PAPER

Application-Level QoS and QoE Assessment of Audio-Video Transmission with TXOP-Bursting by IEEE 802.11e EDCA*

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SUMMARY This paper assesses application-level QoS and Quality of Experience (QoE) in the case where audio and video streams are transferred with the enhanced distributed channel access (EDCA) of the IEEE 802.11e MAC. In EDCA, a station can transmit multiple MAC frames during a transmission opportunity (TXOP); this is referred to as TXOP-bursting. By simulation, we first compare application-level QoS with the TXOP-bursting scheme and that without the scheme for various distances between access point (AP) and stations. In this paper, we suppose that the bit error rate (BER) becomes larger as the distance increases. Numerical results show that TXOP-bursting can improve many metrics of video quality such as average media unit (MU) delay, MU loss ratio, and media synchronization quality, particularly when the AP sends audio and video streams to stations in the downlink direction. We then examine the effect of $TXOP_{Limit}$ on the video quality. Simulation results show that the video quality can be degraded if the value of $TXOP_{Limit}$ is too small. Furthermore, we assess QoE by the method of successive categories, which is a psychometric method. Numerical results show that TXOP-bursting can also improve the QoE. We also perform QoS mapping between application-level and user-level with principal component analysis and multiple regression analysis.
key words: *wireless LAN, IEEE802.11e, EDCA, audio-video transmisson, QoE*

1. Introduction

In recent years, wireless local area networks (LANs) have been widely used to realize indoor high-speed wireless access. Since the popularity of multimedia applications like voice over IP and streaming video are growing rapidly, a demand for support of *Quality of Service (QoS)* in wireless LANs is also increasing.

The IEEE 802.11 Task Group E has worked to enhance the legacy 802.11 MAC to expand support for applications with QoS requirements [1]. In the IEEE 802.11e MAC, *hybrid coordination function (HCF)* is newly supported to extend *distributed coordination function (DCF)* and *point coordination function (PCF)* in the legacy IEEE 802.11 MAC [2]. The HCF has two access methods: *enhanced distributed channel access (EDCA)* and *HCF controlled channel access (HCCA)* [1]. The former is an enhanced version of the DCF and can support relative priority services for multimedia transmission. The latter is an improved polling scheme based on the PCF. In this paper, we study the audio-video transmission with EDCA and focus on a *basic service set (BSS)* of an infrastructure wireless LAN, which includes an *access point (AP)* and stations associated with the AP.

The performance of EDCA has already been studied by many researchers [3]–[15]. References [3] through [7] evaluate the performance of EDCA by simulation considering diverse types of traffic such as voice, video, and data. In each of [5] and [6], the effectiveness of the *transmission opportunity (TXOP)*-bursting is also evaluated. The TXOP-bursting allows a station to send multiple MAC frames during a TXOP if it succeeds in sending the first frame. This scheme is also called the *contention free bursting (CFB)*. Reference [7] investigates the performance of the CFB with a constant value of *frame error rate (FER)*. In [8], the performance of EDCA with the TXOP-bursting is experimentally investigated in multimedia streaming applications. References [9] through [11] examine the *block acknowledgment (Block ACK)* mechanism, which improves channel efficiency by aggregating several acknowledgments into one frame. Reference [12] proposes an adaptive setting scheme of values of *contention window (CW)* to achieve better MAC-layer performance for integrated voice and data transmission. Admission control algorithms to support QoS requirement in IEEE 802.11e are studied in [13]–[15].

The papers mentioned above focus mainly on MAC-level QoS. However, we should consider QoS at each level of the protocol stack in the wireless LANs. Reference [16] identifies six levels of QoS in IP networks: *physical-level, node-level, network-level, end-to-end-level, application-level*, and *user-level*. In multimedia applications, user-level QoS is the most important since the final goal of multimedia services is to provide high user-level (perceptual) QoS for the end-users; this is also referred to as *Quality of Experience (QoE)* in ITU-T [17]. In addition, application-level QoS should also be evaluated since it is closely related to QoE [16]. In particular, in continuous media transmission, media synchronization quality is very important as application-level QoS [16] since temporal relations between *Media Units (MUs)*** should be preserved. In [18], the authors have studied application-level QoS and QoE of audio-video transmission in the downlink (AP-to-station) direction with EDCA under the assumption of an error-free channel and single MAC frame transmission per

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**An MU is the unit of information that is delivered from a source station to a destination station at the application-level.

TXOP.

This paper assesses application-level QoS and QoE in the case where audio and video streams are transferred with the TXOP-bursting of EDCA. We first examine how the TXOP-bursting can improve application-level QoS in the presence of transmission errors through simulation. In the simulation, we assume that audio and video streams are transferred between the AP and stations with the IEEE 802.11e EDCA over the IEEE 802.11b physical layer at a channel rate of 11 Mbps [19]. In wireless LANs, a station can fail to send MAC frames owing to transmission errors. In this case, queue length of the source buffer becomes long since MAC frames for retransmission as well as newly generated ones are kept in the source buffer. Therefore, the TXOP-bursting may be attractive because more than one MAC frame can be sent successively.

This paper then assesses QoE by a subjective experiment. Since QoE is directly related to human perception, we utilize a psychometric method referred to as the *method of successive categories* [20]. We also carry out QoS mapping between application-level and user-level with *principal component analysis* and *multiple regression analysis*. As a result, we obtain multiple regression lines to estimate the QoE from the application-level QoS. The novelties of this paper are as follows.

