PAPER Special Issue on Mobile Multimedia Communications

A New Resource Management Scheme for Multiple Video Transmission in Wireless Environment

Ryoichi SHINKUMA[†], *Student Member*, **Minoru OKADA**[††], *and* **Shozo KOMAKI**[†], *Regular Members*

SUMMARY In this paper, we propose a new resource management scheme for multiple video stream transmission in a wireless environment. The proposed scheme adaptively determines transmission parameters such as the number of assigned time slots, modulation format, and forward error correction (FEC) code rate according to the required bit rate and error sensitivity of the video stream as well as the channel state. The computer simulation results show that the proposed scheme drastically improves the image quality degradation due to channel errors.

key words: *wireless communications, radio resource management, adaptive transmission, video stream transmission*

1. Introduction

Wireless home link [1] is a broadband wireless interfaces amongst a home server and the user terminals. It enables home users to connect to the Internet as well as to retrieve audio and video streams from the home server. In order to accommodate user terminals to receive high quality audio and video streams such as MPEG-2 (Moving Picture Experts Group) at the same time, the wireless home link requires high transmission rate. IEEE802.11a and b, Home RF, and Wireless 1394 standards are candidates for the air interface of the wireless home link capable of providing a few tens Mbps of digital transmission. The wireless channel is not only strictly band-limited but also impaired by the channel error due to noises, interferences, fading, and shadowing. That is, it is difficult to accommodate multiple video streams and WWW data in one wireless home link connection.

The video coding formats for video transmission through error prone and low bit rate channels are proposed in [2]–[5]. MPEG-4 [2] can immediately recover the image quality degraded by channel errors with new coding techniques such as the periodic resynchronization markers and RVLC (Reversible Variable Length Codes). In [3], the feedback-based video coding techniques for error-prone environment has been proposed. This technique selects the coding mode of each block (inter, intra, for predictive or non predictive coding, respectively) and, the number and location of synchronization marker, such that, given the channel condition and the decoder error concealment. However, the video sources of digital TV, digital video or web streaming service, which can be requested by the users in the wireless home link, have already been encoded with MPEG-1 or MPEG-2 format. Therefore, in order to applying new coding schemes such as MPEG-4 and feed-back adaptive coding to the wireless home link, the home gateway must convert the format of video sources by using transcoding techniques [4], [5].

On the other hand, reference [6]–[8] have proposed the efficient error control scheme for wireless video transmission. The unequal error protection scheme proposed in [6] divides video sources into some classes based on the error sensitivity and assigns high rate FEC (Forward Error Correction) coding to the error sensitive data. This scheme must analyze the detail structures of video sources, for example, DCT (Discrete Cosin Transform) coefficients, motion vectors and quantizer levels. The new ARQ (Automatic Repeat reQuest) scheme for video transmission in a wireless environment has been proposed in [7] and [8]. Although the retransmission is the error control for non-real time data, they can apply ARQ error control to the real time streams by reducing retransmission delay.

However, since the conventional schemes assume the single video transmission, they cannot make efficient utilization of the radio frequency. In wireless video transmission, we should focus the total video quality of two or more video streams since users share the strictly limited bandwidth. L. Hanzo, C.H. Wong and P. Cherriman have proposed an adaptive wireless transmission scheme for mobile video telephony in [9]. This system chooses video transmission modes, which are formed by the coding mode, video rate, frame rate, FEC code rate, ARQ type and modulation level, according to the channel condition and the number of users. Although the multiple video transmission is assumed in this scheme, the radio resource management for multi-users is not performed. This is because it makes equal assignment of bandwidth and transmission format for each user. In addition, as wireless video environment, they don't assume the wireless home link but the mobile environment.

Manuscript received February 8, 2002.

Manuscript revised May 10, 2002.

[†]The authors are with the Department of Communications Engineering, Faculty of Engineering, Osaka University, Suita-shi, 565-0871 Japan.

[††]The author is with the Graduate School of Information Science, Nara Institute of Science and Technology, Ikoma-shi, 630-0101 Japan.

As a resource management protocol, IEEE802.11e working group is now making a standard to give the QoS (Quality of Service) control capability to IEEE802.11b and IEEE802.11a wireless LAN. However, its final draft has not been released. Whitecap2 [10] is a protocol based on the IEEE802.11e. It gives the priority to the real time traffics such as the video, voice and audio stream in order to guarantee the delay performance by assigning much bandwidth to the high priority packets, namely, dynamic TDMA (Time Division Multiple Access). However, Whitecap2 is not sufficient for resource management of multiple video streams in a wireless channel.

