# Efficient and Reliable MPEG-4 Multicast MAC Protocol for Wireless Networks

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***Abstract*—Multicasting is the transmission of data to a group of nodes identified by a single destination address. Furthermore, multicasting is considered as an appropriate transmission method for multimedia services. Multimedia applications are expected to become more prevalent over mobile ad hoc networks in the near future. Therefore, achieving reliability in multimedia communications is an important task. Video compression technologies are about reducing and removing redundant video data so that a digital video file can be effectively sent over a network. With modern compression standards, such as Moving Picture Expert Group 2 (MPEG-2), MPEG-4, and H.264, which is also known as advanced video coding MPEG-4 part 10, losses of different frames have different impacts on video quality. In this paper, we propose a leader-based reliable multicast medium-access control layer protocol for multimedia applications to enhance video quality. We present a Markov chain model and numerical formulation of our proposed protocol. Simulation results show that our proposed method is better than other protocols in terms of the number of decodable frames, peak signal-to-noise ratio (PSNR), and video quality.**

***Index Terms*—Multicasting, peak signal-to-noise ratio (PSNR), reliability.**

## I. Introduction

WIRELESS multimedia services are major applications of next-generation wireless networks [1]. Furthermore, to reduce storage space and to transmit video over bandwidth-limited networks, compression of the video bit stream is essential. Video compression technologies such as Moving Picture Expert Group 2 (MPEG-2), MPEG-4, and MPEG-4 advanced video coding (AVC), which is also known as H.264, use motion compensation, where image frames are broken up into blocks on precoded frames. H.264/AVC achieves significantly improved compression efficiency compared with the preceding MPEG-4 and MPEG-2 coding standards [2].

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Multicasting allows a sender to deliver data to a group of multicast members with only one transmission, and as a consequence, this improves bandwidth efficiency [3]. Despite the benefits of multicasting a data packet, multicasting faces several problems. Error control in multicasting is complicated by the fact that a particular packet might be received by some, but lost for others, due to disparate environmental conditions. Receiving an acknowledgement (ACK) frame from everyone to decide whether the multicast data was received successfully or not is a challenge. Multicast receivers cannot simultaneously transmit the ACK frame because too much overhead is required to respond with a packet to the multicast source. Overhead increases as the number of nodes increases. Current standards [4]–[6] state that multicast should be transmitted without requiring any link layer feedback. Therefore, multicast is a transmission mode that does not attempt to guarantee the reception.

A method to attain the reliability in multicasting is proposed by Kuri and Kasera *et al.* [7] for wireless local area networks (WLANs). Gupta and Srimani [8] proposed an adaptive protocol for reliable multicast. Yu and Choi [9] proposed a reliable busy-tone medium access control (MAC) protocol. Gupta *et al.* [10] proposed an extension to the IEEE 802.11 standard MAC, called 802.11MX, to improve the link-level reliability for multicast data. Kim and Kim *et al.* [11] proposed two methods to reduce the number of retransmissions and to decrease the backoff duration. There are few proposals for reliable multicast [7]–[11]. However, the transmission failures of the different types of frames have different impacts on video quality. Moreover, most of the previous proposed methods involving video streaming over WLANs treat video traffic as an aggregated stream. They do not consider the nature of three different frame types and group-of-pictures (GOP) structures defined by Seeling and Reisslein [2].

The main contributions of this paper are summarized as follows. First, we show the impact of different frame losses on video quality. Second, we propose a leader-based reliable multicast MAC layer protocol for multimedia applications and provide the theoretical analysis of the proposed protocol with the help of Markov chain model. Our proposed protocol tries to minimize the frequency of retransmission to prohibit the sender from retransmitting the lost packets of B- and P-frames. Finally, both numerical and simulation results are presented to show the performance of the proposed protocol.

The rest of this paper is organized as follows. Section II reviews the related work. Section III discusses the proposed protocol in detail. Section IV presents the system model and analysis. Section V shows the performance evaluation in detail, and finally, Section VI concludes this paper.

## II. Related Work

To attain reliability and to minimize control overhead in multicasting, Kuri *et al.* [7] proposed a leader-based protocol (LBP) for WLANs. This protocol chooses one of the multicast receivers for the exchange of ready-to-send (RTS), clear-to-send (CTS), and ACK frames. However, LBP does not consider the numerous parameters associated with video compression techniques, such as frame types and frame size. The negative ACK (NACK)-based automatic repeat request (ARQ) mechanism of LBP provides reliable multicast data transmissions with small control frame overhead. As a result, it is widely adopted by various multicast protocols [12], [13]. However, combining the NACK-based ARQ with the aggregated MAC protocol data unit (A-MPDU) results in inefficient retransmissions [10]. Kim and Kim [11] proposed an efficient retransmission method for multicast over contention-based wireless networks. The proposed protocol is composed of two parts, i.e., contention window adjustment and making a decision on retransmission. If any multicast member acknowledges, even if others do not, the contention window is reset to initial value. This is because, if at least one member acknowledges the sender, it means that there might be no collision in one-hop communication. The retransmission decision is based on the target packet delivery ratio. The retransmission is stopped when the packet delivery ratio is higher than the target packet delivery ratio. As a result, the proposed protocol increases the network performance by reducing unnecessary processing time. However, the work is carried out under the assumption that some packet losses of streaming video or audio are tolerated.

