movie, however, it remains lower even if the $\alpha$ is reduced to 0.80.

Similar results can also be seen in Figs. 2.7 through 2.12, which indicates the MU loss ratio and mean square error of inter-stream synchronization for video, respectively. For video with the I picture pattern, the MU loss ratio starts to increase only when the $\alpha$ is set to below 0.80. This is because even if the TXOP duration for video flows is
2.4 Application-level QoS Assessment

reduced, the video MUs are stored temporarily at the source buffer before being sent in the next SI. This shows that the SS scheme keeps the application-level QoS at high level for CBR traffic by using shorter TXOP duration than that of the TGc scheme.

These figures also indicate that the video flows with lower bit rates are more affected by the value of $\alpha$ than the ones with higher bit rates. This is because the ceiling function used in Eq. (1.5) gives more extra TXOP duration for a high bit rate than for a low bit rate.

2.4.1.2 The SS scheme for VBR traffic

We then discuss the SS scheme in the case of VBR traffic, namely, the IPPPPP picture pattern.

In this case, the average MU delay, MU loss ratio and mean square error of inter-stream synchronization degrade drastically when the value of $\alpha$ is reduced. Especially, the MU loss ratio increases drastically when $\alpha$ is smaller than 0.90.

Moreover, the MU loss ratios for lower bit rates are higher than the ones for higher bit rates in the case of the IPPPPP picture pattern when the value of $\alpha$ is large. However, the results turn oppositely for Music video and Sport when the value of $\alpha$ becomes less than 0.85. This is due to lack of transmission time for the video with higher bit rates compared to the video with lower bit rates when the value of $\alpha$ is too small.

From these results, we see that the SS scheme is not efficient to be used with the VBR traffic.

2.4.1.3 The MPDS scheme

From the simulation results of the SS scheme, we have noticed that this scheme is inefficient and cannot keep high QoS for VBR traffic. For this type of traffic, we propose the MPDS scheme. Figures 2.4 through 2.6 show that the average MU delay for video is kept low regardless the value of $\alpha$. We can see in these figures that no MU loss is observed and that the mean square error of inter-stream synchronization is kept almost constant below the threshold value ($6400 \text{ ms}^2$) for high-quality lip sync [15] regardless of the value of the static video TXOP duration. This is because the MPDS scheme utilizes available remaining channel capacity for additional TXOP duration allocation.
so that it can absorb the burstiness of VBR traffic even when the basic TXOP duration is reduced. However, when the number of multimedia stations increases and no more remaining channel capacity is available, the MPDS scheme is comparable to the SS scheme.

In Figs. 2.4 through 2.6, we have not shown the average MU delay for video when \( \alpha < 0.75 \) or \( \alpha > 1.05 \). The average MU delay for the MPDS scheme is kept low even if \( \alpha < 0.75 \) or \( \alpha > 1.05 \). This is because enough TXOP duration is allocated to each multimedia station, though the basic TXOP duration increases and the additional TXOP duration decreases as \( \alpha \) becomes bigger. In the case of video flows with average bit rate of 800 kb/s, for example, when \( \alpha > 2.30 \), not all four multimedia stations can be admitted in the MPDS scheme.

![Figure 2.4: Average MU delay for video versus \( \alpha \) (Music video)](image-url)
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Figure 2.5: Average MU delay for video versus $\alpha$ (Sport)

Figure 2.6: Average MU delay for video versus $\alpha$ (Movie)
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Figure 2.7: The MU loss ratio for video versus $\alpha$ (Music video)

Figure 2.8: The MU loss ratio for video versus $\alpha$ (Sport)
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Figure 2.9: The MU loss ratio for video versus $\alpha$ (Movie)

Figure 2.10: Mean square error of inter-stream synchronization versus $\alpha$ (Music video)
2. MULTIMEDIA PRIORITY DYNAMIC SCHEDULING SCHEME

Figure 2.11: Mean square error of inter-stream synchronization versus $\alpha$ (Sport)

Figure 2.12: Mean square error of inter-stream synchronization versus $\alpha$ (Movie)
2.4.2 The effect of the number of multimedia stations

Let us evaluate the effect of the number of multimedia stations on application-level QoS. In this simulation, we compare two scheduling schemes: the TGe and the MPDS schemes. In the MPDS scheme, we set $\alpha = 1.00$; in this case the basic TXOP duration in the MPDS scheme is equal to the TXOP duration in the TGe scheme. Note that the results for the TGe scheme are the same as those for the SS scheme when $\alpha$ is equal to 1.00.

Figure 2.13 through 2.15 show the average video MU delay as a function of the number of multimedia stations for the three contents. These figures show that the MPDS scheme outperforms the TGe scheme if the number of multimedia stations is small. Especially, the difference in the average video MU delay between the two scheduling schemes becomes larger as the number of multimedia stations decreases. This is because when the number of multimedia stations becomes smaller, the remaining time for the additional TXOP in the MPDS scheme increases.

Note that the MPDS scheme is efficient only when extra bandwidth is available. The MPDS scheme can absorb burstiness of VBR traffic by allocating additional TXOP; thus, it reduces the MU output delay. The performance of the MPDS scheme deteriorates with the increase of the number of multimedia stations where the results become comparable to or better than that of the TGe scheme. When extra bandwidth is not available, the MPDS scheme works just as the SS and TGe scheme.

Meanwhile, in the case of the TGe scheme, the average video MU delay is not affected by the number of multimedia stations as the HC allocates constant TXOP duration. Note that under the simulation conditions, at most 8 multimedia stations can be admitted in the TGe and the MPDS schemes when the admission control is carried out. If the number of multimedia stations is more than eight, only the first eight stations will be admitted, while the admission requests of the rest of the multimedia stations will be rejected because there is no enough bandwidth available.

It should be noted that we assume in our simulation that all channel capacity is used for the CFP. Under a mixture of the HCCA and EDCA traffic, the admitted number of multimedia stations will be decreased because the channel capacity for the HCCA becomes smaller.
Furthermore, we have confirmed through simulation that the MPDS scheme can also improve the mean square error of inter-stream synchronization compare to the TGe scheme.

Figure 2.13: Average MU delay for video versus the number of multimedia stations (Music video)
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Figure 2.14: Average MU delay for video versus the number of multimedia stations (Sport)

Figure 2.15: Average MU delay for video versus the number of multimedia stations (Movie)
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2.5 QoE Assessment

In this section, we first assess QoE (i.e., user-level QoS) of the TGc, the SS, and the MPDS schemes by a subjective experiment. From the assessment of the application-level QoS, we found that the video quality of the SS scheme deteriorates when the $\alpha$ value for the video flow is reduced. However, through application-level QoS only, it is hard to make a conclusion that the QoE is good or not. This is because the results of the MU delay, the MU loss ratio and the mean square error of inter-stream synchronization each produce different degree of QoS degradation. Therefore, we examine the effect of the $\alpha$ value of the SS and MPDS schemes on QoE. In the QoE assessment, we utilize the method of successive categories.

We will then present a method for determining an appropriate value of $\alpha$.

2.5.1 Psychological scale

In this chapter, the results obtained from the rating-scale method are calculated into the interval scale with the law of categorical judgment.

To verify the obtained interval scale, we have performed Mosteller’s test. From the Mosteller’s test, after removing some values, we cannot reject the hypothesis that the obtained interval scale fits the observed data at a significance level of 0.05. Thus, we refer to the interval scale as the psychological scale.

Figures 2.16 through 2.21 depict the measured psychological scale for Music video, Sport, and Movie, respectively. In these figures, the number of multimedia stations is set to 4. We have selected the minimum value of the psychological scale as the origin. Note that, in the interval scale, we can select an arbitrary origin and any unit of scale. In these figures, each of the four horizontal dotted lines indicates the lower boundary of a category.

From these figures we find that in the case of the I picture pattern, the psychological scale for the SS scheme decreases if the value of $\alpha$ is set lower than 0.85 for Music video and 0.80 for the Sport and Movie. We can say that in the cases of the SS scheme for the three contents, about 85% of the TXOP duration in the TGc scheme is enough to obtain approximately the highest QoE, especially when we use the contents with a CBR characteristic. However, the QoE for the IPPPPP picture pattern degrades more rapidly with the decrement of the $\alpha$ value. This means that the SS scheme performs
well only for transmission of video with the I picture pattern. Moreover, the video flows with lower bit rates are more affected by the value of $\alpha$ than the ones with higher bit rates.

We also find that the MPDS scheme keeps the QoE high for the three contents on any conditions. This is due to the allocation of additional TXOP duration performed by the MPDS scheme. This shows that the MPDS scheme is suitable for transmission of video with the IPPPPP picture pattern. As the number of multimedia stations is set to 4, extra bandwidth is always available. This is why the QoE of the MPDS scheme performs well. If the number of multimedia stations increases, QoE for the MPDS scheme degrades as the available extra bandwidth decreases; even in this case, the MPDS scheme is comparable to the SS scheme.

Figure 2.22 shows the admitted number of multimedia stations in the SS scheme as a function of $\alpha$, which is equivalent to the static TXOP duration for video. From Fig. 2.22, we observe that when the $\alpha$ value is set to 0.80, the maximum number of multimedia stations that can be accommodated by the HC with the admission control is 10, while at most 8 multimedia stations can be admitted in the TGc scheme. As seen from the results of QoE, the TXOP duration obtained when $\alpha$ is equal to 0.80 achieves relatively high QoE especially in the case of video with the I picture pattern.

Thus, we can say that the SS scheme can support a larger number of multimedia stations as a BSS than the TGc scheme under the condition that the QoE is kept high for video with the I picture pattern.
Figure 2.16: Psychological scale (Music video - SS scheme)

Figure 2.17: Psychological scale (Music video - MPDS scheme)
2.5 QoE Assessment

Figure 2.18: Psychological scale (Sport - SS scheme)

Figure 2.19: Psychological scale (Sport - MPDS scheme)
2. MULTIMEDIA PRIORITY DYNAMIC SCHEDULING SCHEME

Figure 2.20: Psychological scale (Movie - SS scheme)

Figure 2.21: Psychological scale (Movie - MPDS scheme)
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![Graph showing admitted number of multimedia stations versus α]

Figure 2.22: Admitted number of multimedia stations versus α

2.5.2 Determining an appropriate value of α

So far, we have discussed the cases of given values of α. In order to take advantage of reducing the TXOP duration while keeping the QoE high, we need a method for determining an appropriate value of α.