In this paper, we propose a new resource management scheme for multiple video stream transmission with a simple and efficient algorithm in a wireless environment. The proposed scheme employs an adaptive transmission and dynamic resource assignment. The proposed scheme determines the transmission parameters for each video packet according to the error sensitivity, required bit rate, and the channel state. The transmission parameters consist of the number of time slots, modulation level, and FEC code rate. For example, when two video packets are transmitted, more error-sensitive one is transmitted with reliable transmission format such as QPSK (Quadrature Phase Shift Keying) with FEC code. On the other hand, error-insensitive one is sent to the user by 64 QAM (Quadrature Amplitude Modulation), which is the bandwidth efficient transmission format. In this process, the proposed system assigns time slots with meeting bit rate required by each video stream. The conventional schemes have tried to improve the degradation of the video quality for single video stream, while the proposed scheme is capable of improving the video quality by the resource management for two or more video streams. In addition, in the proposed scheme, since the error sensitivities of the video packets are estimated only by analyzing the video header information, the proposed scheme does not require the high computational cost.

The rest of the paper is organized as follows. Section 2 proposes the multiple video transmission scheme, followed by the computer simulation results shown in Sect. 3. Finally, concluding remark is given in Sect. 4. In this paper, we assume only the MPEG-2 format as video coding format in the home gateway.

2. Multiple Video Stream Transmission System

2.1 System Description

Figure 1 shows the block diagrams of proposed multi video transmission system. The home gateway is a gateway to the Internet, as well as the digital TV and digital video storage. In the home gateway, first, the MPEG analyzer reads the headers of input MPEG data and sends the profile, which describes the structure of input MPEG data, to the error-sensitivity estimator. The error-sensitivity estimator calculates the error sensitivity value, α, from the profile. Then, the MPEG analyzer sends MPEG sources, the value of required bit rate, β, and α to the radio agent. The proposed scheme employs the dynamic TDM (Time Division Multiplexing) for downlink. The radio agent determines the transmission parameters such as the number of time slots, u, modulation format, k, and RS (Reed-Solomon) code rate, R, for the transmitted video packets. The proposed algorithm minimizes the mean error after decoding by determining the transmission parameters according to the channel BER (Bit Error Rate). The channel CNR (Carrier to Noise power Ratio) is estimated at each user terminal and sent back to the home gateway. The channel BER for determining the transmission parameters is then calculated from the estimated CNR. Further detail of the algorithm is explained in the next section. In the PPU (Packet Processing Unit), the data-link preamble, which indicates u, k and R decided by the radio agent, is attached to each transmitted packet. The time slot manager, RS encoder and 2^k-QAM modulator are controlled according to the parameters shown by the preamble. In addition, the preambles of the packets are transmitted with BPSK (Binary Phase Shift Keying), which is the most reliable modulation format.

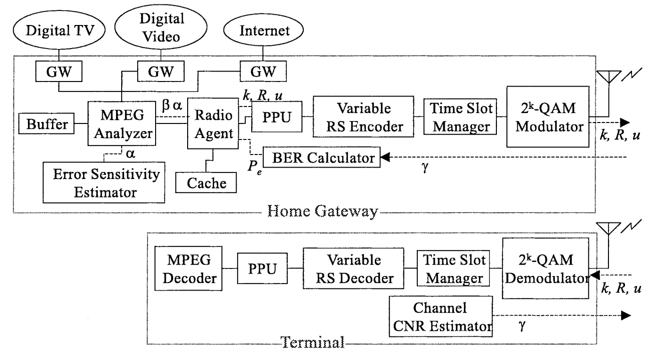


Fig. 1 Block diagram of proposed multiple video transmission system.

On the other hand, the receiver on the terminal demodulates the preambles data in the received signals and performs the demodulation and error control for the received packets according to the preamble messages. The time slot manager on the terminal picks up own requested packets in all the received packets. The PPU re-build the video streams from the received packets and applied to the MPEG decoder.

2.2 Resource Management Algorithm

TDM frame in the proposed system is shown in Fig. 2. In the following we assume the number of users is two in

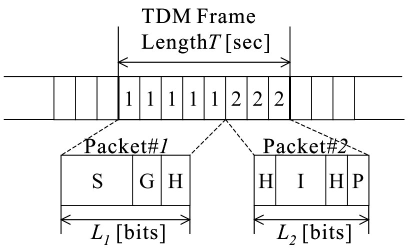


Fig. 2 Transmitted video packets for two users in TDM frame.

Table 1 Error sensitivity classes for MPEG-2 data.

Class S	Sequence header part
Class G	GOP header part
Class H	Picture header part
Class I	I picture
Class P	P picture
Class B	B picture

order to simplify the analysis. This implies two packets per frame. In this condition, the required bit rate for user $\#i(=1,2)$ satisfies the following inequality.

$$\beta_i \le \Delta f k_i R_i u_i / U; (i = 1, 2), \qquad (1)$$

where U is the number of slots per TDM frame. Δf [Hz] is the total bandwidth and β_i, u_i, k_i and R_i are the required bit rate, the number of slots, the modulation level, and the FEC code rate for user $\#i$, respectively. By multiplying Eq. (1) by the frame time interval T [sec], we can obtain the following relations

$$L_i = \beta_i T \le \Delta f T k_i R_i u_i / U, \qquad (2)$$

where L_i is the packet length per frame for user i.