- Application-level QoS in a noisy environment with the TXOP-bursting scheme and that without the scheme are compared by simulation for various values of the distance between the AP and stations. In our simulation an increase of the distance means a larger value of *bit error rate (BER)*.
- The effect of the direction of audio-video transmission with the TXOP-bursting on application-level QoS is examined. In our simulation application-level QoS is evaluated in the case where audio-video streams are transferred in the uplink (stations-to-AP) direction as well as the downlink (AP-to-stations) direction.
- Effects of values of $TXOP_{Limit}$ on the application-level QoS are studied. The $TXOP_{Limit}$ indicates the maximum duration of a TXOP and is an important MAC parameter for efficient MAC frame transmission utilizing the TXOP-bursting.
- QoE in a noisy environment with the TXOP-bursting is assessed on the basis of subjective experimental results. QoS mapping between application-level and user-level is also performed with principal component analysis and multiple regression analysis.

The rest of the paper is organized as follows. Section 2 describes the IEEE 802.11e EDCA briefly. Section 3 specifies a simulation model, and Sect. 4 gives numerical results from simulation and discusses the application-level QoS. Section 5 investigates QoE by subjective experiment. Section 6 gives a conclusion of this paper.

2. IEEE 802.11e EDCA

In this section we briefly describe the transmission procedure for a MAC frame with the IEEE 802.11e EDCA. In this paper, a MAC frame means an *MAC protocol data unit (MPDU)*.

The IEEE 802.11e EDCA introduces *access categories (ACs)* to support differentiated channel access for applications with QoS requirements. In EDCA, eight priority levels according to the 802.1D bridge specification are mapped into four ACs [1]. A station which supports EDCA has an individual source buffer for each of the four ACs, and the channel access function based on the *carrier sense multiple access with collision avoidance (CSMA/CA)* is independently carried out per AC.

In EDCA, a station uses *arbitration interframe space (AIFS)* for the contention process. When a wireless station generates a data frame in its source buffer of access category AC, it senses the state of the channel to judge whether another station is transmitting any frame or not. If the medium is judged to be idle for at least $AIFS[AC]$, the station can transmit the MAC frame. If the destination station receives the MAC frame correctly, it sends an *acknowledgment frame (ACK)* back to the source station after a *(short interframe space (SIFS)* period. When the medium is judged to be busy, the station waits for an $AIFS[AC]$ after the channel becomes idle, and then it selects a random backoff period. The station decreases its backoff timer while the channel is idle. When the backoff timer becomes 0, the station transmits the MAC frame.

The backoff period for access category AC is a multiple of the slot-time duration and is selected uniformly in the range of 0 through $CW[AC]$ slot-times. The initial value of $CW[AC]$ is $CW_{min}[AC]$; then, for the n-th retransmission, CW is set to $2^n(CW_{min}[AC] + 1) - 1$. When CW becomes a predetermined maximum value $CW_{max}[AC]$, it remains at $CW_{max}[AC]$. In EDCA, four ACs can be used, and the values of $CW_{min}[AC]$ and $CW_{max}[AC]$ are selected per AC.

EDCA can support service differentiation by setting different values of $AIFS[AC]$, $CW_{min}[AC]$, and $CW_{max}[AC]$ according to ACs; that is, as these values are smaller, the priority of the AC is higher.

In addition, the IEEE 802.11e MAC defines a TXOP as an interval of time when a particular station has right to initiate frame exchange sequences onto the wireless medium [1]. In EDCA, a station can transmit multiple MAC frames during a TXOP from the source buffer of an AC, until the time reaches the $TXOP_{Limit}[AC]$ [2]. We refer to this scheme as the TXOP-bursting in this paper. Figure 1 shows transmission of two Mac frames during a TXOP. During a

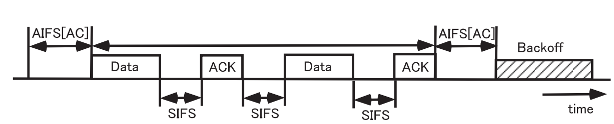

Fig. 1 IEEE 802.11e EDCA TXOP-bursting.

TXOP, successive frame exchange sequences are separated by SIFS. If a TXOP ends, the station goes into backoff. The TXOP-bursting can reduce the overhead due to contention if a station has more than one MAC frame in a source buffer.

3. Simulation Model

This paper first evaluates the application-level QoS of audio-video transmission between the AP and wireless stations by simulation with *network simulator version 2 (ns2)* [21].

Figure 2 illustrates the system configuration used in simulation of this study. In this paper we focus on a single BSS which includes an AP, multimedia stations, and data stations. The number of multimedia stations and that of data stations are denoted by M_M and M_D, respectively. All multimedia and data stations are located at the same distance (say R) from the AP.

In the simulation, we assume that a pair of audio (voice) and video streams is transferred between the AP and each of the multimedia stations in the uplink or downlink direction. The audio and video are transmitted as separate transport streams by using UDP/IP. Table 1 shows the specifications of the audio and video in the simulation. We use an audio stream of ITU-T G.711 μ-law and two MPEG1 video streams. We have prepared two kinds of the average bit rate, which is denoted by V_R here, for the same content (a clip of music video): V_R=400 kbps and 800 kbps. A video MU is transferred as one or more UDP datagrams. We assume that the maximum size of a UDP datagram is 1472 bytes in its payload.

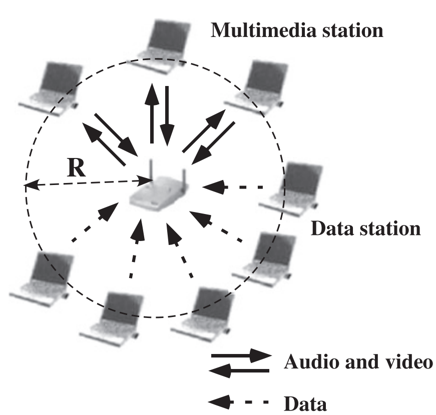

Fig. 2 System configuration for simulation.