In the case of CBR traffic, we found that the SS scheme can guarantee high QoE with shorter TXOP duration than that of the TGe scheme. To determine an appropriate value of α, one method we can easily consider is eliminating the ceiling function used in the calculation of the TXOP duration. This method derives an α value of 0.83. From our QoE assessment, we found that the TXOP duration for α = 0.83 achieves high QoE for Sport and Movie in the case of the I picture pattern. For Music video, the QoE is not highly degraded.

Meanwhile, in the case of VBR traffic, we recommend α = 1.00 in the MPDS scheme. In most cases of our QoE assessment, the QoE of the IPPPPP picture pattern for the MPDS scheme is equivalent to or better than that of the TGe scheme. Furthermore, QoE for the MPDS scheme depends on the number of multimedia stations. We have
found in Figs. 2.13 through 2.15 that the application-level QoS for $\alpha = 1.00$ in the MPDS scheme is also equivalent to or better than that for the TGs scheme even if the number of multimedia stations is eight. However, the QoE for the MPDS scheme can be improved for $\alpha$ greater than 1.00. The finding of the optimum value of $\alpha$ to keep the QoE high should be for further study.

2.6 QoS Mapping

Next, we perform QoS mapping between application-level QoS and QoE with multiple regression analysis. In the analysis, the application-level QoS parameters are considered as the predictor variables while the QoE parameter is considered as the criterion variable. 11 application-level QoS parameters are used in the QoS mapping as defined in Chapter 1.

As a result of the principal component analysis, we adopted the first two principal components since the cumulative contribution rate for the first two principal components becomes 92.7%. Therefore we can classify the application-level QoS parameters into two groups. We then performed multiple regression analysis of all combinations of the predictor variables. The predictor variables of the adopted combination were statistically tested whether they make significant contributions to the multiple regression line. As a result, we remove the predictor variables that belong to audio because we only change the TXOP duration for video only. Finally, we found combination of the application-level QoS parameters which make the highest contribution rate adjusted for degree of freedom.

**group A** $R_v, C_v, L_v$

**group B** $D_v, E_v, E_{int}$

Here, the estimated value of the psychological scale $\hat{S}$ for each video content and its bit rate are shown below. In the equation to be shown, $R^*$ refers to the contribution rate adjusted for degrees of freedom.

$$\hat{S}_{music}^{600} = 4.462 - 0.559C_v - 0.003D_v , \quad R^* = 0.991 \quad (2.3)$$
In the above equation, its subscript means content type, while its superscript means the average video bit rates. From these equations, we found that the QoE decreases with the increase of $C_v$ and $D_v$. By using these equations, we can estimate QoE from application-level QoS parameters when we change the value of $\alpha$ for video flow.

Figures 2.16 through 2.21 also show the estimated psychological scale derived from Eq. (2.3) through Eq. (2.11). From these figures, we notice that the estimated psychological scale values have the same tendency as the measured ones. This is because the contribution rate adjusted for degrees of freedom for Eq. (2.3) through Eq. (2.11) are near to 1.0. That means the equations can estimate the psychological scale values with high accuracy.
2. MULTIMEDIA PRIORITY DYNAMIC SCHEDULING SCHEME

2.7 Conclusions

In this chapter, we have proposed the SS and MPDS schemes and compared them with the TGc scheme in terms of application-level QoS and QoE. We also performed the QoS mapping between the application-level QoS parameters and QoE parameter with multiple regression analysis. Three types of content were used in the simulation: Music video, Sport, and Movie. They have been encoded in two types of picture pattern: I and IPPPPP, each at three different bit rates. We studied the effect of the picture patterns, bit rates, and content types on QoS. We also examined the effect of TXOP duration on the QoS.

In the SS scheme, we have learned that the QoE for video with the I picture pattern is kept high even when the TXOP duration for video flow is reduced. This shows that it is possible to guarantee QoE with less bandwidth; so more flows can be admitted and served with QoS guarantee. However, the QoS level for the video with the IPPPPP picture pattern deteriorates with the decrease of the value of $\alpha$. This shows that the SS scheme is not suitable for the VBR traffic transmission.

The MPDS scheme outperforms the SS and TGc schemes when the number of stations is small. This shows that the MPDS scheme is suitable for transmission of video with the IPPPPP picture pattern. However, even if the performance of the MPDS scheme deteriorates with the increase of the number of stations, the results are comparable to or better than that of the SS and TGc schemes.
Chapter 3

QoE Guarantee in Error-Prone Channel: Part I

3.1 Introduction

From the analysis of application-level QoS and QoE assessment in the previous chapter, we notice that the TG scheme has three disadvantages.

First, the TG scheme cannot guarantee QoS for VBR traffic because it does not take the data rate and packet size fluctuation into account.

Secondly, in the TG scheme, the TXOP duration for each station is calculated on the assumption that the channel is error-free. However, transmission errors can occur in WLANs owing to shadowing, multipath fading, and interference. In this case, the TXOP duration becomes insufficient because channel capacity for retransmission traffic is not allocated. In wireless multimedia transmission, packet scheduling should be designed so as to accommodate retransmission traffic as well as newly generated traffic to improve QoS in a noisy environment.

Thirdly, the derived number of arriving MSDUs calculated from Eq. (1.5) in Chapter 1 becomes smaller than the real number; this leads to insufficient channel allocation in the TG scheme. For example, in the case of video flow used in our study, one video frame is generated in every 50 ms, the mean date rate $\rho$ is 800 kbps and nominal MSDU size $L$ is 1500 bytes; therefore, the video frame is fragmented into 4 MSDUs. Thus, 8 MSDUs are generated in an SI when SI is 100 ms. On the other hand, according to Eq. (1.5), the number of arriving MSDUs becomes 7 MSDUs if SI is 100 ms. Note that
this problem occurs since the derived number of arriving MSDUs in an SI is calculated from only MAC-level parameters, though the real number of arriving MSDUs highly depends on video frame rate at the application-layer.

There are numerous studies trying to improve the inefficiency of the TGc scheme. In [40, 42, 60, 61, 62], various methods are proposed to improve QoS for transmission of VBR traffic. However, these studies have been focusing only on MAC-level QoS; that is, the QoS is assessed in terms of the MAC-level throughput and MAC frame delay. In addition, references [40] and [62] examine MAC-level QoS considering channel transmission error. However, these studies have not explicitly discussed surplus bandwidth allocation for retransmission traffic.

With regard to QoS at upper levels, reference [63] examines the application-level QoS of video transmission with a resource allocation scheme in terms of peak signal-to-noise ratio (PSNR). In the scheme, one video flow is sent as more than one MAC flow. In addition, Chapter 2 proposes the MPDS scheme for transmission of VBR traffic and have assessed application-level QoS and QoE in the case where audio and video are transferred from stations to the AP in a error-free channel.

In this chapter, we try to solve the second and third problems of the TGc scheme described earlier to support QoS of audio and video transmission by the IEEE 802.11e HCCA. We focus on evaluation of CBR traffic by utilizing video with I-picture pattern with CBR characteristic. To cope with the second problem, we design a simple strategy to allocate surplus bandwidth for MPDU retransmission in the TGc scheme. In the proposed scheme, the HC allocates additional TXOP duration in an SI on the basis of the number of corrupted MPDUs in the previous SI. Furthermore, the proposed scheme calculates the number of MSDUs generated by a station in an SI, using the generation interval of audio samples and that of video frames at the application-layer to overcome the third problem.

We then evaluate the application-level QoS and QoE of the proposed scheme and the TGc scheme. Application-level QoS is important in multimedia transmission since multimedia flows like audio and video have temporal structure to be preserved. In this chapter, the media synchronization quality is regarded as the major part of the application-level QoS. We examine the effect of maximum number of retransmission on the QoS and discuss the superiority of the proposed scheme to the TGc scheme. We also show the relationships between the number of stations and the application-level
QoS to discuss an admission control scheme in a noisy environment. Furthermore, we also examine the effect of SI on the QoS and QoE in the proposed scheduling scheme since they can be highly affected by the polling interval.

The rest of the chapter is organized as follows. In Section 3.2, we present the proposed scheme. Next in Section 3.3, we specify the experimental methodology. Section 3.4 presents numerical results of application-level QoS. The results of QoE assessment are presented in Section 3.5. Section 3.6 shows QoS mapping between the application-level QoS parameters and QoE. Finally, Section 3.7 concludes this chapter.

### 3.2 Proposed Scheduling Scheme

In this chapter, we propose a method of allocating additional TXOP duration. The proposed scheme is designed to fulfill insufficient TXOP duration in the event of transmission error. The procedure of TXOP allocation in the proposed scheme is shown in Fig. 3.1.

![Figure 3.1: Procedure of TXOP allocation in the proposed scheme](image)
In the proposed scheme, the HC first calculates the basic TXOP duration in a similar way to that of the TGc scheme but with replacement of Eq. (1.5). In order to derive the number of MSDUs generated in an SI accurately, we propose a new method for calculating the number of MSDUs.

In the proposed method, information about \(\text{inter-MU (media unit)}\) time is required. This is because the video frame is generated periodically and fragmented into several MSDUs. Therefore, the calculation based on an inter-MU time can give an accurate value of the number of MSDUs. Using the \(\text{inter-MU}\) time together with the TSPEC, we first compute the number of MSDUs arriving within an \(\text{inter-MU}\) time as

\[
n_i^P = \left\lfloor \frac{\text{inter-MU}_i \times \rho_i}{L_i} \right\rfloor
\]

where \(\text{inter-MU}_i\) denoted the inter-MU time for flow \(i\). The superscript \(P\) means the proposed scheme. Then, the number of MSDUs that arrives in an SI is computed as

\[
N_i^P = \left\lfloor \frac{\text{SI}}{\text{inter-MU}_i} \times n_i^P \right\rfloor
\]

Then, the basic TXOP duration is calculated with Eq. (1.5) by replacing \(N_i^{TGe}\) with \(N_i^P\).