The proposed system divides all bytes of MPEG-2 data into six classes as shown in Table 1 according to their error sensitivities. Class S, G, H, I, P and B are sequence header data, GOP (Group Of Pictures) header data, picture header data and I/P/B picture data, respectively. α_X is the error-sensitivity value of class X ($X \in \{S, G, H, I, P, B\}$). For instance, the packet #1 in Fig. 2 consists of the sequence header data, GOP header data and picture header data, so the error-sensitivity estimator in Fig. 1 sets the value of the error-sensitivity for packet #1, α_1, to the maximum value in α_S, α_G and α_H. On the other hand, the packet #2 includes the picture header data, I picture data and P picture data, so the value of error-sensitivity, α_2, is the maximum value in α_H, α_I and α_P. Then, we define the mean error of the video image after decoding, ϵ, as

$$\epsilon = \alpha_1 P_{e1}(k_1, R_1, \gamma_1) + \alpha_2 P_{e2}(k_2, R_2, \gamma_2), \qquad (3)$$

where P_{ei} is bit error rate of user $\#i$ estimated by the BER estimator in Fig. 1. The channel CNR, γ_i, are feedback from each terminal. The goal of the proposed algorithm is to minimize ϵ.

Figure 3 illustrates the flowchart diagram of the proposed algorithm. At first, we set $u_1 = u_2 = U/2$.

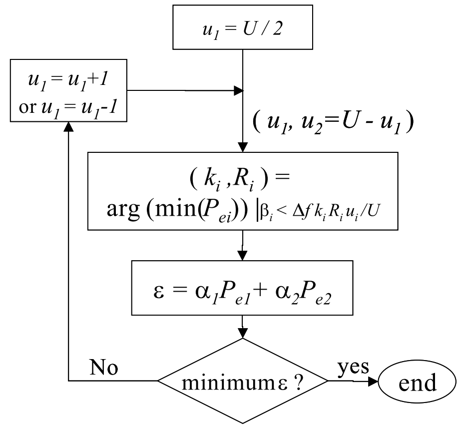


Fig. 3 Flowchart of proposed radio resource management algorithm.

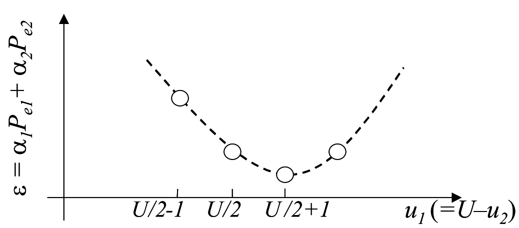


Fig. 4 Method to minimize mean error after decoding, ϵ.

After the number of slots, u_i, is decided, k_i and R_i are chosen for minimizing the mean error after decoding. In this minimization, k_i and R_i must satisfy the inequalities shown in Eq. (1). Therefore, if the number of slots assigned to user $\#i$ is small, the bandwidth efficient transmission parameters, high k_i and low R, will be chosen. In the proposed scheme, the code rate of RS code, R_i, is restricted to

$$R_i = \frac{C_i}{2^{k_i} - 1}; (C_i \le 2^{k_i} - 1), \qquad (4)$$

where C_i is the information symbol in code blocks. Figure 4 illustrates the mean error after decoding against the number of slots for the first user, u_1. The mean error after decoding, ϵ^0, ϵ^+ and ϵ^- are for the assigned number of slots $u_1 = U/2$, $U/2$+1 and $U/2$-1, respectively. If ϵ^0 is less than ϵ^+ and ϵ^-, we can decide $u_1 = u_2 = U/2$. On the other hand, if ϵ^+ is less than the others, the number of slots for user #1, u_1, is increased one by one (u_2 is decreased). If ϵ^- is least, u_1 is decreased (u_2 is increased). ϵ is enclosed to minimum value with increasing or decreasing u_1. In this way, the minimization for ϵ is carried out.

This algorithm is performed in frame-by-frame basis. The proposed algorithm assigns much radio resource to the user whose the channel CNR is low. Furthermore, reducing the resource for the error insensitive packets, more resource is assigned to error sensitive packets in order to give more reliable transmission.

3. Simulation Description

The computer simulation model is illustrated in Fig. 5. We assume that two users request MPEG-2 streams to the home gateway. User #1 requests the video source #1 (trees), and user #2 requests the video source #2 (jet). Table 2 shows the format of video sources in the simulation. For simplicity, the video streams are encoded in the CBR (Constant Bit Rate) mode, although the proposed scheme is capable of being encoded in the VBR (Valuable Bit Rate) mode. The MPEG-2 streams "trees" and "jet" require bit rates of 6.68 [Mbps] and 0.976 [Mbps], respectively. The packets for two video streams are transmitted by the home gateway through AWGN channel on the assumption that the receiver can compensate for the multipath fading. The proposed system employs QPSK, 16 QAM and 64 QAM as modulation formats, and valuable rate RS code as FEC error control. Table 3 shows the system configuration of the simulation.