Gupta and Srimani [8] proposed an adaptive protocol for reliable multicast in mobile multihop networks. The protocol uses a shared core-based multicast tree and is independent of the underlying unicast routing protocol. However, the protocol did not address the problem of how to perform reliable multicasting in the presence of node failure in mobile ad hoc networks. The protocol also requires each node to acknowledge reception directly back to the source, thus suffering from ACK implosion. Gossain *et al.* [14] proposed multicast-aware MAC protocol (MMP). MMP uses the MAC header to support ACK-based data delivery. After sending the data packet, the transmitter waits for the ACK frame from each destination, known as ACK-based multicast (ABM) protocol. ACK frames from the destination nodes are sent in sequential order to prevent collision among ACK frames at the transmitter. The overhead increases as the number of nodes increases, resulting in throughput degradation. Yu and Choi [9] proposed a reliable busy-tone MAC protocol by using a busy tone that improves data throughput and reliability. To provide reliability, reliable busy-tone MAC uses two busy-tone channels and one control-tone channel. However, throughput and reliability come at the cost of additional transceivers. Lim *et al.* [15] discussed reliable and efficient multicast protocol (REMP) for scalable video streaming. REMP dynamically adjusts the number of transmissions of control frames. In a stable channel condition, access point exchanges control frames only with the selected multicast receivers. In a dynamic channel condition, control frames are exchanged with all multicast receivers, which may increase overhead and reduce overall system performance.

Kim *et al.* [16] suggested orthogonal frequency-division multiple access (OFDMA)-based multicast ACK (OMACK) to allow each receiver to transmit one of two binary phase-shift keying symbols denoting an ACK or NACK frame on a unique subcarrier with OFDM symbols. OMACK is an efficient wireless multicast MAC protocol with small control overhead in multihop wireless ad hoc networks and increases system throughput. However, the higher throughput does not always mean better video quality as proven by Xiao *et al.* [17]. The work of Campolo *et al.* [18] suggests a reliable multicast MAC protocol called reliable adaptive multicast protocol (RAMP) for multihop networks. RAMP ensures a high packet delivery ratio and reduces control overhead. To keep control overhead low, RAMP limits the use of multicast RTS and multicast CTS frames to the first packet of a multicast data flow. There is no handshake for the following packets. The unreliable and error-prone nature of the wireless channel can cause severe degradation in performance due to such handshake processes. Gupta *et al.* [10] proposed an extension to the IEEE 802.11 standard MAC called 802.11MX to improve link-level reliability for multicast data. Because they use a tone-based mechanism to signal the NACK frame, there is no collision of NACK frames. The authors also proposed a dual busy tone to reduce packet collisions due to node mobility. However, higher data throughput and reliability of 802.11MX come at the cost of additional transceivers. Debnath [19] proposed a novel quality-of-service (QoS)-aware MPEG-4 video delivery algorithm and calculated wastage in bandwidth because of the loss of intracoded frames (I-frames). When an I-frame is lost, there is severe impact on that particular GOP, and bandwidth is wasted transmitting the remaining predicted frames (P-frames) and bidirectional frames (B-frames) in a GOP. Therefore, retransmission of the I-frames can increase bandwidth usage. Tang and Rondao Alface [20] developed a Markov chain to describe the error propagation process inside a GOP. Losing a packet in a frame will not only affect the current frame but will also propagate the initial error to subsequent frames due to hierarchical interframe coding. Tang and McKinley [21] showed that packet loss is exacerbated when more receivers start sending feedback packets. Moreover, multicasting feedback causes more data packet loss than unicasting. Their results indicate that the loss density, which directly affects the amount of feedback from receivers, has a significant impact on the performance of reliable multicast protocol in WLANs [21].

## III. Proposed Protocol

### A. Impact of Different Frame Losses on the Quality of Video Stream Transmission

Here, we provide the impact of different frame losses on video quality and the error propagation process in a GOP due to loss of an I-frame. We test the effect of different frame losses on video quality by conducting simulations using video framework Evalvid [22], which is a complete framework and toolset for evaluating the quality of video transmitted over real or simulated communication networks. Simulations are performed in network simulator 2 (NS-2) version 2.35 [23] over multicast-based network environments.

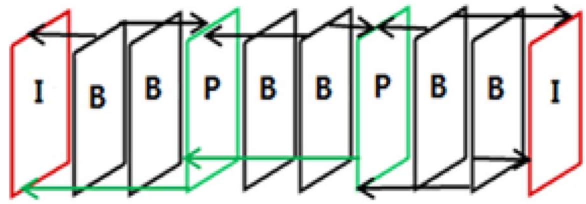



Fig. 1. Sample of a video frame for GOP ($N = 9, M = 3$).

To measure the video quality on multicast receivers, the peak signal-to-noise ratio (PSNR) is calculated with different frame loss rates. PSNR is one of the most widespread objective metrics used to assess application-level QoS of video transmissions. Such objective methods are described by the International Telecommunication Union [24] as

$$\mathrm{PSNR}(n)_{db} = 20 \log_{10} \left\{ \frac{V_{\text{peak}}}{\sqrt{\mathrm{MSE}(n)}} \right\} \tag{1}$$

where $V_{\text{peak}} = 2^k - 1$ is the maximum possible pixel value of the image where $k$ is the number of bits per pixel. For example, when a pixel is represented by 8 bits per sample, $V_{\text{peak}}$ is 255.

Mean square error (MSE) is an estimate of error variance, and the value of MSE is given as

$$\mathrm{MSE}(n) = \frac{\sum_{i=1}^{N_{\text{col}}} \sum_{j=1}^{N_{\text{row}}} \left[Y_s(n,i,j) - Y_D(n,i,j)\right]^2}{N_{\text{col}} N_{\text{row}}} \tag{2}$$

where $N_{\text{col}}$ and $N_{\text{row}}$ are the total number of columns and rows in the input images, $i$ and $j$ are the current column and row positions, $n$ is the current frame number, and $Y_s$ and $Y_D$ are the luminous components of the source and destination image, respectively, as defined by Ke *et al.* [22].