Table 1 Specifications of audio and video.

item	audio	video
coding scheme	G.711 μ –law	MPEG1
image size [pixels]	–	320 × 240
picture pattern	–	IPPPPP
average MU size [bytes]	1000	5000
average MU rate [MU/s]	8	20
average inter-MU time [ms]	125	50
average bit rate [kbps]	64	400, 800
measurement time [s]	20	20

The data stations generate fixed-size UDP datagrams of 1472 bytes each in its payload at exponentially distributed intervals and send them to the AP. The average load per data station is 1000 kbps. The header size of a UDP datagram and that of an IP datagram are 8 bytes and 20 bytes, respectively.

The parameter values we use in the simulation are as follows. Table 2 shows parameter values of EDCA. These are default EDCA parameter values in [1]. We also use parameter values specified in the IEEE 802.11b standard for the DSSS physical layer at the channel rate of 11 Mbps; that is, the duration of a slot is equal to 20 μs, $DIFS$=50 μs, and $SIFS$=10 μs [19].

In the following numerical results, we set M_M=3 and M_D=5 from the following reason. We have confirmed through simulation that application-level QoS of audio and video transmission for $M_M \leq 3$ is kept high under specifications of audio and video described in Table 1 and IEEE 802.11 physical layer at the channel rate of 11 Mbps when BER is very small. We have also confirmed through simulation that uplink video quality and downlink one deteriorate if $M_M > 4$ and $M_M > 3$, respectively, when V_R=800 kbps. In addition, we have also found that audio and video quality is hardly affected by data traffic when M_D=5.

In the simulation we utilize freespace model of ns2 as the propagation model [22]. In ns2, the signal strength of a MAC frame can be calculated by the propagation model and distance between the transmitter and receiver. In calculating SNR, we assume receiver noise strength based on Orinoco 802.11b Card [22]. We also use empirical curves of BER versus SNR provided by Intersil wireless LAN chipset to obtain BER [23]. In the following sections, we will show numerical results for 145 m ≤ R ≤ 170 m and will examine the effect of TXOP-bursting on application-level QoS. We have confirmed through simulation that video quality for R ≤ 145 m is kept high even if TXOP-bursting is not carried out under our propagation model and traffic specifications described in this section. On the other hand, if R ≥ 170 m, video quality becomes low owing to transmission errors even if TXOP-bursting is utilized. The relationship between distance R and BER is shown in Table 3.

In addition, we also make the following assumptions in

Table 2 Parameters of EDCA.

media	AC	AIFS[μs]	CW_{min}	CW_{max}
audio	3	50	7	15
video	2	50	15	31
data	1	70	31	1023

Table 3 Relationship between distance R and BER.

R [m]	BER [$\times 10^{-5}$]
145	1.8
150	2.4
155	3.1
160	4.1
165	5.6
170	7.0

the simulation.

1. When a station cannot receive an ACK for the first MAC frame in a burst within a certain timeout period because of collision or transmission error, it goes into backoff and contends for the medium again.
2. The interval of the timeout is equal to the sum of an SIFS, transmission time of an ACK, and an $AIFS[AC]$. This interval is long enough for a station to judge whether the transmission of a MAC frame is successful or not.
3. When a station succeeds in sending the first MAC frame of access category AC, it obtains a TXOP and retains the medium for an interval of $TXOP_{Limit}[AC]$. This interval is indicated in Duration/ID field of the MAC header. However, if the station does not need to keep the medium for an interval of the $TXOP_{Limit}[AC]$ to finish sending all pending MAC frames and corresponding ACK frames, it retains the medium only for an interval required to transmit the pending MAC frames.
4. If a station cannot receive an ACK for a MAC frame after the first one within the timeout period during the TXOP, it keeps the medium and tries to retransmit the same MAC frame after the timeout occurs.
5. The maximum allowable number of retransmissions of a MAC frame is seven.
6. Each source buffer at the MAC layer in a station or the AP can accommodate a maximum of 50 MAC frames; a newly generated MAC frame is discarded if its buffer does not have space to accommodate the MAC frame.
7. There are no hidden stations, and the *Request To Send (RTS)/Clear To Send (CTS)* frames are not exchanged before transmission of a MAC frame.

4. Application-Level QoS Assessment

In this section we present simulation results of the application-level QoS of audio-video transmission with EDCA. The duration of each simulation run was taken to be 20 sec. We calculated the 95-percent confidence intervals of the simulation results. However, if the interval is smaller than the size of the corresponding simulation symbol in the figure, we do not show it there.

4.1 Application-Level QoS Parameter

In our study, we adopt nine application-level QoS parameters to evaluate audio and video quality. First, we use the *MU loss ratio*, which is denoted by L_a for audio and L_v for video, and indicates the ratio of the number of MUs lost to the number of MUs generated by multimedia stations or the AP. Second, we adopt *throughput* for audio T_a and that for video T_v. The throughput is defined as the average number of bits in a second transferred from the application layer of a source to that of the destination. Third, we use the *average MU delay*. This parameter is denoted by D_a for audio and D_v for video. The average MU delay means the average time from the moment an MU is generated at a source until the moment the MU is received at the destination. These parameters are used to evaluate the efficiency of audio and video transfer.

In addition, we treat the *coefficient of variation of output interval* for audio C_a and that for video C_v. The coefficient of variation of output interval represents the smoothness of the output flow and is the ratio of the standard deviation of MU output interval to the average output interval. We also select the *mean square error of inter-stream synchronization E_{int}*. The mean square error of inter-stream synchronization is an indicator of "lip-sync" and is the average square of 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 means the output time of the corresponding audio MU plus the difference between the timestamps of the two MUs. The coefficient of variation of output interval and mean square error of inter-stream synchronization is used to measure intra-stream and inter-stream synchronization quality, respectively. As the values of these parameters decrease, the quality of media synchronization becomes better.