In the proposed system, the error sensitivity for each class, α_S, α_G, α_H, α_I, α_P and α_B, is required. In order to decide α for each class, we performed the subjective assessment for image quality of video data. The subjective assessment performed in this paper is based on the ITU-R (International Telecommunication Union - Radio communication sector) 500–10 [11]. In §2.1 of ITU-R 500, the recommended general viewing conditions are described. In our tests, 17-inch CRT (Cathode Ray Tube) display with an aspect ratio of over 3 is used. That is, the height of the display is 0.20 [m], so PVD (Preferred Viewing Distance) = 1.8 [m]. In the simulation, since the observers assess the radical degradation of image quality due to bit errors, the precise adjustment of the monitor conditions are not required. As video sources for assessments, we use "trees" for user #1 and "jet" for user #2. In the video source #1, "trees," the background keeps still, while the tree trunks are shaken and some leaves are fallen by wind. On the other hand, the video source #2, "jet," is the image that the jet airplane moves in the air, and the background changes slowly. Although two video characteristics are different, both of them show the degradation of image quality clearly. The number of the observers is twenty, all of whom are no experts. The experts are not required in easy assessments. Most of the observers are university students. Their ages are from seventeen to twenty five and they consist of three females and seventeen males. Since the number of video samples for test is very large, we divided the assessment into the several sessions. The DSIS (Double-Stimulus Impairment Scale) method, which is indicated in §4 of ITU-R 500–10, was employed as subjective assessment method. DSIS Variant 2, where the reference sequence and the test sequence are presented twice, is employed in our tests. Figure 6 is the five-grade impairment scale of DSIS. Setting the maximum value of the evaluation to 5, the scores are measured based on the distance from zero. From the test results, the MOS (Mean Opinion Score) values of each test are calculated.

At first, we decide "acceptable BER bound" for each class with the subjective assessment. The acceptable BER bound is the rate of random errors that gives the image quality of 3 in MOS. Next, we set the error sensitivity α_X of each class to the inverse of the acceptable BER bound. For example, if the acceptable BER

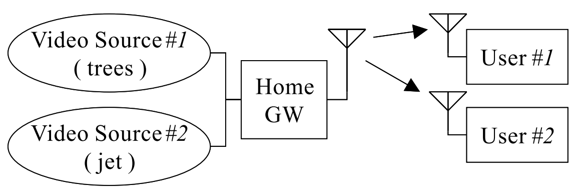


Fig. 5 Simulation model for multi video transmission.

Table 2 Formats of two video streams.

Video compression format	MPEG-2
MPEG Decoder	Software decoder w/o any error concealment functions
Frame rate	30 [fps]
Picture structure	Frame structure
GOP	I-P-B-B-P-B-B-P-B-B-P-B-B
Concealment motion vectors in intra macro blocks	not included
Brightness/color format	4:2:0
Bit rate	CBR("trees"=6.68 [Mbps], "jet"=0.976 [Mbps])
Data size	"trees"=8.4 [Mbyte], "jet"=1.9 [Mbyte]

Table 3 Simulation parameters.

Channel model	AWGN
CNR for user #1, γ_1	17–21 [dB]
CNR for user #2, γ_2	21 [dB]
Total bandwidth, Δf	2.2 [MHz]
Down link Multiplexing	Time Division Multiplexing
Frame length	10 [msec]
Time slots per Frame, U	32
Modulation types	QPSK, 16 QAM, 64 QAM
FEC code	Reed-Solomon code (Valuable rate)

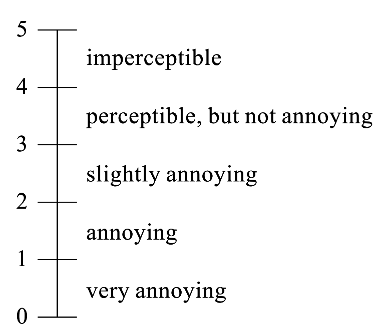


Fig. 6 Five-grade impairment scale of double-stimulus impairment scale method.