On top of the basic TXOP duration, the HC then allocates additional TXOP duration. The HC keeps monitoring the number of corrupted MPDU owing to transmission errors in every SI. In the event of transmission error, the HC does not acknowledge the corrupted MPDU, and the retransmission counter of the corresponding station will be incremented. If the HC has received \(E\) (say) corrupted MPDUs from a station in an SI, the HC allocates additional TXOP duration to the station for retransmission of \(E\) MPDUs with the nominal MSDU size in the next SI. If the HC receives corrupted MPDUs, it cannot distinguish the flow types between audio and video. For this reason, the HC allocates additional TXOP duration for transmission of MPDUs with the nominal MSDU size of the video flow, which has a bigger MSDU size than that of audio.

As the surplus bandwidth reduces with the increase of admitted flows, the HC might not be able to allocate enough additional TXOP duration. In this case, the HC distributes the surplus bandwidth for stations in a round-robin basis; that is, additional

---

1The transmission unit of data at the application layer. A video MU is defined as a video frame and is transferred as one or more UDP datagrams.
3.3 Experimental Methodology

TXOP duration for retransmission of one nominal-sized MSDU will be allocated to each station until the surplus bandwidth is fully occupied. Note that the maximum additional TXOP duration for a station in an SI is the duration to transmit $E$ MPDUs.

3.3 Experimental Methodology

In this section, we first present the simulation conditions used for the application-level QoS assessment. We then elaborate on the methods of subjective experiment used in the QoE assessment.

3.3.1 Simulation conditions

Figure 3.2 illustrates the system configuration used in the simulation. We focus on a single BSS which includes an AP and several multimedia stations. We assume the IEEE 802.11b physical layer based on DSSS with a channel data rate of 11 Mb/s. No movement of wireless stations is assumed during the simulation. In the simulation, two types of traffic are considered: audio and video. Each multimedia station sends stored audio and video streams to the AP as two separate transport streams using UDP/IP.

![Figure 3.2: System Configuration](image-url)
3. QOE GUARANTEE IN ERROR-PRONE CHANNEL: PART I

Table 3.1 summarizes media specifications of audio-video flows used in the simulation. We use an audio flow of ITU-T G.711 μ-law and an H.264 video flow. A video MU is defined as a video frame and is transferred as one or more UDP datagrams. An audio MU consists of 1000 audio samples, which corresponds to a single UDP datagram. The audio bit rate is constant at 64 kb/s, and the video bit rate is 800 kb/s on average.

We use Music video and Sport as the video content. The Music video shows scenes of a Japanese female singer dancing while singing. For the Sport, scenes of a football match with a commentator's voice have been chosen.

Table 3.1: Specifications of audio and video

<table>
<thead>
<tr>
<th></th>
<th>Audio</th>
<th>Video</th>
</tr>
</thead>
<tbody>
<tr>
<td>coding scheme</td>
<td>G.711 μ-law</td>
<td>H.264</td>
</tr>
<tr>
<td>image size [pixel]</td>
<td>–</td>
<td>320 × 240</td>
</tr>
<tr>
<td>picture pattern</td>
<td>–</td>
<td>I</td>
</tr>
<tr>
<td>average MU size [byte]</td>
<td>1000</td>
<td>5000</td>
</tr>
<tr>
<td>average MU rate [MU/s]</td>
<td>8</td>
<td>20</td>
</tr>
<tr>
<td>average inter-MU time [ms]</td>
<td>125</td>
<td>50</td>
</tr>
<tr>
<td>average bit rate [kb/s]</td>
<td>64</td>
<td>800</td>
</tr>
<tr>
<td>measurement time [s]</td>
<td>60</td>
<td></td>
</tr>
</tbody>
</table>

In the simulation, we set the beacon interval to 1000 ms. The nominal MSDU size for audio and that for video are set to 1000 bytes and 1500 bytes, respectively. Moreover, the CAP ratio is 0.8, which means at most 80% of the bandwidth within SI is allocated for the HCCA.

In modeling of the wireless transmission error, we utilize the signal-to-noise ratio (SNR) based on Orinoco 802.11b Card [64] and use an empirical curve of bit error rate (BER) versus SNR provided by Intersil WLAN chipset [65]. In the simulation, we assume that the distance between the AP and each multimedia station, denoted as $R$, is the same and is set to 145 m, 150 m, 155 m, 160 m, 165 m, and 170 m, which correspond to BER as shown in Table 3.2.
3.4 Application-Level QoS Assessment

Table 3.2: Relationship between distance and BER

<table>
<thead>
<tr>
<th>Distance, R [m]</th>
<th>BER</th>
</tr>
</thead>
<tbody>
<tr>
<td>EF</td>
<td>Error Free</td>
</tr>
<tr>
<td>145</td>
<td>$1.8 \times 10^{-5}$</td>
</tr>
<tr>
<td>150</td>
<td>$2.4 \times 10^{-5}$</td>
</tr>
<tr>
<td>155</td>
<td>$3.1 \times 10^{-5}$</td>
</tr>
<tr>
<td>160</td>
<td>$4.1 \times 10^{-5}$</td>
</tr>
<tr>
<td>165</td>
<td>$5.6 \times 10^{-5}$</td>
</tr>
<tr>
<td>170</td>
<td>$7.0 \times 10^{-5}$</td>
</tr>
</tbody>
</table>

3.3.2 Methods for subjective experiment

The subjective experiment was conducted as follows. We first made test samples for subjective assessment by actually outputting the audio and video MUs with the output timing obtained from the simulation. Here, we set the number of multimedia stations to 4. In the assessment, we use a PC with headphones and a 17 inch-LCD display. The number of assessors is 18, and their ages were 20s. The assessors were asked to classify the test samples into a certain number of categories each assigned an integer. Here, we use five categories of impairment of the rating-scale method: “imperceptible” assigned integer 5, “perceptible, but not annoying” 4, “slightly annoying” 3, “annoying” 2, and “very annoying” 1.

3.4 Application-Level QoS Assessment

In this section, we compare the TGes scheme and the proposed scheme in terms of application-level QoS in audio-video transmission. We first study the effect of SI on the application-level QoS. Next, we assess the effect of maximum number of retransmission on the QoS. Then, we evaluate how the number of admitted stations affects the QoS.

In the case of the TGes scheme, the calculation of arriving number of MSDUs can be inaccurate as mentioned in Section 3.1. Therefore, we also introduce a modified TGes scheme where the arriving number of MSDUs in an SI is calculated by Eq. (3.2), in the same way as the proposed scheme. In the modified TGes scheme, however, additional
TXOP duration for retransmission traffic is not allocated. In the following numerical results, TGe' means the modified TGe scheme.

### 3.4.1 The effect of SI

First, we study the effect of SI on the application-level QoS. In the simulation, we set various values of MSI in order to derive the SI values ranging from 25 ms to 500 ms. The number of multimedia stations is set to 4. In addition, the maximum retry number for retransmission attempt is set to 4. An MPDU will be discarded if this number is exceeded. In the figures to be shown, notation "TGe'[EF]'", for instance, refers to the result of the modified TGe scheme on the error free (EF) channel condition. Meanwhile, notation "Proposed[160]" refers to the results of the proposed scheme under a channel condition that the distance between the AP and the station is 160 m.

<table>
<thead>
<tr>
<th>SI (ms)</th>
<th>25</th>
<th>50</th>
<th>71.4</th>
<th>100</th>
<th>125</th>
<th>143</th>
<th>167</th>
<th>200</th>
<th>250</th>
<th>333</th>
<th>500</th>
</tr>
</thead>
<tbody>
<tr>
<td>Eq. (1.5)</td>
<td>2</td>
<td>4</td>
<td>5</td>
<td>7</td>
<td>9</td>
<td>10</td>
<td>12</td>
<td>14</td>
<td>17</td>
<td>23</td>
<td>34</td>
</tr>
<tr>
<td>Eq. (3.2)</td>
<td>2</td>
<td>4</td>
<td>6</td>
<td>8</td>
<td>10</td>
<td>12</td>
<td>14</td>
<td>16</td>
<td>20</td>
<td>27</td>
<td>40</td>
</tr>
<tr>
<td>Actual</td>
<td>2</td>
<td>4</td>
<td>6</td>
<td>8</td>
<td>10</td>
<td>12</td>
<td>14</td>
<td>16</td>
<td>20</td>
<td>27</td>
<td>40</td>
</tr>
</tbody>
</table>

Figure 3.3 shows the average MU delay for video as a function of the SI. We can observe in this figure that the average MU delay for the TGe' scheme and that of the proposed scheme become very small for all the values of SI if the channel is error-free. Meanwhile, the average MU delay for the TGe scheme becomes larger and fluctuates if SI increases beyond 50 ms. This is because the TGe scheme provides inaccurate numbers of arriving MSDUs from Eq. (1.5).

Table 3.3 shows the derived number of arriving MSDUs from Eq. (1.5), that from Eq. (3.2), and actual number of arriving MSDUs for an SI. In Table 3.3, we find that the proposed scheme derives the same number of arriving MSDUs as the actual number. However, the number in the TGe scheme becomes smaller than the actual one. This leads to an insufficient TXOP duration to transmit all MSDUs. As a result, the remainder of the MSDUs will accumulate in the source buffer; this increases the average MU delay.
Figure 3.3 also shows that the average MU delay for the TGe’ scheme increases with the increase of BER. On the other hand, in the proposed scheme, the average MU delay does not become larger when SI is larger than 100 ms except the case of 170 m. This is because in the proposed scheme additional TXOP duration is allocated. However, the average MU delay for $R = 170$ m becomes larger because many MPDUs are retransmitted. In addition, this figure reveals that the average MU delay in the proposed scheme increases if SI becomes smaller than 100 ms since the fraction of the surplus bandwidth to the SI is too small to allocate additional TXOP duration to all stations.

Figure 3.4 shows the coefficient of variation of output interval for video. This figure shows that for all schemes, the coefficient of variation of output interval increases with the increase of SI. As a video MU is generated in every 50 ms, it is ideal to send the MU immediately after the generation. However, when the SI is above 50 ms, several video MUs with different generation time are accumulated first before being sent. Therefore, the increase of SI will increase the number of accumulated MUs, which leads to increase of the coefficient of variation of output interval.