In addition, we also use *data throughput* as an application-level QoS parameter to evaluate the efficiency of information transfer from data stations. The data throughput is the average number of bits in a second transferred from the application layer of a data station to that of the AP.

4.2 Comparison of TXOP-Bursting and No TXOP-Bursting

We now discuss how the TXOP-bursting can improve the application-level QoS in the presence of transmission errors. Figures 3 and 4 show the average MU delay for audio and that for video, respectively, as a function of the distance R between the AP and each station. In Fig. 5 we plot the MU loss ratio for video as a function of R. Figure 6 reveals the data throughput versus R. We also suppose the average bit rate for video V_R=800 kbps.

In Figs. 3 through 6, we show four cases: TXOP-bursting (Downlink), No TXOP-bursting (Downlink), TXOP-bursting (Uplink), and No TXOP-bursting (Uplink). TXOP-bursting means that MAC frames are transmitted with the TXOP-bursting scheme, while in No TXOP-bursting, only one MAC frame is sent in a TXOP. (Downlink) and (Uplink) indicate stream transmission in the downlink direction and that in the uplink one, respectively. We also set $TXOP_{Limit}[3]$=3.264 ms and $TXOP_{Limit}[2]$=6.016 ms. $TXOP_{Limit}[3]$ and $TXOP_{Limit}[2]$ correspond to $TXOP_{Limit}$ for audio and that for video, respectively. These are default values of the standard [1]. We also assume that the data stations do not use the TXOP-bursting.

First, we discuss the audio quality of EDCA, using Fig. 3. From this figure, we can find that the values of the average MU delay for the four cases are less than 20 ms for all the values of R shown. This is because audio transmis-

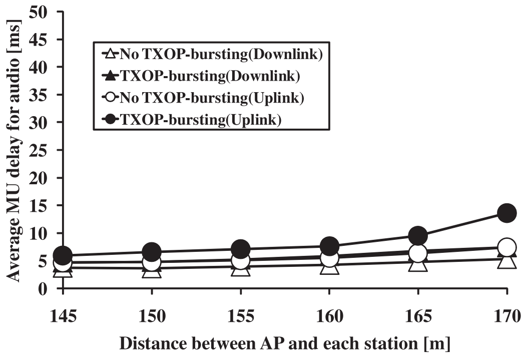

Fig. 3 Average MU delay for audio (V_R=800 kbps, $TXOP_{Limit}[3]$ = 3.264 ms, $TXOP_{Limit}[2]$=6.016 ms).

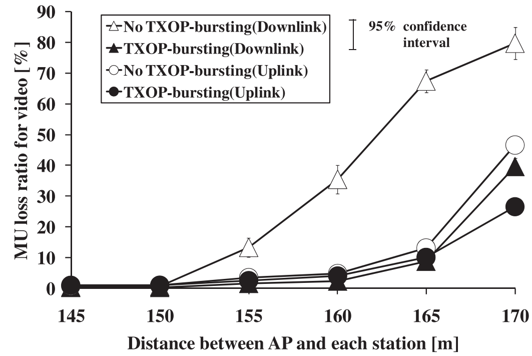

Fig. 5 MU loss ratio for video (V_R=800 kbps, $TXOP_{Limit}[3]$=3.264 ms, $TXOP_{Limit}[2]$=6.016 ms).

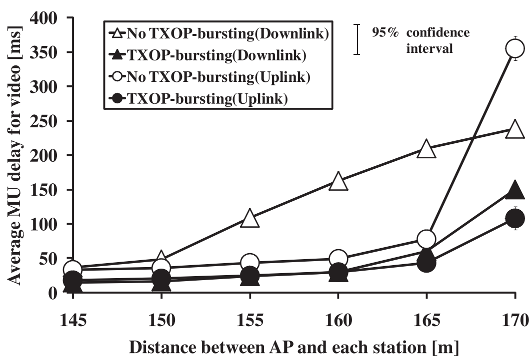

Fig. 4 Average MU delay for video (V_R=800 kbps, $TXOP_{Limit}[3]$=3.264 ms, $TXOP_{Limit}[2]$=6.016 ms).

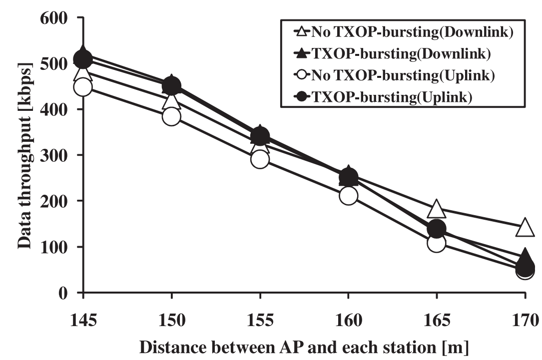

Fig. 6 Data throughput (V_R=800 kbps, $TXOP_{Limit}[3]$=3.264 ms, $TXOP_{Limit}[2]$=6.016 ms).

sion is given higher priority than video and data transmission. We should note in this figure that the average MU delay for TXOP-bursting becomes slightly larger than that for No TXOP-bursting since an interval of a TXOP for video transmission becomes longer if the TXOP-bursting scheme is utilized. We have confirmed through simulation that the MU loss ratio for audio is less than 1 % and the throughput for audio is about 64 kbps for all the values of R shown.

We next discuss the video quality, referring to Fig. 4. Figure 4 shows that the values of the average MU delay for TXOP-bursting (Downlink) are much smaller than those for No TXOP-bursting (Downlink). In the case of downlink video transmission, many video MUs are generated and are kept into the source buffer of the AP because all downlink video MUs are transferred only by the AP. Therefore, the TXOP-bursting scheme is very attractive since the AP can send more than one video MU successively in a TXOP. On the other hand, this figure shows that the TXOP-bursting for the uplink video transmission is not so effective as that for the downlink video transmission. This is because uplink video MUs are transmitted by separate multimedia stations. It should also be noted in Fig. 4 that the difference in the average MU delay between TXOP-bursting (Uplink) and No TXOP-bursting (Uplink) begins to increase as R becomes larger beyond 165 m. In the case where $R > 165$ m, the wireless channel is congested with video transmission owing to excessive contention by retransmitted MAC frames.