MPEG-4 and H.264 are widely used standards for video compression and contain I-frames, P-frames, and B-frames [2]. The I-frame is used as a reference frame to start a new GOP. A typical GOP order is $IBBPBBPBBPBB$. If an I-frame is lost, all P- and B-frames up to the next I-frame are of no use. However, losses of P- or B-frames have no significant impact on video quality. The GOP structure is often referred to by two numbers, e.g., $M = 3$, $N = 9$. The first number indicates the distance between two anchor frames (I or P); it is the GOP size. The second is the distance between two full images (I-frames); it is the GOP length. The GOP structure for $M = 3$ and $N = 9$ and frame interdependence is shown in Fig. 1. The loss effect of I-, P-, and B-frames on PSNR is shown in Fig. 2. In Fig. 2, frame index 287 (an I-frame) is not decoded since some packets belonging to the I-frame are lost; hence, the following P- and B-frames also show lower PSNR values. The same effect can be also observed for frame index 305 (an I-frame) and 314 (an I-frame).

Fig. 2 also highlights the effect of the loss of P- and B-frames on PSNR. Loss of a packet of P-frame index 329 shows a lower PSNR value; however, there is no propagating effect. On the other hand, with a B-frame index of 250, the PSNR value is acceptable. In summary, as shown in Fig. 2, the loss of an

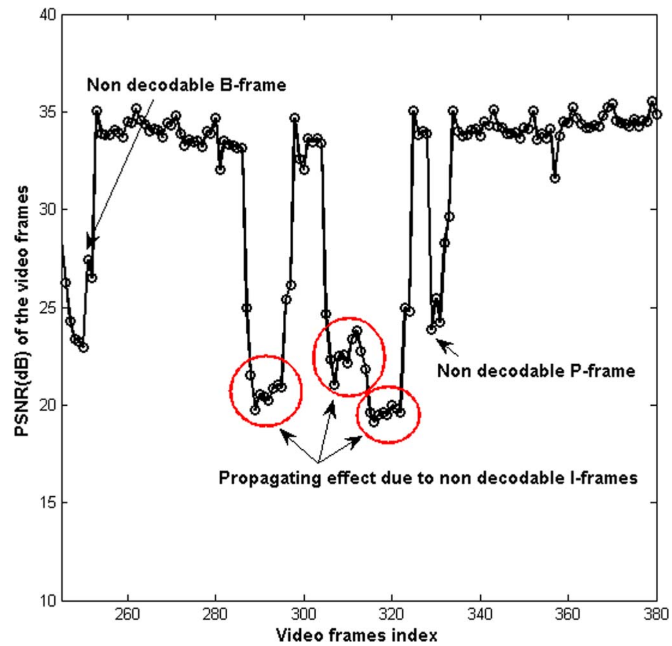



Fig. 2. Impact of I-, P-, and B-frame losses on PSNR.

I-frame has the worst impact on the performance of MPEG-4 transmissions compared to the loss of P- and B-frames. Therefore, the transmission protocol of this paper is designed to provide an efficient way to achieve reliability of I-frames.

### *B. Proposed Protocol Description*

Our proposed protocol, which is called reliable multimedia multicast (RMM), is an extension of LBP for multimedia applications. In LBP, a receiver is selected as a leader for a multicast group. A sender transmits an RTS frame to all receivers. The leader transmits a CTS frame in reply. After receiving the CTS frame, the sender starts transmitting a data frame. The leader sends an ACK frame in reply if the data frame is received successfully; otherwise, it does nothing. If any nonleader receivers detect errors in the received data frame, a NACK frame is sent. If the sender receives an ACK frame, the transmission of the data frame is done. Otherwise, the sender repeats the whole procedure and retransmits the data frame up to the maximum retry limit.

Since wireless channels mostly suffer from low bandwidth and high bit-error rates due to noise, interference, and multipath fading channels, the packet loss rate is high. As a consequence, retransmissions occur frequently. When the traffic load nears or exceeds network capacity, the retransmissions themselves increase network traffic, which increases frame collisions. As a result, a loss of transmitted packets frequently occurs, and transmission delays increase, which also causes dropped packets. Finally, all these situations increase retransmissions.

That is, the aforementioned situations recursively occur, and finally, network performance degrades. In this respect, RMM tries to minimize the frequency of retransmissions. To achieve this, RMM prohibits the sender from retransmitting lost packets of B- and P-frames. However, the sender is allowed to

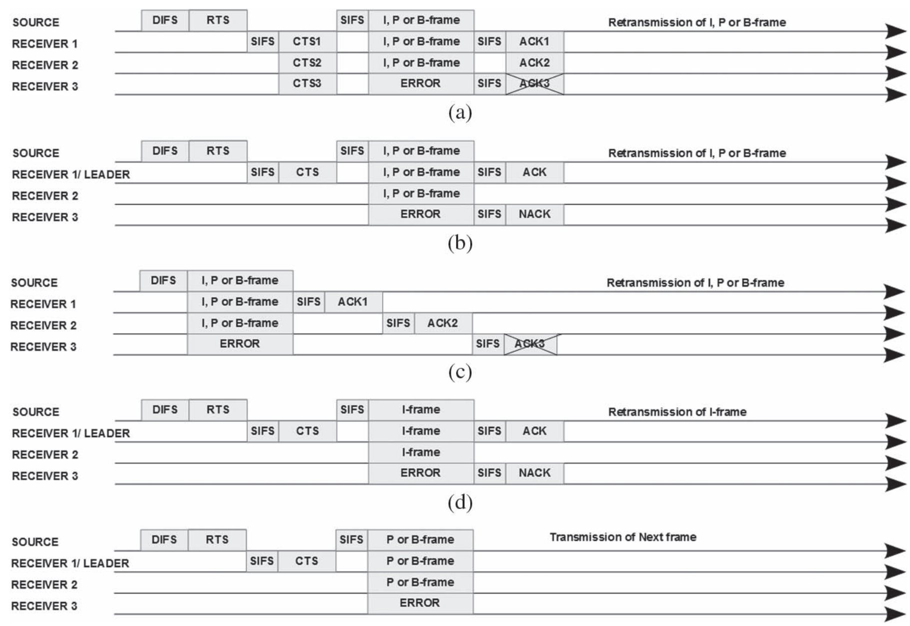



Fig. 3. (a) I-, P-, or B-frame transmission timeline of OMACK. (b) I-, P-, or B-frame transmission timeline of LBP. (c) I-, P-, or B-frame transmission timeline of ABM. (d) I-frame transmission timeline of RMM. (e) P- or B-frame transmission timeline of RMM.

retransmit a lost packet of an I-frame because the loss of an I-frame greatly impacts the received video quality, as shown in Section III-A. An example scenario using the proposed protocol is shown in Fig. 3(d) and (e). The transmission timeline of I-, P-, and B-frames of OMACK, LBP, and ABM is shown in Fig. 3(a)–(c), respectively. If any receiver in the multicast group fails to receive an I-, P-, or B-frame successfully, a NACK frame is transmitted and the sender retransmits the lost frame. In the ABM protocol, ACK frames are sequentially transmitted if the data packet is successfully received, or the NACK frame is transmitted if not.