We have performed subjective experiment where we actually output the audio and video flows according to the output timing obtained from the simulation. From the subjective experiment, we have found that the subjective quality (QoE) for TGe' [EF] and that for Proposed[EF] are both good when the SI is below 100 ms over an error-free channel. However, the QoE for both schemes begins to deteriorate when the SI increases beyond 100 ms since the coefficient of variation of output interval becomes larger. We have also confirmed that the QoE for the TGe’ scheme becomes lower than that for the proposed scheme in a noisy environment.
3. QOE GUARANTEE IN ERROR-PRONE CHANNEL: PART I

Figure 3.3: Average MU delay for video versus SI

Figure 3.4: Coefficient of variation of output interval for video versus SI
3.4.2 The effect of the maximum number of retransmission

Next, we examine the effect of the maximum number of retransmissions on the application-level QoS. In the simulation, we consider only one multimedia station, and the SI is set to 100 ms, which is the minimum value where the application-level QoS does not deteriorate owing to insufficient bandwidth to allocate additional TXOP duration. The maximum retry number for retransmission attempt is set from 0 to 5. We compare the TGc' scheme and the proposed scheme.

In the figures to be shown, notation “TGc'-4” and “Proposed-4”, for instance, refer to the results of the modified TGc scheme and the proposed scheme, respectively, when the maximum number of retransmission is set to 4.

Figure 3.5 shows the MU loss ratio for video as a function of the distance between AP and each station. In this figure, the MU loss ratio for the both scheme is approximately the same because the BER is the same. The MU loss in the simulation is due to the MPDU retransmission that exceeds the maximum number of retransmission. We find in Fig. 3.5 that a larger maximum number of retransmission reduces the MU loss ratio. In addition, we have confirmed through simulation that the average MU delay for the proposed scheme is kept low even if the maximum number of retransmission increases, though the average MU delay for the TGc scheme deteriorates as the maximum number increases.

Meanwhile, Fig. 3.6 illustrates the MSDU loss ratio for video. This figure shows the same tendency to the MU loss ratio. However, we observe that the MU loss ratio takes higher values than the MSDU loss ratio. This is because a video MU is fragmented into several MSDUs. Therefore, the loss of one or more MSDUs of the fragmented MU leads to the loss of the corresponding whole MU.

In addition to the video with picture pattern I, we have also conducted a similar simulation with an IPPPPP picture pattern video. From the simulation results, we have observed that the MU loss ratio for the IPPPPP video becomes larger than those for the I video. This is because a video frame is decoded after all frames within a GOP are received correctly.
Figure 3.5: MU loss ratio for video versus $R$

Figure 3.6: MSDU loss ratio for video versus $R$
3.4 Application-Level QoS Assessment

3.4.3 The effect of the number of admitted stations

Finally, we investigate the effect of the number of admitted stations on the application-level QoS. In this simulation, we set the SI to 100 ms. Meanwhile, the maximum retry number for retransmission attempt is set to 4. We increase the number of multimedia stations from 1 to 5. It should be noted that on the simulation conditions the maximum number of admitted stations in the TGe' scheme is 5, which can obtain from Eq. (1.7).

Figures 3.7 and 3.8 indicate the average MU delay for audio and video, respectively, as a function of the number of multimedia stations. From these figures, we can observe that in the case of the TGe' scheme, the average MU delay for audio and that for video deteriorates as the distance increases. This is because additional TXOP duration is not allocated in the TGe' scheme.

From Fig. 3.7, we can find that the average audio MU delay for the proposed scheme becomes very small except for the case where the number of stations is five and the distance is 170 m. Meanwhile, we can see in Fig. 3.8 that the average video MU delay for the proposed scheme begins to increase if the number of stations becomes larger than four for the distances 150 m and 160 m, and three for 170 m.

We also make a similar observation in Fig. 3.9, which plots the mean square error of inter-stream synchronization; it deteriorates even if the number of stations is less than or equal to 5.

From the above observations, we can say that application-level QoS for the proposed scheme deteriorates even if the number of stations is less than the maximum number of admitted stations calculated from Eq. (1.7) in an error-prone channel. Therefore, a new admission control method is needed to provide QoS support in a lossy channel condition.

One simple method is that the HC monitors channel capacity for retransmission traffic for several SI’s and estimates additional capacity for retransmission traffic. In this case, the admission control can be done as follows:

$$\frac{TXOP_{k+1}^P + C_{k+1}^P}{SI} + \sum_{i=1}^{k} \frac{TXOP_i^P + C_i^P}{SI} \leq \frac{T - T_{CP}}{T}$$

(3.3)

where $C_i^P$ is the estimated additional TXOP duration of flow $i$ for the current channel condition. We have confirmed through simulation that the maximum number of admitted stations for the distances 150 m, 160 m, and 170 m are 4, 3, and 2, respectively.
Figure 3.7: Average MU delay for audio versus the number of multimedia stations

Figure 3.8: Average MU delay for video versus the number of multimedia stations
3.5 QoE Assessment

In this chapter, the results obtained from the rating-scale method are calculated into the interval scale with the law of categorical judgment.

To verify the obtained interval scale, we have performed Mosteller’s test. From the Mosteller’s test, after removing some values, we cannot reject the hypothesis that the obtained interval scale fits the observed data at a significance level of 0.01. Thus, we refer to the interval scale as the psychological scale.

Figures 3.10 through 3.13 illustrate the measured psychological scale as a function of the SI for Music video and Sport, respectively. In these figures, we selected the minimum value of the psychological scale as the origin of the ordinate, and each of four horizontal dotted lines indicates the boundary of a category.

From Figs. 3.10 through 3.13, we first find that the values of the psychological scale for the TGe scheme fluctuate if SI increases beyond 50 ms. This is because the TGe
scheme provides inaccurate numbers of arriving MSDUs from Eq. (1.5). This leads to an insufficient TXOP duration to transmit all MSDUs. As a result, the remainder of the MSDUs will accumulate in the source buffer; this increases the average MU delay.

These figures also shows that the QoE for the TGe’ scheme and that for the proposed scheme become “imperceptible” or “perceptible but not annoying” if SI is less than 125 ms when the channel is error-free. However, the QoE deteriorates gradually as the value of SI becomes larger. A larger value of SI leads to an increase of delay jitter, which interrupt the smoothness of audio-video playback.

We then observed that the psychological scale for the proposed scheme becomes higher than that for the TGe’ scheme for the distances 150 m and 160 m. This is because in the proposed scheme, additional TXOP duration is allocated when transmission error occurs. However, in the case of $R = 170$ m, the psychological scale becomes low even if the proposed scheme is selected since many MPDUs are retransmitted. In addition, these figures reveal that the psychological scale for the proposed scheme deteriorates if SI becomes smaller than 71.4 ms for $R = 160$ m and 50 ms for $R = 150$ m, respectively. This is because the fraction of the surplus bandwidth to the SI is too small to allocate enough additional TXOP duration.

We next examine the effect of content types on the QoE. We notice in Figs. 3.10 through 3.13 that the psychological scale for Sports of the proposed scheme does not deteriorate if $R$ is less than or equal to 160m when $SI > 71.4$ ms. On the other hand, in the case of Music video the values of the psychological scale for Proposed[150] and those for Proposed[160] become smaller than those for Proposed[EF]. In the case of Music video, the lyric of the song are shown during the playback. Therefore, Music video is affected more significantly by inter-media synchronization quality than Sport. We have confirmed through simulation that mean square errors of inter-stream synchronization between audio and video for Proposed[150] and Proposed[160] become larger than that for Proposed[EF]. However, in the case of $R = 170$ m, the QoE for Sport becomes “very annoying”, though that for Music video indicates “annoying”. This is because Sport is more sensitive to video quality than Music video. We have also confirmed through simulation that video MU loss ratio becomes more than 10 % when $R = 170$ m.
3.5 QoE Assessment

Figure 3.10: Psychological scale (Music video - TGe and TGe’ schemes)

Figure 3.11: Psychological scale (Music video - Proposed scheme)
3. QOE GUARANTEE IN ERROR-PRONE CHANNEL: PART I

Figure 3.12: Psychological scale (Sport - TGe and TGe' schemes)

Figure 3.13: Psychological scale (Sport - Proposed scheme)
3.6 QoS Mapping

Next, we perform QoS mapping between application-level QoS and QoE with multiple regression analysis. In the analysis, the application-level QoS parameters are considered as the predictor variables while the QoE parameter is considered as the criterion variable. 11 application-level QoS parameters are used in the QoS mapping as defined in Chapter 1.

As a result of the principal component analysis, we adopted the first four principal components since the cumulative contribution rate for the first four principal components becomes 91.551%. Therefore we can classify the application-level QoS parameters into four groups.

- **group A**: $R_a, L_a, L_v$
- **group B**: $E_a, E_v, C_a$
- **group C**: $R_v, D_v, C_v$
- **group D**: $D_a, E_{int}$

We then performed multiple regression analysis of all combinations of the predictor variables. The predictor variables of the adopted combination were statistically tested whether they make significant contributions to the multiple regression line. Finally, we found combination of the application-level QoS parameters which make the highest contribution rate adjusted for degree of freedom.

The obtained multiple regression lines for Music video and Sport are as follows:

\[
\hat{S}_{music}^{TGe} = 5.2 + 5.47C_a - 6.2 \times 10^{-4}D_v - 5.8 \times 10^{-2}D_a, \quad R^* = 0.926 \quad (3.4)
\]

\[
\hat{S}_{music}^{TGe'} = 4.46 + 1.1 \times 10^{-1}L_v - 1.8C_a - 3.2 \times 10^{-4}D_v - 7.6 \times 10^{-5}D_a, \quad R^* = 0.917 \quad (3.5)
\]

\[
\hat{S}_{music}^{Proposed} = 65.11 - 7.56R_a - 2.2C_a - 4.3 \times 10^{-4}D_v + 9.8 \times 10^{-9}E_{int}, \quad R^* = 0.893 \quad (3.6)
\]
3. QOE GUARANTEE IN ERROR-PRONE CHANNEL: PART I

\[
\hat{S}_{\text{sport}}^{\text{TGe}} = 5.6 + 6.36 \times 10^{-5} E_a - 7.4 \times 10^{-4} D_v - 2.1 \times 10^{-2} D_a , \quad R^* = 0.893 \quad (3.7)
\]

\[
\hat{S}_{\text{sport}}^{\text{TGe'}} = 4.4 + 1.9 \times 10^{-1} L_v - 1.6 C_a - 4.8 \times 10^{-4} D_v , \quad R^* = 0.944 \quad (3.8)
\]

\[
\hat{S}_{\text{sport}}^{\text{Proposed}} = 4.9 + 3.6 \times 10^{-2} L_v - 1.9 \times 10^{-4} E_a - 8.2 \times 10^{-4} D_v + 3.0 \times 10^{-8} E_{\text{int}} , \quad R^* = 0.926 \quad (3.9)
\]

In the above equation, \( \hat{S} \) represents the estimate of the psychological scale, its subscript means content type, while its superscript means the packet scheduling schemes. \( R^* \) denotes the contribution rate adjusted for degrees of freedom.