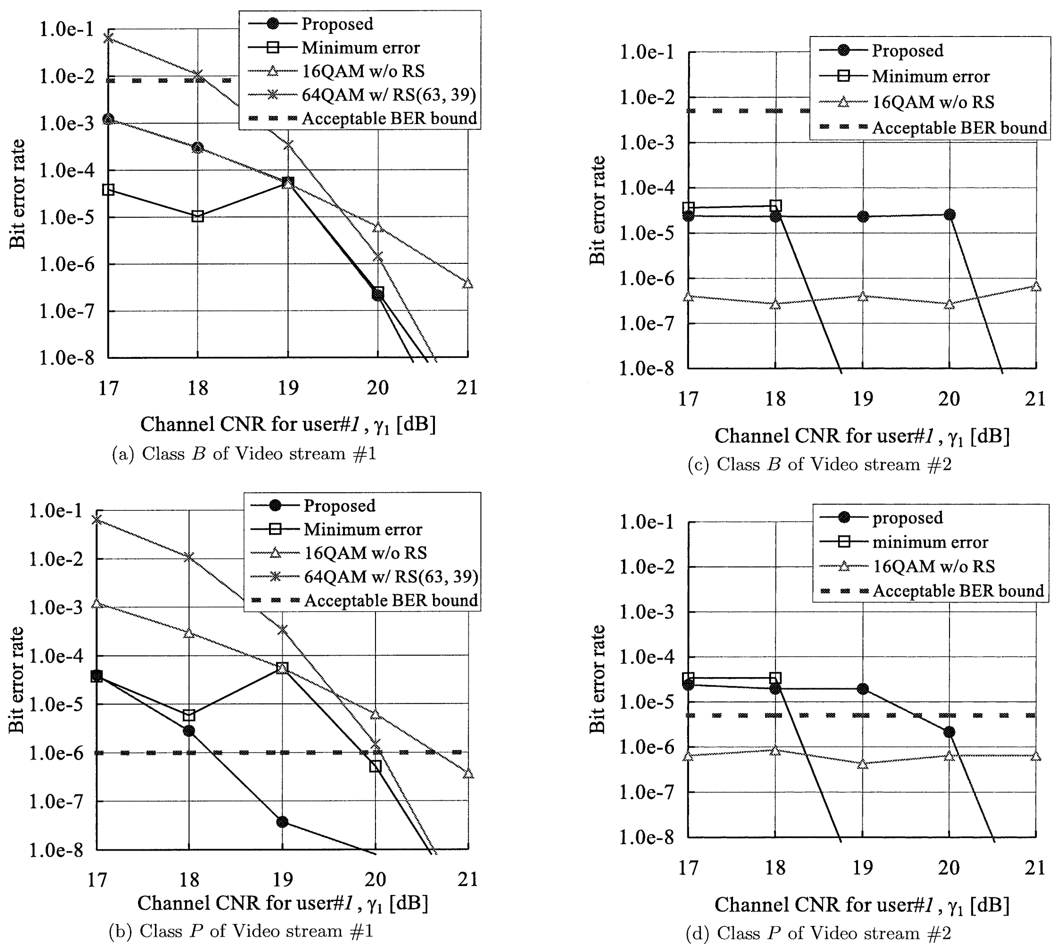
bound for class X is 10^{-3}, $\alpha_X = 1/10^{-3}$. In this way, we estimated the error-sensitivity parameters shown in Table 4.

3.1 BER Performance for Class Picture P/B

Figure 7 shows the bit error rate of class P and B for the user #1 and #2 against the channel CNR for user #1, γ_1. The channel CNR for user #2, γ_2, is 21 [dB]. In this figure, the BER performance of the proposed scheme is shown as well as the algorithm that minimizes mean error after decoding ϵ with α_X=1, "minimum error scheme." The minimum error scheme is different from the proposed scheme in the point of whether the error sensitivity is considered or not. For comparison purpose, as the conventional schemes, "16 QAM w/o RS" and "64 QAM w/ RS (63, 39)" are plotted. 16 QAM and 64 QAM with RS (63, 39) just satisfy the required bit rate of two video streams. In addition, the broken lines indicate the acceptable BER bound. The acceptable BER bound is the bit error rate that gives MOS valued 3. For example, in case of the video source #1, the acceptable BER bound of class P and class B are 1.0×10^{-6} and 8.0×10^{-3}. The class P is more sensitive to channel bit errors than class B. Assuming γ_2 = 21 [dB] regardless of γ_1, 16 QAM gives the constant performance against γ_1 in Figs. 7(c) and (d). (BER of 64 QAM with RS(63, 39) is lower than 10^{-8} in these figures.) From Figs. 7(a) and (b), it is clear that 16 QAM and 64 QAM with RS(63, 39) give poor performances in the low CNR region. Especially, in case of class P, the bit error rates of 16 QAM and 64 QAM with RS(63, 39)

Table 4 Error sensitivity for each class of two video streams.

Error sensitivity	"trees"	"jet"
α_S	$1/(1.0\times10^{-9})$	$1/(1.0\times10^{-9})$
α_G	1.0	1.0
α_H	$1/(1.0\times10^{-2})$	$1/(5.0\times10^{-3})$
α_I	$1/(2.0\times10^{-6})$	$1/(5.0\times10^{-6})$
α_P	$1/(1.0\times10^{-6})$	$1/(5.0\times10^{-6})$
α_B	$1/(8.0\times10^{-3})$	$1/(5.0\times10^{-3})$