The detailed RMM algorithm is shown in Algorithm 1.

**Algorithm 1**: RMM MAC Protocol

```
loop
    if node == sender then
        broadcast RTS frame
        if CTS frame is heard then
            start multicasting transmission
        else
            go to loop
        end if
        if ACK frame is not received for I-frame then
            retransmit I-frame
        end if
        if NACK frame is received for I-frame then
            retransmit I-frame
        end if
    end if
    if node == receiver and node == leader node then
        if ready to receive data then
            send CTS frame
        end if
        if frame_type == I and no error then
            send ACK frame
        end if
    end if
    if node == receiver and node != leader node then
        if frame_type == I and error then
            send NACK frame
        end if
    end if
end loop
```

## IV. System Model and Analysis

### A. System Model

Reliable multicast MAC layer protocols can be classified as ABM, OMACK, and LBP. This section compares the performance of the RMM protocol against ABM, OMACK, and LBP. We use the analysis method used by Kim *et al.* [16] and Bianchi and Tinnirello [25]. The system consists of $N$ nodes, including a multicast source and $N-1$ multicast members. We assume that each node always has a packet available for transmission (a saturated condition). The duration of backoff is determined by contention window $(W)$ sizes, which are initially set to $W_{\min}$. The $W$ value is used to randomly select the number of slot times $\sigma$ in the range $[0, W-1]$, which is used for the backoff duration. With an unsuccessful transmission, the $W$ value is updated to $2W$ as long as it does not exceed $W_{\max}$. Let us adopt the notation $W_i = 2W_{i-1}$, where $i \in \{1, \ldots, m\}$ is the backoff stage, and $m$ is the maximum backoff stage, such that $W_{\max} = 2^m W_{\min}$.

### *B. Transmission and Failure Probability*

A discrete and integer timescale is adopted: $t$ and $t+1$ correspond to the beginnings of two consecutive changes in the backoff time counter. We refer to the time interval between $t$ and $t+1$ as the counter time slot. The counter time slot is of variable time duration, whereas the slot time is a constant time duration period. Because the decrement of the backoff time counter stops when the channel is busy, the time interval between the beginning of two consecutive backoff time counter instants may be much longer than the constant slot time duration. Let us denote as $X$ an event wherein a node transmits a packet into a counter time slot. We focus on transmission probability $\tau = \Pr(\mathbf{X})$ that a node transmits a packet into a counter time slot. $p_c$ is the probability that the transmitted packet sees a collision on the channel. Channel conditions such as shadowing and fading are assumed to generate a constant packet loss probability $p_e$ for all wireless connections. When $p_e = 0$, channel conditions are ideal. Let $p_{f,A}$, $p_{f,L}$, $p_{f,0}$, $p_{f,R,I}$, $p_{f,R,P}$, and $p_{f,R,B}$ be the failure probabilities of ABM, LBP, OMACK, I, P, and B packets of the RMM protocol, respectively. While failure of data transmission generally happens when the transmitter does not receive ACK for the transmitted data packet because of a collision or another channel condition, failure of each protocol can be differently defined as follows. In ABM, LBP, and OMACK-based protocols, a transmitter receives ACK frame for all types of packets. However, in the RMM protocol, the transmitter receives ACK frame only for packets belonging to an I-frame from the leader node, and NACK is transmitted if any receiver node fails to receive a packet belonging to I-frames. In LBP, a transmitter confirms ACK frame only from the leader node. However, NACK frame is transmitted if any receiver node fails to successfully receive a packet. In ABM, a transmitter confirms ACK frame from all the member nodes. Therefore, failure probabilities can be written as

$$p_{f,A} = p_{f,L} = p_c + (1-p_c)\left[1-(1-p_e)^{N-1}\right] \quad (3)$$

$$p_{f,R,I} = p_{p,I}\left[p_c + (1-p_c)\left[1-(1-p_e)^{N-1}\right]\right] \quad (4)$$

$$p_{f,R,P} = p_{p,P}\left[p_c + (1-p_c)\left[1-(1-p_e)^{N-1}\right]\right] \quad (5)$$

$$p_{f,R,B} = p_{p,B}\left[p_c + (1-p_c)\left[1-(1-p_e)^{N-1}\right]\right] \quad (6)$$

where $p_{p,I}$, $p_{p,P}$, and $p_{p,B}$ are the probabilities that a packet belongs to an I-frame, a P-frame, and a B-frame, respectively.

The failure probability of the OMACK-based ACK protocol can be written [16] as

$$p_{f,O} = p_c + (1-p_c)\left[1-\sum_{i=0}^{m}(1-p_e)^{E[r_i]}Pr(b=i)\right] \quad (7)$$

where $\Pr(b=i)$ is the probability that a transmitter is found in the backoff stage $i$, and $r_i$ is the number of receivers that do not return ACK frame until backoff stage $i$ is reached. $r_i$ is a binomial random variable, and its mean value is given as

$$\begin{aligned} E[r_0] &= N-1 \\ E[r_1] &= E[r_0]p_{f,O} = (N-1)p_{f,O} \\ E[r_{i+1}] &= E[r_i]p_{f,O} = (N-1)(p_{f,O})^i, \text{ for } i=0,\ldots,m-1. \end{aligned} \quad (8)$$

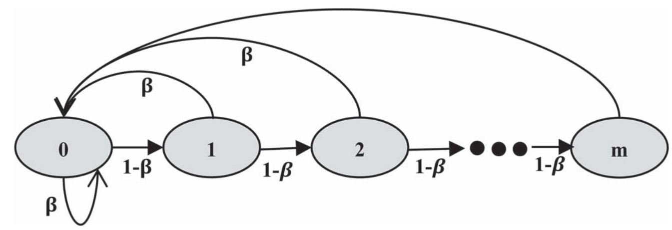



Fig. 4. Markov chain model for the backoff stage.