Figures 3.10 through 3.13 also show the estimated psychological scale derived from Eq. (3.4) through Eq. (3.9). From these figures, we find that the accuracy of the estimated psychological scale values is not high as the contribution rate adjusted for degrees of freedom for Eq. (3.4) through Eq. (3.9) are not very large. The accuracy of the estimated values of psychological scale decreases with the increase of SI and distance between the AP and the multimedia stations. Nevertheless, the estimated values still follow the tendency of the measured ones.

3.7 Conclusions

In this chapter, we have proposed a packet scheduling scheme for audio-video transmission by IEEE 802.11e HCCA and compared the application-level QoS and QoE of the TGe scheme, the TGe' scheme, and the proposed scheme. We studied the effect of SI, maximum number of retransmission, and number of multimedia stations on the application-level QoS. We have shown that in the TGe scheme the number of arriving MSDUs in an SI can be smaller than that of the actual ones. We have confirmed through simulation that the proposed scheme can improve the QoS of the TGe' scheme in an error-prone channel because the HC can utilize surplus bandwidth to allocate additional TXOP duration on the basis of the number of retransmission to replenish the insufficient bandwidth. We have also proposed an admission control scheme in a noisy environment. We have also presented that the QoE is affected by the content
types. In addition, we have performed QoS mapping between application-level and user-level with multiple regression analysis and obtained multiple regression lines to estimate QoE from the application-level QoS parameters.
Chapter 4

QoE Guarantee in Error-Prone Channel: Part II

4.1 Introduction

In the previous chapter, we have proposed a packet scheduling scheme for transmission of CBR audio-video traffic and have examined application-level QoS and QoE of the scheme in a noisy environment.

In this chapter, we continue our study by focusing on evaluation of video with IPPPPP-picture pattern which has VBR characteristic more remarkably if compared to the video with I-picture pattern used in the previous chapter.

This chapter proposes a cross-layer packet scheduling scheme for audio-video transmission with the IEEE 802.11e HCCA to solve the three problems of the TGc scheme described earlier under lossy channel conditions. In the scheme, the number of MSDUs generated by a station in an SI is calculated in the same way of the approach done in Chapter 3. However, in Chapter 3, channel allocation for VBR traffic is not considered. In the proposed scheme of this chapter, the HC first calculates basic TXOP for each station in an SI using the inter-arrival time of audio samples and that of video frames at the application layer to overcome the third problem. The HC then allocates additional TXOP in the SI to each station on the basis of the queue length of station source buffer. This TXOP allocation is useful to cope with the first and second problems. In addition, we also propose video frame skipping at the MAC-level of a source station. We then compare the TGc scheme and the proposed packet scheduling scheme with and without
4. QOE GUARANTEE IN ERROR-PRONE CHANNEL: PART II

the video frame skipping at the source in terms of application-level QoS and QoE for various values of BER and the number of stations. Furthermore, we also examine the effect of SI on the QoS and QoE in the proposed scheduling scheme since they can be highly affected by the polling interval. Application-level QoS assessment is performed through simulation, and QoE assessment is carried out by subjective experiment.

The rest of the chapter is organized as follows. Section 4.2 describes the proposed packet scheduling scheme. Section 4.3 specifies simulation conditions and methodology of subjective assessment. Sections 4.4 and 4.5 give numerical results of application-level QoS and QoE. Section 4.6 presents QoS mapping between the application-level QoS and QoE. Finally, Section 4.7 concludes this chapter.

4.2 Proposed Scheduling Scheme

In this section we will explain the proposed scheduling scheme. This scheme is a packet scheduling scheme with cross-layer mechanisms between MAC layer and application layer because it sends MPDUs at the MAC layer using traffic information at the application layer. In this section, we first describe how the HC allocates TXOP to each station in the proposed scheme. We then explain the video frame skipping performed by the scheduling scheme.

4.2.1 Channel allocation of the scheduling scheme

In the proposed scheduling scheme, the HC first calculates the basic TXOP duration for a flow in a similar way to that of the TGes scheme. However, the TGes scheme has the third problem described in Section 4.1; that is, the derived number of MSDUs calculated from Eq. (1.5) becomes smaller than the real number. Therefore, we propose a new method for calculating the number of MSDUs arriving in an SI in order to overcome this problem. The procedure of TXOP allocation in the proposed scheme is shown in Fig. 4.1.

In the proposed method, information about inter-MU time for a flow is required to calculate the number of MSDUs arriving from the flow in an SI.
4.2 Proposed Scheduling Scheme

Figure 4.1: Procedure of TXOP allocation in the proposed scheme
4. QOE GUARANTEE IN ERROR-PRONE CHANNEL: PART II

Using the *inter-MU* time together with other TSPEC parameters, the HC first computes the number of MSDUs arriving from flow $i$ within an *inter-MU* time as

$$n_i^P = \left\lfloor \frac{\text{inter}_MU_i \times \rho_i}{L_i} \right\rfloor$$

(4.1)

where *inter-MU*$_i$ denotes the inter-MU time for flow $i$. The superscript $P$ means the proposed scheme. Then, the number of MSDUs that arrives in an SI is computed as

$$N_i^P = \left\lfloor \frac{SI}{\text{inter}_MU_i} \times n_i^P \right\rfloor$$

(4.2)

Then, the basic TXOP duration for flow $i$, which is denoted by $\text{basicTXOP}_i$, is calculated with Eq. (1.6) by replacing $N_i^{TGe}$ in Eq. (1.5) with $N_i^P$ in Eq. (4.2).

Note here that in audio-video transmission with the scheduling scheme, each station has to pass the inter-MU time for audio and that for video from the application layer to the MAC layer and has to exchange the two parameters with the HC as TSPEC parameters. Therefore, extensions of the IEEE 802.11e HCCA MAC are needed to realize this scheme. However, the TXOP calculation for an audio or video flow using the inter-MU time can give an accurate number of MSDUs arriving from the flow.

The HC then calculates additional TXOP for each flow in the SI on the basis of the queue length of the source buffer at the end of the previous TXOP. The QoS control field of the IEEE 802.11e MAC header is utilized to deliver the queue length of the audio buffer and that of the video buffer at the stations. This queue length means the number of MPDUs which could not be transmitted during the previous TXOP because of the insufficient TXOP duration.

After the HC obtains the queue length record during the previous SI and then computes the basic TXOP duration, the HC computes the additional TXOP duration for flow $i$ as follows:

$$\text{addTXOP}_i^P = \frac{\text{queue}_i \times L_i}{R_i}$$

(4.3)

where $\text{queue}_i$ is the number of MPDUs in the (audio or video) queue of flow $i$. The additional TXOP is required for transmission of MPDUs left in the queue.

As the surplus bandwidth reduces with the increase of admitted flows, the HC might not be able to allocate enough additional TXOP duration. In this case, the HC
distributes the surplus bandwidth for each flow in a round-robin basis; that is, additional TXOP duration for transmission of one nominal-sized MSDU will be allocated to each flow until the surplus bandwidth is fully occupied. Note that the maximum additional TXOP duration for a station in an SI is bounded by $\text{addTXO}P^P_i$.

TXOP duration for flow $i$ in the SI is calculated as the sum of the basic TXOP duration and the additional one for flow $i$. When a station generates both audio and video flows, TXOP duration for the station is calculated as the sum of the TXOP duration for audio and that for video.

The proposed scheduling scheme is appropriate for transmission of VBR traffic and retransmission owing to transmission errors because the HC allocates additional TXOP to each station after it gives basic TXOP.

### 4.2.2 Video frame skipping

In this chapter, we also propose video frame skipping at the MAC-level of a source station. In the case of H.264 video transmission, a raw video stream is compressed into three kinds of frames: intra-coded frames (I-frames), predictive-coded frames (P-frames), and bidirectionally-predictive-coded frames (B-frames). The group of successive video frames starts with an I-frame, and all frames before the next I-frame are called group of picture (GOP). Note here that if video degradation occurs in a video frame of a GOP owing to transmission error, it will propagate into all the following video frames within the GOP; for example, in the case of video streams with GOP of IPPPPP sequences, the loss of the I-frame leads to loss of the entire frames of the GOP because the first P-frame requires the preceding I-frame in order to be decoded, the second P-frame requires the first P-frame, and so on.

In the proposed video frame skipping, frame information of a video flow is delivered from the application layer to the MAC layer at a source station. Therefore, the MAC layer knows the frame types of each MPDU. When the station fails to send an MPDU of a GOP, the station drops all the following MPDUs of the GOP according to the frame information received from the application layer; it then tries to send the first I-frame of the next GOP.

Figure 4.2 shows an example of the video frame skipping. In this figure, an I-frame and a P-frame are fragmented into 3 and 2 MPDUs, respectively. In Fig. 4.2 (i), a
station fails to transmit the first MPDU of I-frame 1 owing to channel transmission error. In this case, the HC drops all MPDUs of P-frame 2 to P-frame 6 from its source buffer; it then tries to send the first MPDU of I-frame 7 as shown Fig. 4.2 (ii).

The source video frame skipping can reduce the waste of bandwidth and can increase the channel capacity for transmission of audio-video MPDUs in a noisy environment, though this mechanism is also required an extension of the 802.11e HCCA MAC since the MAC layer needs video frame information at the application-level.

4.3 Experimental Methodology

In this chapter, the application-level QoS is assessed by simulation. Then, the QoE is assessed by subjective experiment. In this section, we first present the simulation conditions used for the application-level QoS assessment. We then elaborate on the methods of subjective experiment.