(a) Class B of Video stream #1

(b) Class P of Video stream #1

(c) Class B of Video stream #2

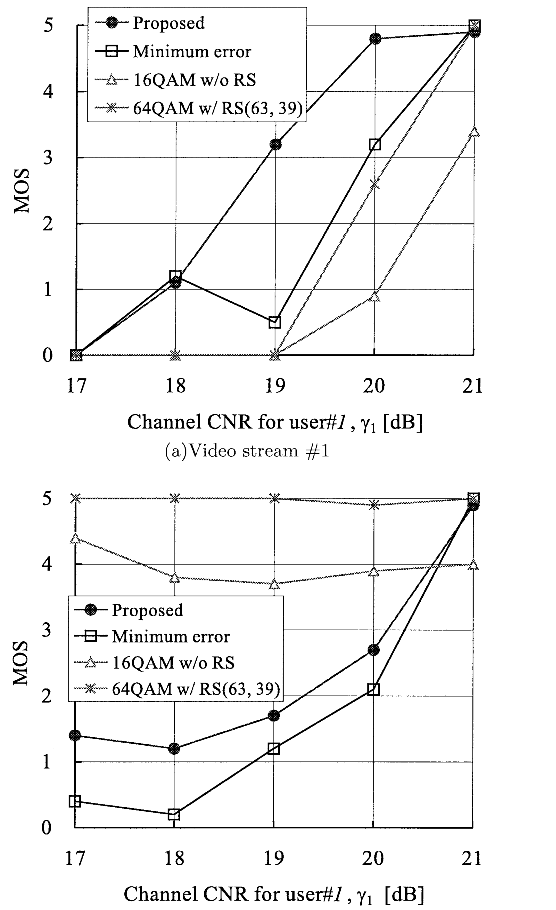
(d) Class P of Video stream #2

Fig. 7 The bit error rate for Class B and P against CNR for user #1, γ_1, assuming channel CNR for user #2, γ_2, = 21 [dB]. The BER performance of proposed scheme is shown as well as "minimum error scheme," which is error minimization algorithm without error-sensitivity estimation. "16 QAM w/o RS" and "64 QAM w/ RS (63, 39)" are plotted. The acceptable BER bound is the bit error rate that gives MOS valued 3.

are much worse than the acceptable BER bound. This error performance would degrade the image quality of video stream #1 seriously in the low CNR region. On the other hand, as shown in Figs. 7(c) and (d), if γ_1 becomes low, although the BER of the proposed scheme and minimum error scheme are degraded, it is close to the acceptable BER bound. This is because, when γ_1 is low, those schemes assign much radio frequency resource to user #1 by reducing the resource for user #2. However, from Figs. 7(a) and (b) we notice that those schemes improve drastically the BER performance at CNR γ_1= 17–18 [dB]. Therefore, we can expect that the degradation of the image quality for the video stream #1 would be improved by them in the low CNR region. Furthermore, from Figs. (c) and (d), the BER performance of the proposed scheme is worse than that of the minimum error scheme at γ_1 = 19–20 [dB]. Since BER of proposed scheme, anyway, is still staying around the acceptable BER bound, it could not lead to the serious degradation in image quality. On the other hand, in case of the video stream #1, since the proposed scheme improve the error performance of class P, which is error sensitive, the video transmission performance for user #1 would be drastically improved.

3.2 MOS Assessment for Video Stream #1/#2

Now, we show the results of the MOS assessment by DSIS method. Figure 8 shows MOS performance of the proposed scheme against the channel CNR for user #1, γ_1. We assume that the channel CNR for user #2, γ_2, is 21 [dB]. For comparison, we also plot the MOS values of the minimum error scheme, which minimizes mean error after decoding without considering error sensitivity, "16 QAM w/o RS" and "64 QAM w/ RS (63, 39)." Figure 8(b) shows the MOS performance of the video stream #2. When the channel CNR, γ_1, is low, the performances of the proposed scheme and minimum error scheme are degraded. This is because these schemes reduce the radio resource of user #2 in order to assign much resource to the user #1, who requests the video through low CNR channel. However, the degradation of proposed scheme is smaller than that of the minimum error scheme since the proposed scheme gives more resource to more important data such as headers and referenced pictures. Figure 8(a) shows the MOS performance of the video stream #1. In case of 16 QAM and 64 QAM with RS (63, 39), the image quality is degraded radically with decreasing CNR, γ_1. In γ_1 = 17–19 [dB], they cannot transmit the video stream #1. On the other hand, the proposed scheme gives much better performance than 16 QAM and 64 QAM with RS (63, 39). This is because the proposed scheme can improve the image quality by assigning much resource to the low CNR user. Furthermore, although the performance of the minimum error scheme is also better than that of 16 QAM and 64 QAM with RS (63,