$p_c$ is assumed to be a constant value, independent of the number of retransmissions that have occurred. Note that $p_c$ is the probability that, in a time slot, at least one of the $N-1$ remaining nodes transmits. In a steady state, each $N-1$ remaining node transmits a packet with probability $\tau$ for each protocol, and $p_c$ is equal to

$$p_c = 1-(1-\tau)^{N-1}. \quad (9)$$

The probability that a node is found in the backoff stage $i$ is given as

$$\Pr(b=i) = \tau\frac{\Pr(b=i|X)}{\Pr(X|b=i)}, \quad i \in (0,\ldots,m). \quad (10)$$

By summing all the values of $i$, we get

$$\sum_{i=0}^{m}\Pr(b=i) = \tau\sum_{i=0}^{m}\frac{\Pr(b=i|X)}{\Pr(X|b=i)} \quad (11)$$

where $\tau$ can be calculated as

$$\tau = \frac{1}{\sum_{i=0}^{m}\frac{\Pr(b=i|X)}{\Pr(X|b=i)}}. \quad (12)$$

A Markov chain model for the backoff stage is shown in Fig. 4, where $\beta$ represents the successful transmission, and $1-\beta$ represents the failed transmission, respectively. The transition probabilities of the backoff stage are given as

$$\Pr\{b(t+1)=i|b(t)=i-1\} = p_f,\ i=1,\ldots,m \quad (13)$$

$$\Pr\{b(t+1)=0|b(t)=i\} = 1-p_f,\ i=1,\ldots,m-1 \quad (14)$$

$$\Pr\{b(t+1)=0|b(t)=m\} = 1. \quad (15)$$

It readily follows that the conditional backoff stage probability $\Pr(b=i|X)$ is a geometric distribution, i.e.,

$$\Pr(b=i|X) = \frac{(1-p_f)p_f^i}{\left(1-p_f^{m+1}\right)}, \quad i \in (0,\ldots,m) \quad (16)$$

where $p_f^i$ is the failure probability of ABM, LBP, OMACK, and RMM at backoff stage $i$. From the independence between transmission cycle and renewal theory, we obtain the conditional transmission probability $\Pr(X|b=i)$ by dividing the average number of counter time slots required in a transmission cycle (exactly one time slot) by the average number of counter time slots required by the node during the complete cycle (i.e., backoff and transmission cycle in backoff stage $i$). Because a time slot corresponds to a backoff counter decrement, the conditional transmission probability is

$$\Pr(X|b=i) = \frac{1}{1+E[c_i]}, \quad i \in (0,\ldots,m) \tag{17}$$

where $E[c_i]$ is the average value of the backoff counter extracted by a node entering stage $i$. $E[c_i]$ is equal to $W_i/2$ under the assumption of a uniform distribution in the range of $(0, W_i - 1)$. Using (12), (16), and (17), we get

$$\tau = \frac{1}{1+\frac{1-p_f}{1-p_f^{m+1}}\sum_{i=0}^{m} p_f^i E[c_i]}. \tag{18}$$

The probability $\Pr(b=i)$ can be expressed as

$$\Pr(b=i) = \tau \frac{(1-p_f)p_f^i}{1-p_f^{m+1}}\left(1+E[c_i]\right). \tag{19}$$

Note that (9) and (18) represent a nonlinear system with two unknowns $\tau$ and $p_f$, which can be solved by using numerical techniques.

### C. Packet Drop Probability

Let $p_{d,I}$ represent the probability that a packet of an I-frame is dropped. If we assume that all multicast receivers have the same channel condition $p_e$, then the average packet drop probability of the I-frame of all multicast receivers can be written as

$$p_{d,I} = \sum_{i=0}^{m} \Pr(\text{drop}|b=i)\Pr(b=i). \tag{20}$$

Packet drop probabilities of P and B packets are equal to the failure probabilities because there is no retransmission for P and B packets with the RMM method. Failures of P and B packets are considered dropped and are represented as $p_{d,P}$ and $p_{d,B}$, respectively.

For other multicast protocols than RMM, a packet in the backoff stage $i$ will be dropped if it reaches the maximum backoff stage $m$ (i.e., it collides for $m-i$ times) and if it collides during the last transmission attempt. In OMACK-based and ABM protocols, a data packet is dropped because of retry limit exhaustion. The backoff stage is updated if any receiver does not return ACK frame. Hence, $\Pr(\text{drop}|b=i)$ for ABM, LBP, OMACK, and ABM are given as

$$\Pr(\text{drop}|b=i) = p_f^{m+1-i}. \tag{21}$$

### D. Number of Decodable Frames

The number of decodable frames is metric, which has been used to evaluate the quality of the video stream, and has been used in previous work [26]. Here, we present the analytical estimation of the number of decodable frames. In GOP, an I-frame is successfully decodable only if all the packets that belong to the specific I-frame are received successfully. P-frames are successfully decodable only if the preceding I-frame and P-frames are decodable and all the packets that belong to the tagged P-frames have been successfully received. B-frames are decodable only if the preceding and succeeding I-frame and P-frames are all decodable and all the packets that comprise the tagged B-frame have been successfully received. Let $R_I$, $R_P$, and $R_B$ be the expected numbers of successfully decodable I-, P-, and B-frames per GOP, respectively [26], and we get