We then observe in Fig. 5 that the MU loss ratio for the downlink video transmission is also significantly improved with the TXOP-bursting scheme if $R > 150$ m. When R is large, video MUs are frequently retransmitted owing to transmission errors. This leads to buffer overflow in the AP. This figure also shows that the MU loss ratio for TXOP-bursting (Uplink) is almost the same as that for No TXOP-bursting (Uplink) except R=170 m. We have also confirmed through simulation that the throughput for video is about 800 kbps if video MU loss does not occur. However, the throughput for video deteriorates as the MU loss ratio increases.

Next, we discuss the data throughput in Fig. 6. This figure indicates that the data throughput decreases as R increases in all the cases. The data transmission is given lower priority than the audio and video transmission. Therefore, the data stations have fewer chances to send data MAC frames if retransmissions of audio and video MUs frequently occur. Figure 6 also reveals that the data throughput slightly decreases if the TXOP-bursting is performed for downlink audio and video transmission when $R > 160$ m. This is because a longer interval of TXOP leads to a smaller capacity allocation for the data transmission on heavy traffic conditions. On the other hand, the data throughput slightly increases if the TXOP-bursting is carried out on the uplink because the overhead due to contention by the multimedia stations is reduced.

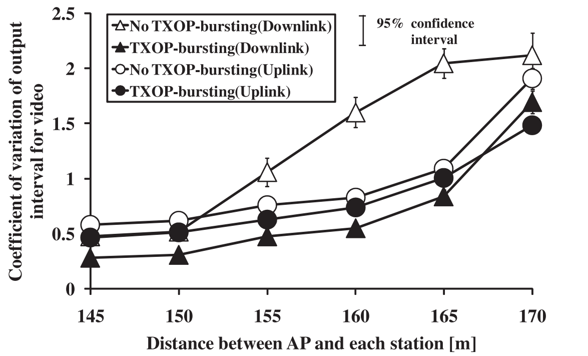

Fig. 7 Coefficient of variation of output interval for video (V_R=800 kbps, $TXOP_{Limit}$[3]=3.264 ms, $TXOP_{Limit}$[2]=6.016 ms).

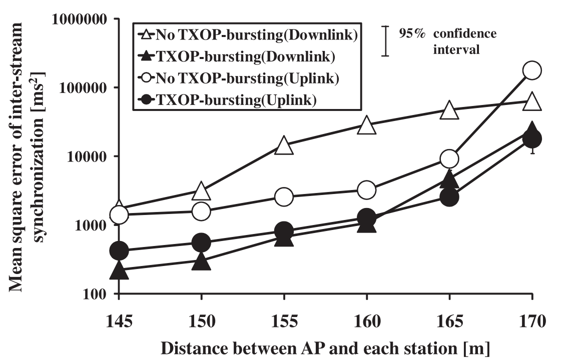

Fig. 8 Mean square error of inter-stream synchronization (V_R=800 kbps, $TXOP_{Limit}$[3]=3.264 ms, $TXOP_{Limit}$[2]=6.016 ms).

We then discuss the media synchronization quality, using Figs. 7 and 8. Figure 7 plots the coefficient of variation of output interval for video as a function of the distance R between the AP and each station. In Fig. 8 we plot the mean square error of inter-stream synchronization as a function of R. In these figures, we set V_R=800 kbps, $TXOP_{Limit}$[3]=3.024 ms, and $TXOP_{Limit}$[2]=6.016 ms.

We can find that Fig. 7 exhibits similar characteristics to Figs. 4 and 5. This figure shows that the values of the coefficient of variation of output interval for video increase for the four cases as R becomes longer since a larger value of BER leads to an increase of retransmission traffic. Figure 7 also shows that the values of the coefficient of variation of output interval for TXOP-bursting (Downlink) are much smaller than those for No TXOP-bursting (Downlink) when R > 150 m. This is because all downlink video MUs are transmitted by the AP only. In the meanwhile, we have confirmed through simulation that values of the coefficient of variation of output interval for audio in the four cases are less than 0.2.

In Fig. 8 we then observe that the mean square error can be improved by the TXOP-bursting for both downlink and uplink cases. If the TXOP-bursting is carried out, the average MU delay for video becomes smaller. Therefore, the difference in the output time between an audio MU and the corresponding video MU decreases as seen from Figs. 3 and 4.

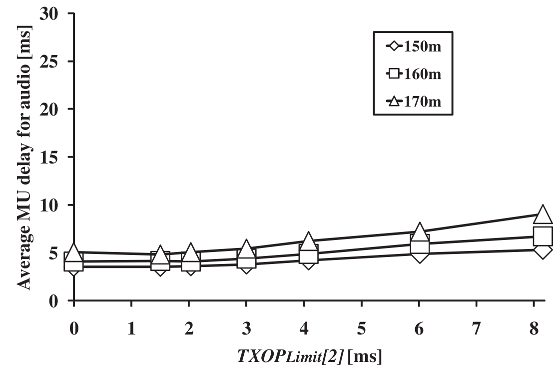

Fig. 9 Average MU delay for audio (V_R=800 kbps, $TXOP_{Limit}$[3]= 3.264 ms).