(a)Video stream #1

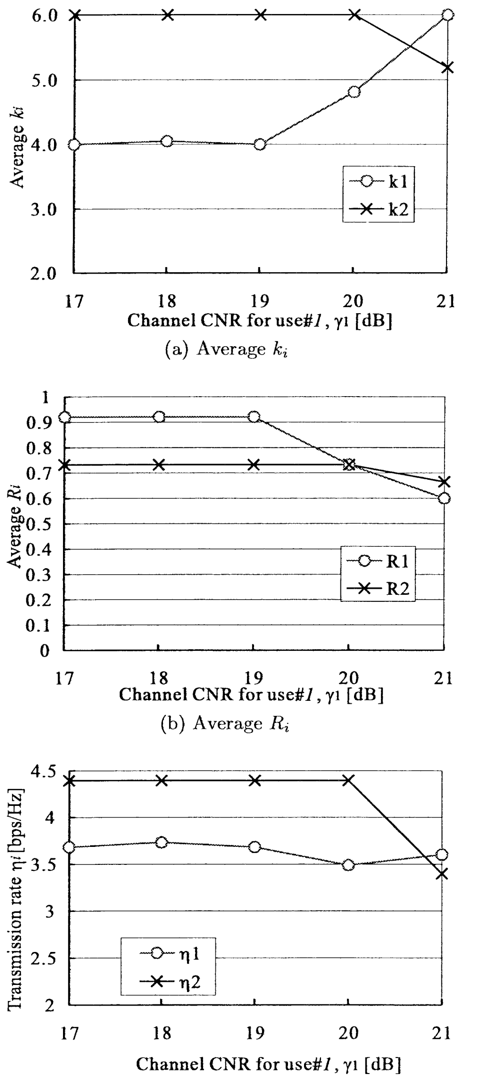
(b)Video stream #2

Fig. 8 MOS assessment results against CNR for user #1, γ_1, assuming channel CNR for user #2, γ_2, = 21 [dB]. The MOS performance of proposed scheme is shown as well as "minimum error scheme," which is error minimization algorithm without error-sensitivity estimation. "16 QAM w/o RS" and "64 QAM w/ RS (63, 39)" are plotted.

39), the proposed scheme improves the image degradation more efficiently than the minimum error scheme by giving resource to error-sensitive data.

3.3 Transmission Formats of Proposed Scheme in Simulation

Figure 9 shows the transmission format of the proposed scheme in the simulation. The channel CNR of user #1 is 17–21 [dB], while that of user #2 is 21 [dB]. The ordinate of Fig. 9(a) indicates average k_i ($i = 1, 2$). k_i is the modulation level assigned to video stream #i in frame by frame basis. The ordinate of Fig. 9(b) is average R_i. R_i is FEC coding rate for video stream #i. From these figures, it is clear that average values of k_i and R_i are changed according to the channel CNR for user #1. Figure 9(c) shows the transmission rate of the transmission format for user #i, η_i, versus channel CNR γ_1. Here, we define η_i [bps/Hz] as,



(a) Average k_i

(b) Average R_i

(c) Transmission rate of transmission format for user #i

Fig. 9 Transmission formats of proposed scheme in simulation. The abscissa indicates channel CNR for user #1. The CNR for user #2, γ_2, = 21 [dB].

$$\eta_i = \text{average}[k_i \times R_i]; (i = 1, 2). \tag{5}$$

From Fig. 9(c), when the channel CNR for user #1, γ_1, is 17–20 [dB], the transmission rate of transmission format for user #1 is lower than that for user #2. Although the transmission format with lower rate is more reliable, it requires more radio frequency resource. In order to assign more radio resource for user #1, the proposed scheme reduces radio resource for user #2, whose CNR is 21 [dB], by employing the bandwidth efficient transmission format. In this way, the proposed scheme gives high reliability to the video stream transmitted through lower CNR channel. On the other hand, when the channel CNR for user #1 and #2 are equally 21 [dB], the transmission rates for both users are almost equal.

4. Conclusion

In this paper, we have proposed a new resource management scheme for multiple video stream transmission in a wireless environment. The proposed scheme determines the transmission parameters, such as modulation format, FEC code rate and the number of time slots per frame, according to the error sensitivities, the required bit rates for the transmitted packets, and the channel state. We have analyzed the bit error rate performance of the proposed scheme by computer simulation. In order to evaluate the system performance, we have assessed the mean opinion score of the transmitted video stream. Computer simulation result and MOS assessment have shown that the proposed scheme gives much better performance than the conventional uniform resource assignment scheme. The MOS of the proposed scheme is even better than that of the "minimum error scheme," which ignores the error sensitivity of the video packet. We, however, have to evaluate the performance of the proposed scheme when the number of video streams is three or more. This gives a limit in our analysis. Although we have assumed that the error sensitivity α is known throughout the paper, there is no clear method to determine it. Further study on a method to estimate the sensitivity without MOS assessment should be required in order to develop a practical system. In addition, many error concealment techniques for MPEG decoding have been studied for many years. We have to investigate the performance in case of applying those techniques to the proposed scheme.

References

[1] M. Nakagawa, "Wireless home link," IEICE Trans. Commun., vol.E82-B, no.12, pp.1893–1896, Dec. 1999.

[2] R. Talluri, "Error resilient video coding in the MPEG-4 standard," IEEE Commun. Mag., vol.26, pp.112–119, June 1998.

[3] G. Côté, S. Shirani, and F. Kossentini, "Optimal mode selection and synchronization for robust video communications over error-prone networks," IEEE J. Sel. Areas Commun., vol.18, no.6, pp.952–965, June 2000.