4.3.1 Simulation conditions

Figure 4.3 illustrates the system configuration used in the simulation. We focus on a single BSS which includes an AP and a various number of multimedia stations. The number of multimedia stations are denoted by $M$. All multimedia stations are located at the same distance (say $R$) from the AP. We assume the IEEE 802.11b physical layer based on DSSS with a channel data rate of 11 Mb/s. No wireless stations movement
is assumed during simulation. Each multimedia station sends stored audio and video streams to the AP as two separate transport streams using UDP/IP.

Table 4.1 summarizes media specifications of audio-video flows used in the simulation. We use an audio flow of ITU-T G.711 μ-law and an H.264 video stream. A video MU is defined as a video frame and is transferred as one or more UDP datagrams. An audio MU consists of 1000 audio samples, which corresponds to a single UDP datagram.

Two types of the contents are used in the simulation: Music video and Sport. The

<table>
<thead>
<tr>
<th>Table 4.1: Specifications of audio and video</th>
</tr>
</thead>
<tbody>
<tr>
<td>Coding scheme</td>
</tr>
<tr>
<td>G.711 μ-law</td>
</tr>
<tr>
<td>Image size [pixel]</td>
</tr>
<tr>
<td>Picture pattern</td>
</tr>
<tr>
<td>Average MU rate [MU/s]</td>
</tr>
<tr>
<td>Average inter-MU time [ms]</td>
</tr>
<tr>
<td>Average bit rate [kb/s]</td>
</tr>
<tr>
<td>Measurement time [s]</td>
</tr>
</tbody>
</table>
Music video shows scenes of a Japanese female singer dancing while singing with two men playing guitar. For the Sport, scenes of a football match with a commentator’s voice have been chosen. Here, we have encoded each video stream with a picture pattern of $IPPPPP$ at bit rate of 800 kb/s on average. The audio bit rate is constant at 64 kb/s.

In the simulation, we set the beacon interval to 1000 ms. The nominal MSDU size for audio and that for video are set to 1000 bytes and 1500 bytes, respectively. Each multimedia station has source buffer for audio MPDUs and that for video ones, separately. Audio transmission is given higher priority than video transmission when a station has both audio MPDUs and video ones. Video frame skipping at the receiver is always performed at the application-level; that is, the AP drops all the following video MUs of the GOP when it fails to receive a video MPDU of a GOP. It should be noted that in the following simulation results, video frame skipping at the source means that the video frame skipping is performed at the MAC-level of the source in addition to that of the application-level of the receiver. Moreover, the CAP ratio is 0.8, which means at most 80% of the bandwidth within SI is allocated for the HCCA. The maximum retry number for MPDU retransmission attempt is set to four. An MPDU will be discarded if this number is exceeded. The SI is set to 100 ms unless otherwise stated.

In modeling of the wireless transmission error, we utilize the SNR based on Orinoco 802.11b Card [64] and use an empirical curve of bit error rate (BER) versus SNR provided by Intersil WLAN chipset [65]. In the simulation, we assume that the distance between the AP and each multimedia station $R$ is set to 145 m, 150 m, 155 m, 160 m, 165 m, and 170 m, which correspond to BER as shown in Table 4.2.

### 4.3.2 Methods for subjective experiment

We first made test samples for subjective assessment by actually outputting the audio and video MUs with the output timing obtained from the simulation. We made a 10 s video clip from simulation results for 60 s. In the assessment, we use a PC with headphones and a 17 inch–LCD display. The subjective assessment was conducted by 17 students; their ages were 20s. The assessors were asked to classify the test samples into a certain number of categories each assigned an integer. Here, we use five
4.4 The Effect of Transmission Error

Table 4.2: Relationship between distance and BER

<table>
<thead>
<tr>
<th>Distance, R [m]</th>
<th>BER</th>
</tr>
</thead>
<tbody>
<tr>
<td>EF</td>
<td>Error Free</td>
</tr>
<tr>
<td>145</td>
<td>1.8 \times 10^{-5}</td>
</tr>
<tr>
<td>150</td>
<td>2.4 \times 10^{-5}</td>
</tr>
<tr>
<td>155</td>
<td>3.1 \times 10^{-5}</td>
</tr>
<tr>
<td>160</td>
<td>4.1 \times 10^{-5}</td>
</tr>
<tr>
<td>165</td>
<td>5.6 \times 10^{-5}</td>
</tr>
<tr>
<td>170</td>
<td>7.0 \times 10^{-5}</td>
</tr>
</tbody>
</table>


4.4 The Effect of Transmission Error

In this section, we compare the TGes scheme, the proposed scheme with and without the source video frame skipping in terms of application-level QoS and QoE. We discuss the effects of transmission error and the number of stations on the QoS. We also examine how the source video frame skipping can improve the QoS of the proposed scheme. We first study the effect of transmission error on the application-level QoS from simulation results. We then show the QoE results from subjective experiment.

In the case of the TGes scheme, the calculation of arriving number of MSDUs in an SI can be inaccurate as mentioned in Sec. 4.1. Therefore, we also introduce a modified TGes scheme where the arriving number of MSDUs in an SI is calculated by Eq. (4.2), in the same way as the proposed scheme. In the modified TGes scheme, however, additional TXOP duration for retransmission traffic is not allocated. In the following numerical results, TGes’ means the modified TGes scheme. The modified TGes scheme is also referred to as the TGes’ scheme in the chapter.

4.4.1 Application-level QoS assessment

First, we will show the results of the application-level QoS for Music video.
Figures 4.4 and 4.5 show the average MU delay for audio and that for video, respectively, as a function of the distance $R$ between the AP and the multimedia stations. It should be noted that BER becomes larger as the distance increases.

In these figures, notation $\text{TGe}'[3]$, for instance, refers to the result of the modified TGe scheme when the number of multimedia stations $M=3$. Meanwhile, notation Proposed[5] and Proposed-Skip[5] refer to the results of the proposed scheme with and without the source video frame skipping, respectively, when $M=5$.

These figures show three values of the number of multimedia stations for the proposed scheme: $M=3$, 4, and 5. On the other hand, the number of multimedia station for the TGe' scheme is fixed at 3. In the TGe' scheme, allocated TXOP duration for a station is almost the same when $M$ is less than or equal to 5. It should be noted that the maximum number of admitted stations is 5, which can obtain from Eq. (1.7). In the case of the proposed scheme, as the number of stations is larger, channel capacity for additional TXOP decreases even if $M$ is less than or equal to 5.

We can find in Figs. 4.4 and 4.5 that in the case of TGe' scheme, the average MU delay for audio and that for video increase drastically as the distance becomes longer. Under noisy channel environments, stations need to retransmit MPDUs when transmission error occurs. However, in the TGe' scheme, TXOP duration for retransmission traffic is not allocated. Therefore, queue length of the source buffer becomes longer as BER increases.

In contrast, these figures show that the average MU delay for the proposed scheme becomes much smaller than that for the TGe'e scheme when $M=3$. This is because the HC allocates additional TXOP duration according to the queue length of the source buffer. However, the average MU delay becomes larger as $M$ increases because the increase of the number of multimedia stations reduces the surplus bandwidth for allocation of additional TXOP duration. In particular, the average MU delay for video becomes very large if the distance is longer than 145 m for Proposed[5] and Proposed-Skip[5], and 160 m for Proposed[4]. Under these conditions, surplus bandwidth for additional TXOP duration becomes insufficient since many MPDUs are retransmitted.

We next examine the effect of the source video frame skipping on the average MU delay for video. Figure 4.5 reveals that the average MU delay of video for the Proposed-Skip scheme is smaller that that for the Proposed scheme if the distance is longer than 155 m for $M=4$, and 150m for $M=5$, respectively. Therefore, we can say that the
source video frame skipping can improve the average MU delay under lossy channel conditions. This is because the source video frame skipping can reduce traffic volume to the wireless channel when the traffic load is heavy owing to retransmission of video MPDUs.

Figures 4.6 and 4.7 depict the MU loss ratio for audio and that for video, respectively, as a function of the distance between the AP and the multimedia stations. We can observe in Fig. 4.6 that the MU loss ratio for audio becomes very small because frame error rate is small since the length of an audio MPDU is short under our simulation conditions.

Figure 4.7 reveals that the MU loss ratio for video increases as the distance becomes longer. In addition, Fig. 4.7 shows that the values of the MU loss ratio for video become almost the same for the TGe', Proposed, and Proposed-Skip schemes. This is because MPDU loss is caused in our simulation only if a station cannot succeed in sending an MPDU within the maximum retry number of retransmission attempt.

Figure 4.8 shows the MSDU loss ratio for video as a function of the distance $R$. We can observe from Figs. 4.7 and 4.8 that in the cases of the TGe’ scheme and the proposed scheme without the source video frame skipping, the MU loss ratio for video becomes much larger than the MSDU loss ratio when $R > 155$ m. A video frame is fragmented into several MSDUs and the loss of an MSDU of a video frame leads to the loss of the corresponding whole video frames. Furthermore, the loss of a video frame leads to the loss of all video frames of a GOP as shown in Section 4.2. Therefore, the values of the MU loss ratio become large even if those of the MSDU loss ratio are small.

We can also find in Fig. 4.8 that the MSDU loss ratio for the proposed scheme with the source video frame skipping is much larger than that for the TGe’ scheme and the Proposed scheme without the video frame skipping if the distance is longer than 155 m. In the case of the Proposed-Skip scheme, many MSDUs are dropped at the source station when the distance is longer than 155 m. This leads to a larger value of MSDU loss ratio. It should be noted that all video frames of the GOP are dropped at the HC if an MSDU of a video frame of a GOP is failed to transmit from a station to the HC.

Figures 4.9 illustrate the coefficient of variation of output interval for video versus the distance between the AP and the multimedia stations. For all the schemes shown in this figure, the coefficient of variation of output interval for video increases when the distance becomes longer than 155 m except the case of Proposed-Skip[5]. This is
because MPDU retransmission occurs more frequently as BER increases. In the case of Proposed-Skip[5], it should be noted that the average MU delay for video deteriorates drastically if the distance increases beyond 145 m (see Fig. 4.5), though the coefficient of variation of output interval is small compared to the other cases.

In Fig. 4.10, mean square error of inter-stream synchronization is shown as a function of the distance. From Figs. 4.5 and 4.10, we can say that the inter-media synchronization quality for the TGe' and proposed schemes highly depends on the average MU delay for video.