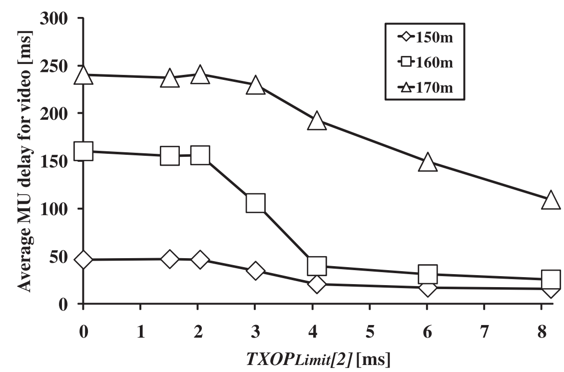

Fig. 10 Average MU delay for video (V_R=800 kbps, $TXOP_{Limit}$[3]= 3.264 ms).

From the above observations, we can say that the TXOP-bursting can improve the application-level QoS of EDCA in the presence of transmission errors, particularly when the AP sends audio and video streams to multimedia stations.

4.3 The Effect of $TXOP_{Limit}$ on the Application-Level QoS

We next examine how the value of $TXOP_{Limit}$ for video affects the application-level QoS. Figure 9 shows the average MU delay for audio as a function of $TXOP_{Limit}$[2]. Figures 10 through 12 reveal the average MU delay, MU loss ratio, and coefficient of variation of output interval for video as a function of $TXOP_{Limit}$[2], respectively. In Figs. 9 through 12, we set V_R=800 kbps. Furthermore, Fig. 13 plots the average MU delay for video when V_R=400 kbps. Figures 9 through 13 indicate simulation results when the AP sends audio and video flows to multimedia stations in the downlink direction. In these figures we set $TXOP_{Limit}$[3]=3.264 ms. Video quality can be improved by TXOP-bursting because one video MU is transferred as more than one MPDU. On the other hand, in audio transmission, one audio MU is sent as one audio MPDU. Therefore, we set $TXOP_{Limit}$[3] to the default value of the IEEE 802.11e standard and examine the effect of the $TXOP_{Limit}$[2] on the application-level QoS. Note that $TXOP_{Limit}$[3]=3.264 ms is

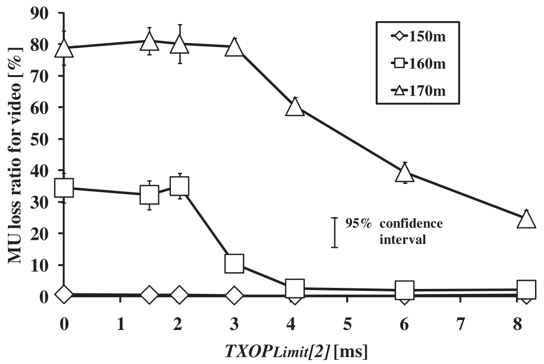

Fig. 11 MU loss ratio for video (V_R=800 kbps, $TXOP_{Limit}$[3]=3.264 ms).

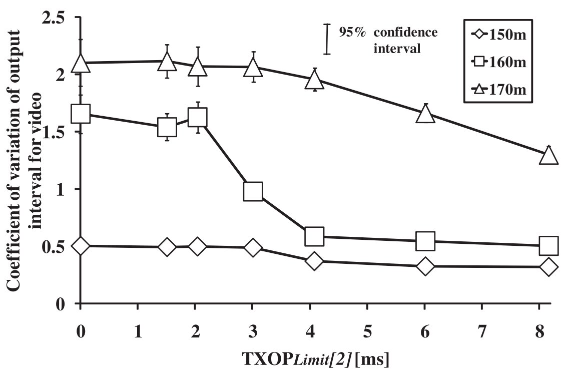

Fig. 12 Coefficient of variation of output interval for video (V_R= 800 kbps, $TXOP_{Limit}$[3]=3.264 ms).

long enough to transmit an audio MU specified in Table 1.

For each of R=150 m, 160 m, and 170 m, we show seven values of $TXOP_{Limit}$[2]: 0 ms, 1.504 ms, 2.040 ms, 3.008 ms, 4.080 ms, 6.016 ms, and 8.160 ms†. $TXOP_{Limit}$[2]=0 ms means that the TXOP-bursting scheme is not used for video transmission.

First, we can find in Fig. 9 that the average MU delay for audio becomes slightly larger as $TXOP_{Limit}$[2] increases since an interval of a TXOP for video transmission becomes longer. However, the values of the average MU delay for audio are less than 10 ms even if $TXOP_{Limit}$[2]=8.160 ms.

We then discuss video quality, using Figs. 10 through 12. In the case of R=150 m, these figures indicate that video quality is kept high for all the values of $TXOP_{Limit}$[2] shown since traffic volume of retransmitted MAC frames is small. Next, we can observe from Figs. 10 through 12 that video quality for R=160 m is very good if the value of $TXOP_{Limit}$[2] is equal to or greater than 4.080 ms. We have confirmed that in the case of V_R=800 kbps, an I picture frame is transferred as about three or four MPDUs. Therefore, the TXOP-bursting scheme is effective since more than one MPDU is sent successively in a TXOP. These figures indicate that the value of $TXOP_{Limit}$[2] should be greater than or equal to 4.080 ms under the simulation conditions. In the case of R=170 m, Figs. 10 through 12 show that a larger value of $TXOP_{Limit}$[2] leads to higher video quality. However, the MU loss ratio for video is more than 20% even if

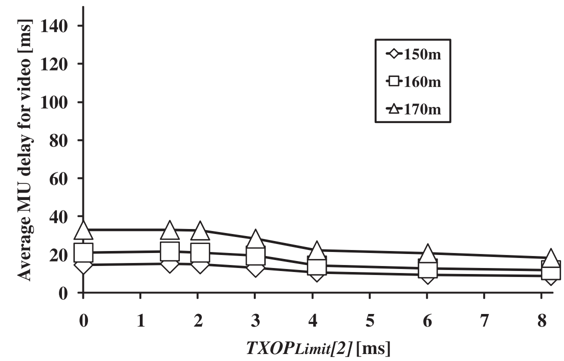

Fig. 13 Average MU delay for video (V_R=400 kbps, $TXOP_{Limit}$[3]= 3.264 ms).