$$R_I = (1-p_{d,I})^{C_I} \tag{22}$$

$$R_P = (1-p_{d,I})^{C_I}\sum_{i=1}^{N_P}(1-p_{d,P})^{iC_P} \tag{23}$$

$$R_B = \left[(1-p_{d,I})^{C_I}(1-p_{d,P})^{N_P C_P} + \sum_{i=1}^{N_P}(1-p_{d,P})^{iC_P}\right] \cdot (M-1)(1-p_{d,I})^{C_I}(1-p_{d,B})^{C_B} \tag{24}$$

where $C_I$ is the average number of packets in one I-frame, $N_P$ is the number of P-frames in one GOP, $C_P$ is the average number of packets in one P-frame, $M$ is the distance between an I-frame and a P-frame in a GOP, and $C_B$is the average number of packets in one B-frame. Therefore, utilizing the drop probabilities $p_d$ in the previous section, the respective number of successfully decodable frames can be analytically estimated.

## V. Performance Evaluation

Failure probability $p_f$ increases as the number of nodes increases because collision probability increases when we increase the number of nodes within transmission range. On the other hand, transmission probability $\tau$ decreases as we increase the number of nodes because sender nodes experience longer backoff duration. The failure probability of RMM is less than that of ABM, LBP, and OMACK. This is because the RMM protocol only receives the ACK frame for packets belonging to I-frames, and there is no ACK frame for packets belonging to P- and B-frames. Therefore, although transmissions of some of P- and B-frames are failed, all transmissions of P- and B-frames are considered successful at the transmitter because no ACK is expected. However, with ABM and OMACK protocols, transmission is successful only after receipt of ACK frames for all packets belonging to I-, P-, and B-frames from all receivers.

In contrast to ABM and LBP, the OMACK protocol shows less failure probability because each multicast member node has a unique pre-assigned subcarrier in OFDMA.

The average packet drop probability is shown in Fig. 5. In ABM, LBP, OMACK, and RMM protocols, packets are dropped because of retry limit exhaustion. Drop probability increases as the number of nodes increases because failure probability increases. Drop probability of the ABM protocol is higher than that of other protocols because each receiver transmits an ACK frame for received all packets belonging to I-, P-, and B-frames in every transmission. That is, requiring feedback packet from all member receivers increases network

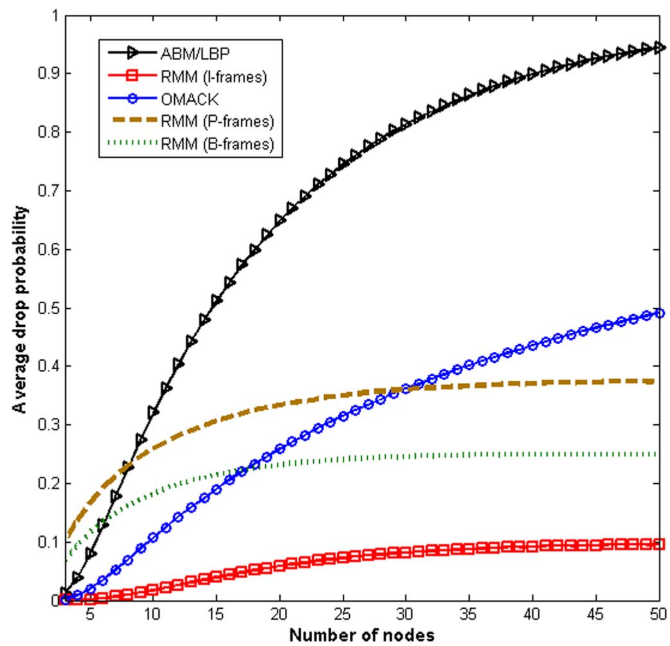



Fig. 5. Average drop probability as a function of the number of nodes.

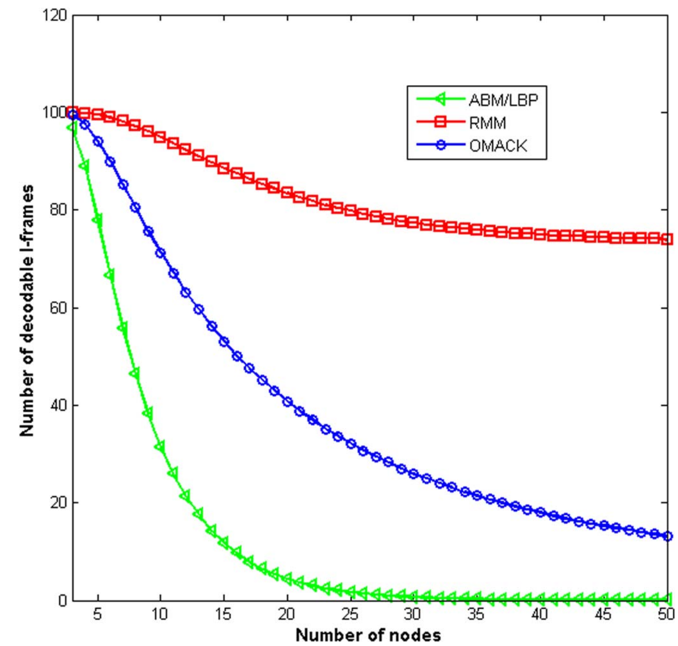



Fig. 6. Number of decodable I-frames as a function of the number of nodes.