Then, we will show application-level QoS for Sport. Figs. 4.11 and 4.12 plot the average MU delay for video, and the mean square error of inter-stream synchronization, respectively, as a function of the distance between the AP and the multimedia stations. From Figs. 4.5 and 4.10 through 4.12, we can find that the values of these application-level QoS parameters for Music video become almost the same as those for Sport.

![Figure 4.4: Average MU delay for audio versus R (Music video)](image-url)
4.4 The Effect of Transmission Error

Figure 4.5: Average MU delay for video versus R (Music video)

Figure 4.6: MU loss ratio for audio versus R (Music video)
4. QOE GUARANTEE IN ERROR-PRONE CHANNEL: PART II

Figure 4.7: MU loss ratio for video versus $R$ (Music video)

Figure 4.8: MSDU loss ratio for video versus $R$ (Music video)
4.4 The Effect of Transmission Error

Figure 4.9: Coefficient of variation of output interval for video versus $R$ (Music video)

Figure 4.10: Mean square error of inter-stream synchronization versus $R$ (Music video)
Figure 4.11: average MU delay for video versus $R$ (Sport)

Figure 4.12: Mean square error of inter-stream synchronization versus $R$ (Sport)
4.4 The Effect of Transmission Error

4.4.2 QoE assessment

We then examine the QoE of the TGe' and the proposed schemes from subjective experimental results.

In this chapter, the results obtained from the rating-scale method are calculated into the interval scale with the law of categorical judgment.

To verify the obtained interval scale, we have performed Mosteller's test [19]. From the Mosteller's test, we cannot reject the hypothesis that the obtained interval scale fits the observed data at a significance level of 0.01. Thus, we refer to the interval scale as the psychological scale.

Figures 4.13 and 4.14 show the measured psychological scale versus the distance between the AP and the multimedia stations for Music video and Sport, respectively. In these figures, we selected the minimum value of the psychological scale as the origin of the ordinate, and each of four horizontal dotted lines indicates the boundary of a category. We applied the law of categorical judgment to the measurement results by the rating-scale method for Music video and those for Sport separately. Therefore, the boundary of each category in Fig. 4.13 and that in Fig. 4.14 have become different values. These figures show the values of the psychological scale for the TGe' scheme and the proposed scheme with and without the source video frame skipping.

From Figs. 4.13 and 4.14, we can observe that the QoE for the three schemes deteriorates as the distance increases. This implies that the QoE deteriorates as the BER becomes larger. These figures also show that the values of the psychological scale for the TGe' scheme become the lowest of the three schemes. This is because the TGe' scheme allocates constant TXOP duration. Therefore, the TGe' scheme is not flexible for packet size fluctuation of the VBR traffic and cannot allocate surplus bandwidth for MPDU retransmission.

We then discuss the effect of content types on the QoE. We find in these figures that the psychological scale for Sport tends to deteriorate more drastically than that for Music video as the distance becomes longer, though we have found that the values of the application-level QoS parameters become almost the same; for example, Proposed[5] for Sport indicates "very annoying" if the distance is longer than 155 m, while Proposed[5] for Music indicates around the boundary between "annoying" and "very annoying" even if the distance is 170 m. This implies that the QoE depends on content types.
4. QOE GUARANTEE IN ERROR-PRONE CHANNEL: PART II

Furthermore, Fig. 4.14 also shows that the QoE for Proposed-Skip[5] becomes higher than that for Propose[5] when the distance is longer than 150 m. This means that the QoE for Sport can be improved to some extent by the source video frame skipping. We can also find in Fig. 4.14 that the QoE for Proposed-Skip[4] becomes higher than that for Proposed[4] when the distance is longer than 160 m.

We then examine which application-level QoS parameters affect the QoE dominantly. For Music video, we can see in Fig. 4.13 that the QoE for the TGf scheme is lower than that for the Proposed and Proposed-Skip schemes. In addition, the difference in the psychological scale value between the Proposed-Skip scheme and the Proposed scheme is small. We can also make similar observations in Fig. 4.4. Therefore, we can say that the average MU delay for audio highly affects the QoE. This makes us confirm that Music video is audio-dominant. In the case of Sport, Fig. 4.14 shows that the QoE for the proposed scheme can be improved by the source video frame skipping when R is long to some extent. We can also make similar observations in Fig. 4.11; therefore, the QoE for Sport is affected by the average MU delay for video. This result implies that Sport is video-dominant.

From the above observations, we can say that the source video frame skipping is effective in improving QoE for Sport when the channel capacity for retransmission is insufficient owing to a long value of the distance.

4.5 The Effect of SI

In this section, we examine the effect of SI on the application-level QoS and QoE. We assess here the QoS for the proposed scheme with the source video frame skipping because we found in the previous section that the QoS for the proposed scheme with the source video frame skipping becomes higher than or nearly the same as that for the proposed scheme without the video frame skipping and the TGf scheme. In the simulation, we set eleven values of MSI, which correspond to SI as shown in Table 4.3. The number of multimedia stations is set to 4.

In the figures to be shown, notation Proposed-Skip(160), for example, in this section refers to the results when the distance between the AP and the multimedia stations is 160 m.
4.5 The Effect of SI

Figure 4.13: Psychological scale (Music video)

Figure 4.14: Psychological scale (Sport)
4. QOE GUARANTEE IN ERROR-PRONE CHANNEL: PART II

Table 4.3: Relationship between MSI and SI (ms)

<table>
<thead>
<tr>
<th>MSI</th>
<th>25</th>
<th>50</th>
<th>75</th>
<th>100</th>
<th>125</th>
<th>150</th>
<th>175</th>
<th>200</th>
<th>250</th>
<th>350</th>
<th>500</th>
</tr>
</thead>
<tbody>
<tr>
<td>SI</td>
<td>25</td>
<td>50</td>
<td>71.4</td>
<td>100</td>
<td>125</td>
<td>143</td>
<td>167</td>
<td>200</td>
<td>250</td>
<td>333</td>
<td>500</td>
</tr>
</tbody>
</table>

4.5.1 Application-level QoS assessment

In the following numerical results, we will show application-level QoS for Music video. We have confirmed through simulation that the application-level QoS parameters for Music video and those for Sport are almost the same.

Figures 4.15 and 4.16 show the average MU delay of audio and that of video as a function of the SI. Figure 4.15 reveals that the average MU delay for audio tends to become larger as the SI increases because the polling interval for each station becomes longer. It should be noted that the average MU delay decreases if SI changes from 125 ms to 143 ms under lossy conditions because $N^P$ calculated from Eq. (4.2) changes from 1 to 2.

We then find in Fig. 4.16 that the average MU delay for video also becomes larger as the distance increases when SI > 100ms. However, this figure also shows that the average MU delay for video deteriorates if the SI decreases below 71.4 ms for Proposed-Skip(150), Proposed-Skip(160), and Proposed-Skip(170), and 50 ms for Proposed-Skip(EF). This is because polling overhead increases if the SI is too small.

Figures 4.17 and 4.18 plot the MU loss ratio for audio and that for video as a function of the SI. We can see in these figures that the MU loss ratio for audio and that for video are almost constant regardless the SI.

Figures 4.19 illustrates the coefficient of variation of output interval for video versus the SI. This figure shows that the coefficient of variation of output interval increases with the increase of SI. This is because a longer polling interval leads to a larger value of MU delay.

Finally, Fig. 4.20 shows mean square error of inter-stream synchronization as a function of the SI. We can see in this figure that Fig. 4.20 exhibits similar characteristics to Fig. 4.16.
4.5 The Effect of SI

Figure 4.15: Average MU delay for audio versus SI (Music video)

Figure 4.16: Average MU delay for video versus SI (Music video)
4. QOE GUARANTEE IN ERROR-PRONE CHANNEL: PART II

Figure 4.17: MU loss ratio for audio versus SI (Music video)

Figure 4.18: MU loss ratio for video versus SI (Music video)
4.5 The Effect of SI

Figure 4.19: Coefficient of variation of output interval for video versus SI (Music video)

Figure 4.20: Mean square error of inter-stream synchronization versus SI (Music video)
4. QOE GUARANTEE IN ERROR-PRONE CHANNEL: PART II

4.5.2 QoE assessment

Figures 4.21 and 4.22 show the measured psychological scale for Music video and Sport, respectively, as a function of the SI. These figures show that the QoE except Proposed-Skip(170) for the two contents deteriorates gradually if the SI increases beyond 100 ms because of longer polling interval. We can also observe in Figs. 4.15, 4.16, 4.19, and 4.20 that the application-level QoS parameters deteriorate as SI increases beyond 100 ms. In the case of Proposed-Skip(170), we find in Figs. 4.21 and 4.22 that the QoE is low for all the values of SI. We see in Fig. 4.18 that the MU loss ratio for video is large when R = 170 m.

Figures 4.21 and 4.22 also show that the QoE becomes low when the SI is below 50 ms except Proposed-Skip(EF) since the fraction of the surplus bandwidth to the SI becomes too small to allocate enough TXOP duration to stations in a noisy environment. From Fig. 4.16 and 4.20, we find that the values of the average MU delay for video and those of mean square error of inter-stream synchronization are very large when SI = 25 ms.

From Figs. 4.21 and 4.22, we can say that the appropriate value of the SI for the two contents is 50 ms, which is equal to the inter-MU time for video traffic. Therefore, the inter-MU time for video should be selected as the SI to achieve high QoE under our simulation conditions.

4.6 QoS Mapping

Next, we perform QoS mapping between application-level QoS and QoE with multiple regression analysis. In the analysis, the application-level QoS parameters are considered as the predictor variables while the QoE parameter is considered as the criterion variable. 11 application-level QoS parameters are used in the QoS mapping as defined in Chapter 1.

First, we deal with QoS mapping in studying the effect of transmission error. As a result of the principal component analysis, we adopted the first three principal components since the cumulative contribution rate for the first three principal components becomes 92.1%. Therefore we can classify the application-level QoS parameters into three groups.
4.6 QoS Mapping

Figure 4.21: Psychological scale (Music video)

Figure 4.22: Psychological scale (Sport)
group A $R_a, R_v, L_a, L_v, E_a, C_v$

group B $D_v, E_v, C_a$

group C $D_a, E_{int}$

We then performed multiple regression analysis of all combinations of the predictor variables. The predictor variables of the adopted combination were statistically tested whether they make significant contributions to the multiple regression line. Finally, we found combination of the application-level QoS parameters which make the highest contribution rate adjusted for degree of freedom.