$TXOP_{Limit}$[2]=8.016 ms. This is because the wireless channel is congested even if the TXOP-bursting is carried out.

Furthermore, Fig. 13 shows that the effectiveness of the TXOP-bursting is very small if V_R=400 kbps. In the case of V_R=400 kbps, the volume of video traffic is small. Therefore, the queue length of the video source buffer in the AP is short.

5. QoE Assessment

In the previous section, we evaluated the application-level QoS for audio-video transmission with the TXOP-bursting through simulation. Numerical results showed that the video quality at the application-level; that is, both intra-stream and inter-stream synchronization, is improved by the TXOP-bursting for downlink audio and video transmission. On the other hand, in the case of uplink multimedia transmission, the MU loss ratio for video with the TXOP-bursting is almost the same as that without the scheme, while the mean square error of inter-stream synchronization is slightly improved by the scheme. Therefore, it is difficult to judge the effectiveness of the TXOP-bursting only by the application-level QoS.

In this section, we assess the QoE for audio-video transmission with the TXOP-bursting in a quantitative way by a subjective experiment. We utilize a psychometric method referred to as the method of successive categories. We also perform QoS mapping between application-level and user-level with principal component analysis and multiple regression analysis.

5.1 Assessment Method

The subjective assessment 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. The number of assessors is 17, and their ages were 20 s. We used five categories of impairment of the *rating-scale method*; that is, each assessor

†Note that the range of the $TXOP_{Limit}$ is 32 μs to 8.160 ms in the 802.11e standard.

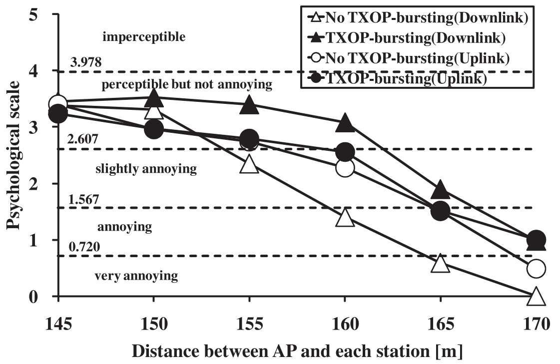

Fig. 14 Psychological scale versus the distance between AP and each station (V_R=800 kbps, $TXOP_{Limit}$[3]=3.264 ms, $TXOP_{limit}$[2]=6.016 ms).

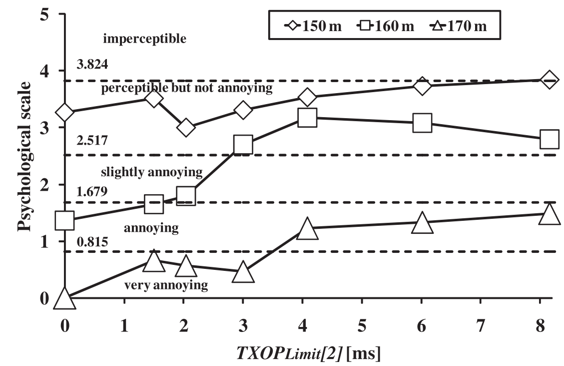

Fig. 15 Psychological scale versus $TXOP_{Limit}$[2] (V_R=800 kbps, $TXOP_{Limit}$[3]=3.264 ms).

was shown the test samples and was asked to classify each sample into one of the following five categories with their scores: "imperceptible" assigned score 5, "perceptible, but not annoying" 4, "slightly annoying" 3, "annoying" 2, and "very annoying" 1.

As the QoE parameter, we utilize the *psychological scale* instead of the *mean opinion score (MOS)*. The MOS is often used in subjective assessment and is calculated by averaging the subjective scores under an implicit assumption that the difference in scores between any two successive categories means the same magnitude of the assessor's sensation. However, this assumption is not always true since the MOS is an *ordinary scale*; the scores assigned to the categories only have a greater-than-less-than relation between them. On the other hand, the psychological scale is an *interval scale*; an interval between the scale values means a distance between amounts of the sensory attribute measured [20].

To verify the obtained psychological scale, we have performed Mosteller's test. From the Mosteller's test, we were not able to reject the hypothesis that the obtained psychological scale fits the observed data at a significance level of 0.01.

5.2 Assessment Results and QoS Mapping

We now examine how the TXOP-bursting can improve the QoE, referring to Figs. 14 and 15. Figure 14 depicts the psychological scale as a function of the distance between the AP and each station in the four cases as earlier. In this figure, we set $TXOP_{Limit}$[2]=6.016 ms. Figure 15 shows the psychological scale for the downlink audio-video transmission versus the $TXOP_{Limit}$[2] for three values of R: R=150 m, 160 m, and 170 m. In these figures, we have selected the minimum value of the psychological scale as the origin of the ordinate, and each of four horizontal broken lines indicates the boundary of a category. We set V_R=800 kbps, and $TXOP_{Limit}$[3]=3.264 ms in these figures.

From Fig. 14 we find that the values of the psychological scale for the four cases become smaller as the distance increases. This implies that the QoE deteriorates as BER becomes larger. We then notice in Fig. 14 that the values of the psychological scale for No TXOP-bursting (Downlink) are much smaller than those for TXOP-bursting (Downlink). Thus, we can say that the QoE can be improved by utilizing the TXOP-bursting for downlink audio and video transmission. In contrast to the downlink case, TXOP-bursting (Uplink) has almost the same values of the psychological scale as those for No TXOP-bursting (Uplink).

