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A Leader-based Reliable Multicast MAC Protocol for Multimedia Applications

Muhammad Khalil Afzal[1], Byung-Seo Kim[2] and Sung Won Kim[1]
[1] Department of Information and Communication Engineering, Yeungnam University, Korea
[e-mail: khalil_78_pk@yahoo.com, swon@yu.ac.kr]
[2]Department of Computer and Information Communication Engineering Hongik University, Korea
[e-mail: jsnb@hongik.ac.kr]

*Corresponding author: Sung Won Kim

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Abstract

Multicasting is an efficient way of group communications because one sender can transmit data to multiple receivers with only one transmission. Furthermore, multicasting is considered 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. 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 system.

Keywords: Multicasting, Multimedia, Reliability, Retransmission

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1. Introduction

In recent years, the demands for video multicast has been increasing rapidly with advances in multimedia technologies. Wireless multimedia multicast conserves bandwidth by streaming a video to multiple nodes [1][2]. Compared to Moving Picture Expert Group 2 (MPEG-2) video compression technology MPEG-4 and H.264 achieves improved compression efficiency [3].

The IEEE 802.11a/b/g/n standards do not provide reliable multicast. Recently, IEEE 802.11aa [4] Task Group has addressed this limitation and targets at significantly improving both the effectiveness in terms of reliability and the efficiency of multicat traffic. Kuri and Kasera [5] proposed a leader-based protocol (LBP) to improve the reliability of multicast traffic for wireless local area network (LAN). This protocol chooses one of the multicast receivers for the exchange of ready-to-send (RTS), clear-to-send (CTS), and acknowledgement (ACK) frames. However, 802.11aa and LBP do 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 to various multicast protocols [6][7]. However, combining the NACK-based ARQ with the aggregated medium access control (MAC) protocol data unit (A-MPDU) results in inefficient retransmissions [8]. Yu and Choi [9] proposed a reliable busy-tone MAC (RBMAC) protocol by using a busy-tone that improves data throughput and reliability. To provide reliability, RBMAC uses two busy-tone channels and one control-tone channel. However, throughput and reliability come at the cost of additional transceivers. Lim et al. [8], 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. In [10], authors proposed an efficient wireless multicast MAC protocol with small control overhead in multi-hop wireless ad hoc networks and increases the system throughput. However, the higher throughput does not always mean better video quality as proven by Xiao et al. [11]. The work in [12] proposed a reliable multicast MAC protocol called reliable adaptive multicast protocol (RAMP) for multi-hop 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. Authors in [13] 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 the NACK frame, there is no collision in NACK frames. Authors further 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. Yigal et al. [14] proposed an adaptive multicast services system for providing scalable and efficient delivery of multicast system with low communication overhead. However, in contrast to their work, our focus is multimedia applications in multicast networks and we also consider a detailed frame structure of MPEG-4 and H.264 and provide a Markov chain-based analytical model. We also show the impact of our proposed protocol on video quality.

There are a few proposals for reliable multicast [8]-[10]. The transmission failures of the different types of frames have different impacts on video quality. However, most of the previous proposed methods involving video streaming over wireless LANs treat video traffic as an aggregated stream. They do not consider the nature of three different frame types and group of picture (GOP) structure defined by Seeling and Reisslein [3]. In this paper, we propose a leader-based reliable multicast MAC layer protocol for multimedia application and provide a Markov chain modeling for the backoff stage of our proposed protocol by considering the different types of frames of MPEG-4 and H.264 compression technologies.

The rest of the paper is organized as follows. Section 2 discusses the proposed protocol in detail. Section 3 presents the performance analysis. Section 4 demonstrates the numerical results, and finally, Section 5 concludes the paper.

2. Proposed Protocol

2.1 Impact of Different Frame Losses on Video Quality

To emphasize the problem statement, we test the effect of different frame losses on video quality by conducting simulations using video framework Evalvid [15]. Evalvid is a complete framework and tool-set for evaluating the quality of video transmitted over real or simulated communication networks. Simulations are performed in network simulator 2 (NS2) version 2.35 [16] over multicast-based network environments. To measure the video quality on multicast receiver, 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 the application-level quality of service (QoS) of video transmissions. Such objective methods are described by International Telecommunication Union [17] as

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

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

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

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

where N_{col} and N_{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, Y_s and Y_D are the luminous component of the source and destination image, respectively, as defined in [15].

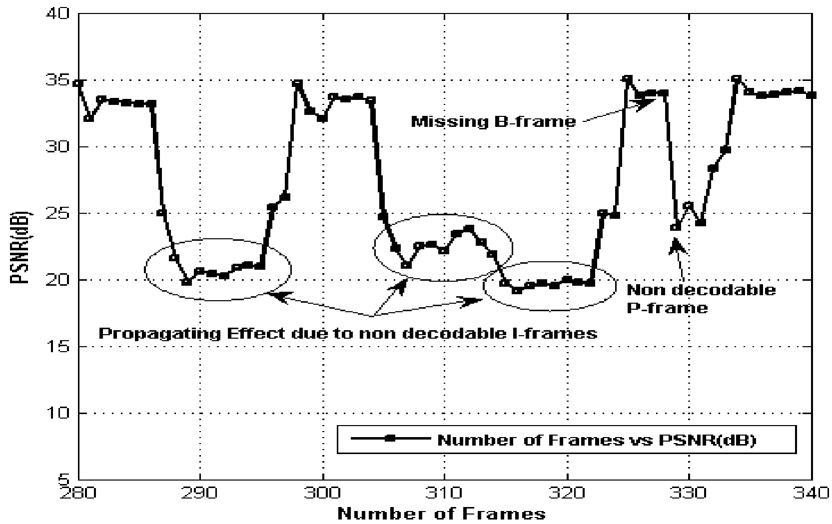


Fig. 1. Loss effect of I, P and B-frames on PSNR.