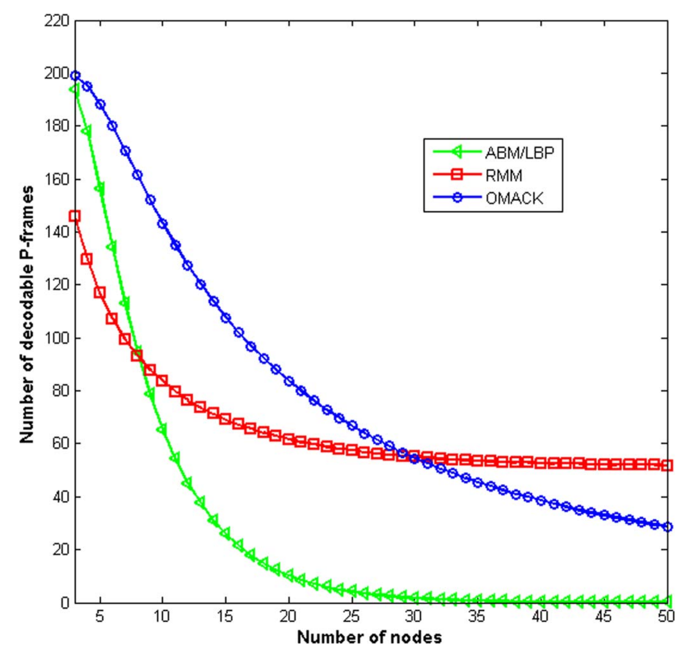



Fig. 7. Number of decodable P-frames as a function of the number of nodes.

overheads, and as a consequence, it increases drop probability. On the other hand, in the RMM protocol, only the leader node is required to transmit the ACK frame for received packets belonging to I-frames so that it minimizes feedback traffic and drop probability. Fig. 5 also shows the drop probabilities of P- and B-frames. The drop probabilities of P- and B-frames are higher than that of I-frames because there is no retransmission for P- and B-frames, and failures for P- and B-frames are considered drops.

H.264 and MPEG-4 video formats are characterized by interframe dependence. If some frames are not successfully received, this will lead to dropping of other successfully received frames because of their interdependence. This leads to a waste of bandwidth, as described by Debnath [19]. Fig. 6 shows the number of decodable I-frames in ABM, LBP, OMACK, and RMM protocols as a function of the number of nodes. The number of decodable I-frames in the RMM protocol is higher than ABM, LBP, and OMACK protocols because RMM protocol has a lower drop probability than ABM, LBP, and OMACK protocols. Fig. 7 shows the number of decodable P-frames. When the number of nodes is less than 30, the number of decodable P-frames is less in the RMM protocol than in the OMACK protocol because there is no retransmission of packets belonging to P-frames. However, when the number of nodes is higher than 30, the number of decodable P-frames in the RMM protocol is higher than OMACK. This is because the effect of successful reception of I-frames overcomes the effect of reception failure of P-frames. If the receiver fails to receive an I-frame successfully, the receiver cannot decode the P-frame properly. Therefore, successful delivery of I-frames also helps increase the number of decodable P-frames. Fig. 8 shows the number of decodable B-frames. B-frames depend on the previous and the next I- or P-frames to decode it. Therefore, successful delivery of I- and P-frames also helps increase the number of decodable B-frames in the RMM protocol.

The transmission of a video sequence is simulated using NS-2 with video framework Evalvid [22]. We considered the *Carphone* and *Grandma* sequences in the quarter common intermediate format with a frame rate of 30 frames/s and GOP pattern $IBBPBBPBB$. In the simulation, the source reads the compressed video file from the video encoder, then fragments the large video frames into smaller segments, and then transmits over simulated wireless networks. The maximum transmitting packet size is 1000 bytes. The IEEE 802.11 protocol [4] is used for the MAC layer protocol with the lowest data rate, i.e., 6 Mb/s, and a communication area of 300 m by 300 m.

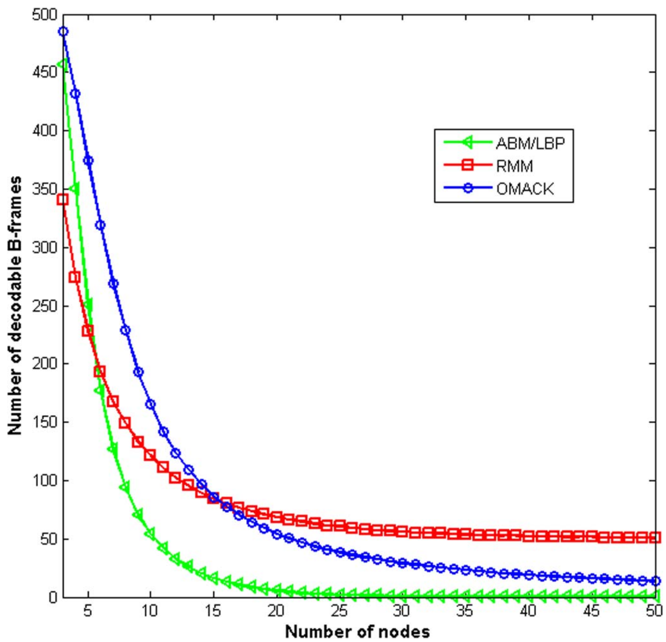



Fig. 8. Number of decodable B-frames as a function of the number of nodes.

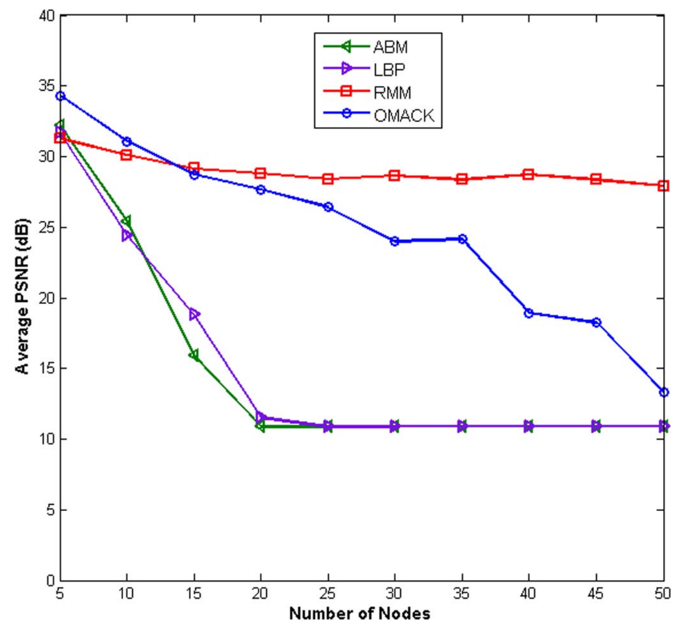



Fig. 9. Average PSNRs of the video sequence.