We have performed QoS mapping between application-level and user-level for Fig. 14 by means of principal component analysis and multiple regression analysis as in [20]. In the analysis, we selected the nine application-level QoS parameters as the *predictor variables* and the psychological scale as the *criterion variable*.

First, we have carried out the principal component analysis to decrease the number of predictor variables [20]. As a result of the principal component analysis, we selected the first three principal components since the *cumulative contribution rate* for the first three principal components becomes 97.807%. On the basis of the principal component analysis, we classified the nine application-level QoS parameters into the following three groups.

group A) D_a, C_a
group B) L_a, T_a, and D_v
group C) L_v, T_v, C_v, and E_{int}

Then, we have performed the multiple regression analysis as in [20]. We calculated multiple regression lines by picking up one predictor variable from each class for all combinations of the predictor variables. As a result, we removed the predictor variables in group C since those variables do not make any significant contributions to multiple regression lines. We have again performed multiple regression analysis and have selected a multiple regression line that achieves the largest value of the *contribution rate adjusted for degrees of freedom*. Finally, we have obtained multiple regression lines for No TXOP-bursting (Downlink), TXOP-bursting (Downlink), No TXOP-bursting (Uplink), and TXOP-bursting (Uplink) as follows:

$$\hat{S}_{NoTXOP\_D} = 5.201 - 0.342D_a - 0.014D_v \quad (R^{*2} = 0.999) \tag{1}$$

$$\hat{S}_{TXOP\_D} = 6.303 - 0.544D_a - 0.009D_v$$

$$(R^{*2} = 0.999) \quad (2)$$

$$\hat{S}_{NoTXOP\_U} = 8.068 - 1.047D_a - 0.010D_v \quad (R^{*2} = 0.962) \quad (3)$$

$$\hat{S}_{TXOP\_U} = 7.126 - 0.764D_a - 0.039D_v \quad (R^{*2} = 0.989) \quad (4)$$

In the above equations, $\hat{S}_{NoTXOP\_D}$, $\hat{S}_{TXOP\_D}$, $\hat{S}_{NoTXOP\_U}$, and $\hat{S}_{TXOP\_U}$ represent the estimate of the psychological scale for No TXOP-bursting (Downlink), TXOP-bursting (Downlink), No TXOP-bursting (Uplink), and TXOP-bursting (Uplink), respectively. R^{*2} denotes the contribution rate adjusted for degrees of freedom. From these equations, we can calculate the estimated values of the psychological scale from the two application-level QoS parameters. In equations (1) through (4), we notice that D_a and D_v make significant contributions to the multiple regression lines.

We next discuss the effect of the value of $TXOP_{Limit}[2]$ on the psychological scale. In the case of $R = 150$ m, Fig. 15 shows that the QoE is kept high for all the values of $TXOP_{Limit}[2]$ shown. This figure also shows that the QoE for R=160 m becomes higher as the value of $TXOP_{Limit}[2]$ increases until 4.080 ms. This means that the QoE becomes low if the value of $TXOP_{Limit}$ for video is too small. In addition, Fig. 15 shows that in the case of R=170 m, the QoE is low even if $TXOP_{Limit}[2]$=8.160 ms.

We have again performed QoS mapping between application-level and user-level for Fig. 15 and have obtained multiple regression lines for $R = 150$ m, 160 m, and 170 m as follows:

$$\hat{S}_{150} = 4.955 - 0.192D_a - 0.022D_v \quad (R^{*2} = 0.808) \quad (5)$$

$$\hat{S}_{160} = 4.784 - 0.206D_a - 0.014D_v \quad (R^{*2} = 0.887) \quad (6)$$

$$\hat{S}_{170} = 3.865 - 6.603L_a - 0.008D_v \quad (R^{*2} = 0.957) \quad (7)$$

In the above equations, $\hat{S}_{150}$, $\hat{S}_{160}$, and $\hat{S}_{170}$ represent the estimate of the psychological scale for $R = 150$ m, 160 m, and 170 m, respectively. Note that in the case of $R = 170$ m, L_a makes a higher contribution to the multiple regression line than D_a. We have confirmed through simulation that the values of L_a for $R = 160$ m and 170 m are 0.01–0.03% and 0.2–0.3%, respectively, for all the values of $TXOP_{Limit}[2]$ shown.

6. Conclusions

This paper studied the application-level QoS of audio-video transmission with EDCA of the IEEE 802.11e MAC through simulation. In addition, the QoE was assessed by a subjective experiment. We examined how the TXOP-bursting can improve the audio and video quality in the presence of transmission errors.

First, we examine the effect of the TXOP-bursting on the application-level QoS when a pair of audio and video streams is transferred between the AP and each station in the uplink or downlink direction. Numerical results showed that the TXOP-bursting can improve many metrics of video quality in terms of average MU delay, MU loss ratio, and coefficient of variation of output interval for various distances between the AP and stations. In particular, this scheme is very effective when the AP transmits video MUs to each multimedia station in the downlink direction. This is because MAC frames for retransmission as well as newly generated ones can be transmitted successively in a TXOP. The simulation results also showed that the video quality even with the TXOP-bursting deteriorates when the distance is too long. We then examined how the value of $TXOP_{Limit}$ affects the video quality and found that the video quality can be degraded if the value of $TXOP_{Limit}$ for video is too small.

Furthermore, we investigated the QoE by a subjective experiment. As a result, we found that the QoE can be improved by utilizing the TXOP-bursting. In addition, we also carried out QoS mapping between application-level and user-level with principal component analysis and multiple regression analysis. As a result, we obtained multiple regression lines to estimate the QoE from the application-level QoS.

Our future work includes investigation of the application-level QoS and QoE with other physical data rates and video coding rates.

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