The obtained multiple regression lines for Music video and Sport are as follows:

\[
\hat{S}_{music}^{TGe'} = 3.49 + 2.82 \times 10^{-4}E_a - 5.3 \times 10^{-4}D_v , \quad R^* = 0.982 \tag{4.4}
\]

\[
\hat{S}_{music}^{Proposed} = 4.65 - 2.4 \times 10^{-4}E_a - 2.8 \times 10^{-4}D_v , \quad R^* = 0.947 \tag{4.5}
\]

\[
\hat{S}_{music}^{Proposed-Skip} = 4.24 - 1.9 \times 10^{-4}E_a - 1.1 \times 10^{-4}D_v , \quad R^* = 0.882 \tag{4.6}
\]

\[
\hat{S}_{sport}^{TGe'} = 3.94 + 3.85L_a - 1.0 \times 10^{-3}D_v + 9.3 \times 10^{-8}E_{int} , \quad R^* = 0.930 \tag{4.7}
\]

\[
\hat{S}_{sport}^{Proposed} = 4.62 - 2.0 \times 10^{-4}E_a - 6.1 \times 10^{-4}D_v , \quad R^* = 0.952 \tag{4.8}
\]

\[
\hat{S}_{sport}^{Proposed-Skip} = 4.62 - 2.3 \times 10^{-4}E_a - 2.6 \times 10^{-4}D_v , \quad R^* = 0.934 \tag{4.9}
\]

In the above equation, $\hat{S}$ represents the estimate of the psychological scale, its subscript means content type, while its superscript means the packet scheduling schemes. $R^*$ denotes the contribution rate adjusted for degrees of freedom. Using Eq. (4.4) through Eq. (4.9), we then plot the estimated psychological scale as shown in Figs. 4.13 and 4.14. From these figures, we find that the accuracy of the estimated values of the psychological scale is high since the contributed rate adjusted for degrees of freedom for Eq. (4.4) through Eq. (4.9) is near to 1.0. However, the accuracy of
the estimated values decreases with the increase of the distance between the AP and
the multimedia stations especially in the case of Music video under the Proposed-Skip
scheme. We notice that the contribution rate adjusted for degrees of freedom for Eq.
(4.6) is lower than the other cases.

Next, we perform QoS mapping in studying the effect of SI using the similar way
described earlier. From the principal component analysis, we adopted the first three
principal components since the cumulative contribution rate for the first three principal
components becomes 97.2%. Therefore we can classify the application-level QoS
parameters into three groups.

**group A**  \( R_a, R_v, L_a, L_v, E_a, E_v \)

**group B**  \( D_a, C_a, C_v \)

**group C**  \( D_v, E_{int} \)

The obtained multiple regression lines for Music video and Sport are as follows:

\[
\hat{S}^{EF}_{music} = 4.79 - 2.3 \times 10^{-2}D_a - 2.7 \times 10^{-5}E_{int} \quad R^* = 0.991 \quad (4.10)
\]

\[
\hat{S}^{150}_{music} = 5.52 - 0.89L_v - 1.6 \times 10^{-2}D_a - 7.1 \times 10^{-8}E_{int} \quad R^* = 0.980 \quad (4.11)
\]

\[
\hat{S}^{160}_{music} = 7.04 - 0.33L_v - 1.2 \times 10^{-2}D_a - 4.2 \times 10^{-8}E_{int} \quad R^* = 0.975 \quad (4.12)
\]

\[
\hat{S}^{170}_{music} = 91.1 - 4.3 \times 10^{-8}E_{int} - 11.3R_a - 6.4 \times 10^{-3}D_a \quad R^* = 0.887 \quad (4.13)
\]

\[
\hat{S}^{EF}_{sport} = 5.17 - 1.9 \times 10^{-2}D_a \quad R^* = 0.957 \quad (4.14)
\]

\[
\hat{S}^{150}_{sport} = 4.96 - 0.179L_v - 1.0 \times 10^{-2}D_a - 1.0 \times 10^{-7}E_{int} \quad R^* = 0.978 \quad (4.15)
\]
4. QOE GUARANTEE IN ERROR-PRONE CHANNEL: PART II

\[ \hat{S}_{s_{160}} = 2.65 + 0.22L_v - 1.9C_a - 1.0 \times 10^{-3}D_v, \quad R^* = 0.993 \]  (4.16)  

\[ \hat{S}_{s_{170}} = 11.41 - 1.5E_a - 6.7 \times 10^{-2}E_v - 9.4 \times 10^{-2}L_v, \quad R^* = 0.932 \]  (4.17)

Here, superscript denotes the distance between the AP and stations. Using Eq. (4.10) through Eq. (4.17), we then plot the estimated psychological scale as shown in Figs. 4.21 and 4.22. These figures show that the accuracy of the estimated psychological scale values is high except for Music video when \( R = 170 \) m, which has lower contribution rate adjusted for degrees of freedom than the other cases.

4.7 Conclusions

In this chapter, we have proposed a new cross-layer packet scheduling scheme for audio-video transmission with the IEEE 802.11e HCCA to solve the problems of the TGe scheme. We have also introduced video frame skipping at source stations to reduce the traffic volume sent to the wireless channel in a noisy environment.

We have compared the TGe’s scheme, the proposed packet scheduling scheme with and without the source video frame skipping in terms of application-level QoS and QoE. Numerical results have shown that the proposed packet scheduling scheme can achieve higher quality than the TGe’s scheme under lossy channel conditions. We also showed that the proposed scheduling scheme can improve the QoS by utilizing the source video frame skipping. Furthermore, we also examined the effect of the SI on the QoS and QoE in the proposed packet scheduling scheme. As a result, we have shown that the appropriate value of SI to realize high QoS is equal to the inter-MU time for video traffic. We have also performed QoS mapping between application-level and user-level with multiple regression analysis and obtained multiple regression lines to estimate QoE from the application-level QoS parameters.

In the simulation of this chapter, the inter-MU time for video and that for audio are set to 50 ms and 125 ms, respectively: that is, the former is shorter than the latter. Our future study includes QoS and QoE assessment using other values of inter-MU time for audio and that for video. We should also investigate the QoS and QoE of the proposed scheme with error concealment.
Chapter 5

Conclusions

In this thesis, we have proposed several packet scheduling schemes designed to provide QoE and QoS guarantee for audio-video transmission over IEEE 802.11e HCCA WLANs. We also performed QoE and QoS assessment by subjective experiments and simulations under various wireless channel conditions.

In Chapter 2, we proposed the SS and MPDS schemes and compared them with the TGe scheme in terms of application-level QoS and QoE. Here, we assume error-free transmission. Three types of content were used in the study: Music video, Sport, and Movie. They have been encoded in two types of picture pattern: I and IPPPPP, each at three different bit rates. We studied the effect of the picture patterns, bit rates, and content types on QoS. We also examined the effect of TXOP duration on the QoS. From the results, we can give conclusions below.

- The MPDS scheme is the appropriate packet scheduling scheme when dealing with VBR traffic like video with the IPPPPP picture pattern because it can absorb the burstiness of this traffic.

- The number of admitted stations can be increased by reducing the TXOP duration especially in the case of CBR traffic. The QoE for CBR traffic like video with I picture pattern is kept high even when the TXOP duration for video flow is reduced.

- The MPDS scheme outperforms the SS and TGe schemes when the number of stations is small. Even if the performance of the MPDS scheme deteriorates with
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the increase of the number of stations, the results are comparable to or better than that of the SS and TGe schemes.

In Chapter 3, we proposed a packet scheduling scheme for transmission of CBR traffic over error-prone channel. We studied the effect of SI, maximum number of retransmission, and number of multimedia stations on the application-level QoS and QoE. We have shown that in the TGe scheme the number of arriving MSDUs in an SI can be smaller than that of the actual ones. We have also discussed an admission control scheme in a noisy environment. From this chapter, we can give conclusions below.

- Extra bandwidth is required for audio-video transmission in a noisy environment. This is because the transmission errors cause frame retransmissions of the corrupted frames.

- The proposed scheme can improve the QoS of the TGe scheme in an error-prone channel because the HC can utilize surplus bandwidth to allocate additional TXOP duration on the basis of the number of retransmission to replenish the insufficient bandwidth.

- QoE deteriorates with the increase of SI regardless the packet scheduling schemes. However, SI with too small value also degrades the QoE since the fraction of the surplus bandwidth to the SI is too small to allocate additional TXOP duration to all stations.

In Chapter 4, we proposed a cross-layer packet scheduling scheme for VBR traffic transmission. We have also introduced video frame skipping at source stations to reduce the traffic volume sent to the wireless channel in a noisy environment. We have compared the TGe scheme, the proposed packet scheduling scheme with and without the source video frame skipping in terms of application-level QoS and QoE. Furthermore, we also examined the effect of the SI on the QoS and QoE in the proposed packet scheduling scheme. From the results, we can conclude as below.

- VBR traffic transmission in an error-prone channel requires additional bandwidth to absorb packet size fluctuation and retransmission of corrupted frames.
The proposed cross-layer packet scheduling scheme can achieve higher quality than the TG scheme under lossy channel conditions especially when the video frame skipping is carried out.

The appropriate value of SI to realize high QoS is equal to the inter-MU time for video traffic.

In this thesis, we have also performed QoS mapping between application-level and user-level with multiple regression analysis and obtained multiple regression lines. From QoS mapping, we can estimate QoE from the application-level QoS parameters.

From the conclusions given above, it is clear that packet scheduling scheme play a prominent role in achieving high QoE and QoS. Moreover, optimization of parameters like SI and TXOP duration is indispensable. Although we have performed the QoS assessment assuming the physical layer of IEEE 802.11b, all of the proposed schemes are also working seamlessly with all existing IEEE 802.11 WLAN standards.

This thesis can help network engineers in design and implementation factors on provisioning QoE and QoS guarantee for audio-video transmission over IEEE 802.11e HCCA WLANs.

Future work includes

- devise an adaptive scheduling scheme where the HC determines the optimum TXOP duration, service interval, and maximum number of retransmissions by measuring application-level QoS parameters which highly affect QoE

- utilize error concealment and error resilience technique of H.264 in the study

- consider cross-layer optimization between the upper layer and lower layer of the OSI protocol stack.
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