[4] G. Reyes, A.R. Reibman, S.-F. Chang, and J.C.-I. Chuang, "Error-resilient transcoding for video over wireless channels," IEEE J. Sel. Areas Commun., vol.18, no.6, pp.1063–1074, June 2000.

[5] T. Yamazaki and J. Matsuda, "Adaptive QoS management for multimedia applications in heterogeneous environments: A case study with video QoS mediation," IEICE Trans. Commun., vol.E82-B, no.11, pp.1801–1807, Nov. 1999.

[6] R. Pelz, "An unequal error protected px8 kbit/s video transmission for DECT," Proc. IEEE VTC'94, pp.1020–1024, Sept. 1994.

[7] N. Matoba, Y. Kondo, M. Yamashina, and T. Tanaka, "A study on reducing transmission delay in mobile video communication systems," IEICE Trans. Fundamantals, vol.E80-A, no.7, pp.1281–1287, July 1997.

[8] B. Girod and N. Färber, "Feedback-based error control for mobile video transmission," IEEE J. Sel. Areas Commun.,

vol.16, no.3, pp.358–368, April 1998.
[9] C.H. Wong, P. Cherriman, and L. Hanzo, "Burst-by-burst adaptive wireless video telephony over dispersive channels," Proc. GLOBECOM'99, vol.5, no.9, pp.204–208, Dec. 1999.
[10] "Whitecap2 wireless network protocol white paper," Cirrus Logic white paper, DS555WP1, Sept. 2001.
[11] ITU-R Recommendation 500-10, "Method for the subjective assessment of the quality of television pictures," International Telecommunication Union, March 2000.
[12] J.G. Proakis, Digital Communications, Fourth Edition, McGraw-Hill International Edition, Electrical Engineering Series, pp.274–278, 463–465, 2001.

Appendix: Bit Error Rate for 2^k-QAM with Reed Solomon Code

The symbol error rate of 2^k-QAM is given by

$$P_M = 1 - (1 - P_{\sqrt{M}})^2,$$

where $P_{\sqrt{M}}$ is the symbol error rate of PAM (Pulse Amplitude Modulation) with one-half the average power in each quadrature signal of the equivalent QAM system. By appropriately modifying the probability of error for M-ary PAM, we obtain

$$P_{\sqrt{M}} = \left(1 - \frac{1}{2^{k/2}}\right)\mathrm{erfc}\sqrt{\frac{3}{2^k - 1} \cdot \frac{\gamma}{2}},$$

where γ is the channel CNR. Now assume that RS (N, C) code where $N = 2^k - 1$ is the codeword length and the C is the information symbol per block. Since RS (N, C) code is capable of correcting up to $t = (N-C)/2$ symbol errors, the symbol error probability of 2^k-QAM with Reed Solomon code is given by

$$P_{es} = \frac{1}{N}\sum_{i=t+1}^{N} i \binom{N}{i} P_M^i (1 - P_M)^{N-i}.$$

Furthermore, if the symbols are converted to binary digits, the bit error rate is

$$P_{eb} = \frac{2^{k-1}}{2^k - 1} P_{es}.$$

Although P_{eb} is the worst value of the bit error rate, it is a good approximation of the true value.

Ryoichi Shinkuma was born in Osaka, Japan on October 3, 1976. He received the B.E. and M.E. degrees in Communication Engineering from Osaka University, Osaka, Japan, in 2000 and 2001, respectively. He is currently pursuing the Ph.D. degree at Osaka University. He is engaging in the research on multimedia wireless communication systems.

Minoru Okada received the B.E. degree in communications engineering from the University of Electro-Communications, Tokyo, Japan, in 1990 and the M.E. and Ph.D. degrees both in communications engineering from Osaka University, Osaka, Japan, in 1992 and 1998, respectively. Since 1993, he was with the Department of Communications Engineering, Osaka University, as a Research Associate. From 1999 to 2000, he was with the University of Southampton, U.K., as a Visiting Research Fellow. In 2000, he joined the Graduate School of Information Science, Nara Institute of Science and Technology, Nara, Japan, where he is currently an Associate Professor. Dr. Okada is a member of the Institute of Electrical and Electronics Engineers (IEEE), and the Institue of Image Information and Television Engineers of Japan (ITE). He received the Young Engineer Award from IEICE in 1999.

Shozo Komaki was born in Osaka, Japan, in 1947. He received B.E., M.E. and Ph.D. degrees in Electrical Communication Engineering from Osaka University, in 1970, 1972 and 1983 respectively. In 1972, he joined the NTT Radio Communication Labs., where he was engaged in repeater development for a 20-GHz digital radio system, 16-QAM and 256-QAM systems. From 1990, he moved to Osaka University, Faculty of Engineering, and engaging in the research on radio and optical communication systems. He is currently a Professor of Osaka University. Dr. Komaki is a senior member of IEEE, and a member of the Institute of Television Engineers of Japan (ITE). He was awarded the Paper Award and the Achievement Award of IEICE, Japan in 1977 and 1994 respectively.