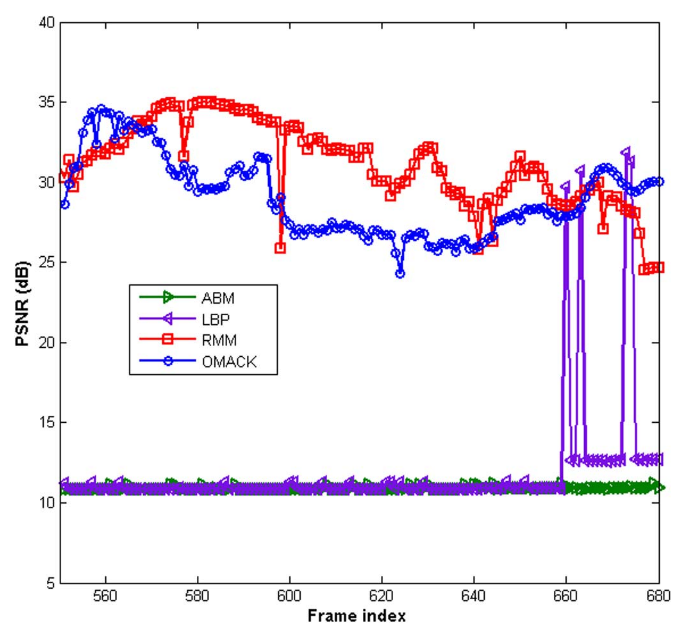



Fig. 10. PSNRs of the video sequence (when the number of nodes is 20).

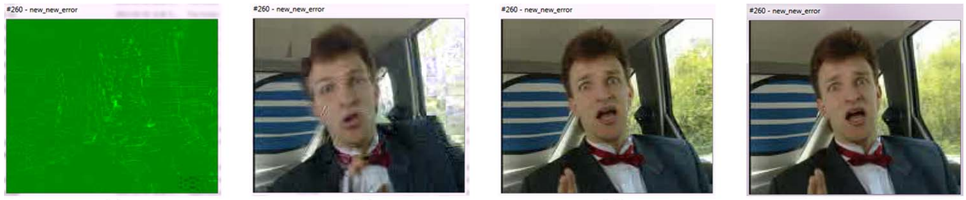
TABLE I
PSNR TO MOS CONVERSION TABLE

| PSNR (dB) | MOS |
|---|---|
| > 37 | 5 ( Excellent) |
| 31 ~ 37 | 4 (Good) |
| 25 ~ 31 | 3 (Fair) |
| 20 ~ 25 | 2 (poor) |
| < 20 | 1(bad) |

As explained in Section III, PSNR is one of the most widespread objective metrics to assess the application-level QoS of video transmissions. The other measure is known as subjective quality metrics. This metric of the human quality impression is usually given on a scale that ranges from 1 (worst) to 5 (best) and is also known as mean opinion score, as shown in Table I [22], [27]. The average PSNRs with different protocols are compared in Fig. 9, which shows the effect of packet drop probability on the average PSNR of the video sequence. When we increase the number the nodes, the average PSNR of the video sequence decreases in all cases because the drop probability increases, as shown in Fig. 5. The average PSNR of the OMACK protocol is better than RMM up to 15 nodes. However, the average PSNR of the RMM protocol is higher than the ABM, LBP, and OMACK protocols when we increase the number of nodes to 15 nodes because of the lower drop probability of the RMM protocol. In the RMM protocol, reduced retransmission decreases feedback traffic and increases PSNR. Since the feedback shares the communication channel with the forward data traffic, the intensity of the feedback affects the reliability of the multicast protocol.

However, with the OMACK protocol, the average PSNR is acceptable at up to 25 nodes. With ABM and LBP protocols, the drop probability is high, which reduces the average PSNR. This is further supported by the PSNR plot-versus-frame index shown in Fig. 10.

Figs. 11 and 12 show the effect of packet loss on the video quality of the video sequences *Carphone* and *Grandma*, respectively. Simulations are performed when the number of nodes is 20. The video quality of the RMM protocol is better than ABM, LBP, and OMACK because packet loss can cause the decoder to function incorrectly with some or all the information after the occurrence of the error, and this means that part or all of the decoded video will be distorted or completely lost, as shown in Figs. 11 and 12. Packet loss can cause a video decoder to lose synchronization within the sequence. Successful reception of the I-frames helps the decoder to resynchronize the entire scene, minimize error propagation within the GOP, and increase the average PSNR of the video sequence.



(a) (b) (c) (d)

Fig. 11. Frame index 260 received via (a) ABM, (b) LBP, (c) OMACK, and (d) RMM protocols.

(a) (b) (c) (d)

Fig. 12. Frame index 538 received via (a) ABM, (b) LBP, (c) OMACK, and (d) RMM protocols.

## VI. Conclusion

Multimedia applications around the world cause a significant global increase in network traffic. Multicasting is an efficient method compared with unicasting when supporting multimedia applications. Wireless channels mostly suffer from low bandwidth and high bit-error rates due to noise, interference, and multipath fading channels. Thus, the packet loss rate is high, and reliability is important in multimedia multicasting. Gathering acknowledgement from multiple receivers increases overhead. Overhead increases as the number of nodes increases. Therefore, in this paper, we have proposed a reliable multicast MAC layer protocol for multimedia applications. The proposed protocol only retransmits the important I-frames, and there is no retransmission of P- and B-frames in order to minimize the number of retransmissions. Results show that the proposed protocol can enhance video quality by increasing the number of decodable I-frames.

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