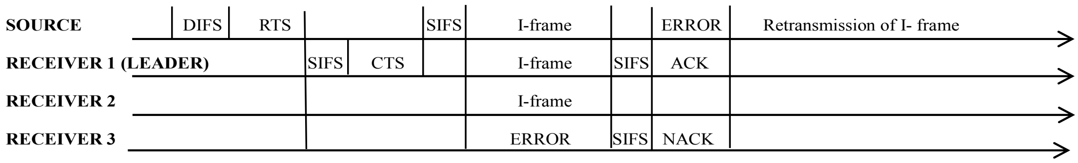
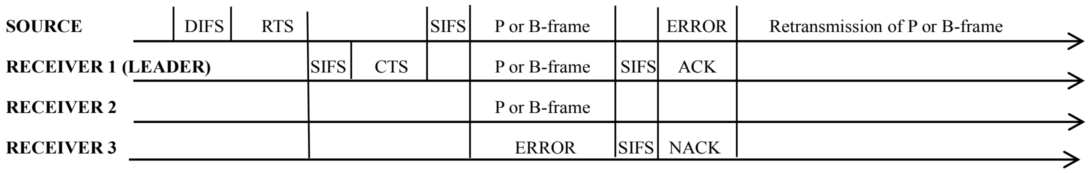
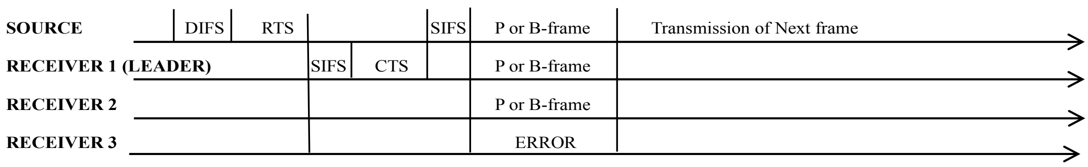


Fig. 2. I-frame transmission time line of LBP and RMM protocol.



(a)



(b)

Fig. 3. **(a)** P or B-frame transmission time line of LBP. **(b)** P or B-frame transmission time line of RMM Protocol.

MPEG-4 and H.264 are widely used standards for video compression and contain Intra-coded frames (I-frames), Predicted frames (P-frames) and Bidirectional frames (B-frames) [3]. The I-frame is used as a reference frame to start the 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 and B-frames have no significant impact on video quality. The loss effect of I, P and B-frames on PSNR can be seen in **Fig. 1**. In **Fig. 1**, frame number 287 (an I-frame) is not decoded since some of the packets belonging to I-frame are lost, so the following P and B-frames also show lower PSNR values. The same effect can also be observed for frame number 305 (an I-frame) and 314 (an I-frame). **Fig. 1** also highlights the effect of the loss of P and B- frames on PSNR. Loss of a packet of P-frame number 329 shows a lower PSNR value, however there is no propagating effect. On the other hand, with B-frame number 328, the PSNR value is acceptable. In summary, as shown in **Fig. 1**, the loss of an I-frame gives 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.

2.2. Proposed Protocol Description

Our proposed protocol, named Reliable Multimedia Multicast (RMM), is an extension of LBP for multimedia applications. In LBP, a receiver is selected as the 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 non-leader 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 as shown **in Fig. 2** and **Fig. 3(a)**.

Since wireless channel mostly suffers 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 is near or exceeds the network capacity, the retransmissions themselves increase network traffic, which increases frame collisions. As a result, loss of transmitted packets occur frequently, and the transmission delays increase, which also causes packet drops. Finally, all these situations increase retransmissions; that is, the aforementioned situations occur recursively, and finally, network performance degrade. In this perspective, 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 retransmit a lost packet of an I-frame because the loss an of I-frame greatly impacts the received video quality as shown in Section 2.1. An example scenario using the proposed protocol is shown in **Fig. 3(b)**.

3. Performance Analysis

3.1 System Model

This section compares the performance of the RMM protocol against LBP. We use the analysis method used in [10][18]. The system consists of N nodes, including a multicast source and N - 1 multicast members. RMM protocol is for single-hop network and it is assumed that source and multicast receivers are always within the communication range of each other. We consider that every node has always a packet available for transmission (saturated conditions). The duration of backoff is determined by the contention window (W) sizes, which are initially set to W_{min}. The W value is used to randomly select the number of slot times (σ) in the range $[0, \mathrm{W}-1]$, which is used for the backoff duration. In the case of 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 $i \in \{1, \ldots, B\}$ is the backoff stage and B is the maximum backoff stage such that $W_{max} = 2^B W_{min}$.

3.2 Transmission and Failure Probability

A discrete and integer time scale 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. 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 the event wherein a node transmits a packet into a counter time slot as X. We focus on transmission probability $\tau = \Pr(X)$, that a node transmits a packet into a counter time slot. $p_{collision}$ 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 of wireless connections. When $p_e = 0$, channel conditions are ideal. Let $p_{failure}^{LBP}$, $p_{failure,I}^{RMM}$, $p_{failure,P}^{RMM}$, and $p_{failure,B}^{RMM}$ be the failure probabilities of LBP, I, P and B packets of RMM protocol, respectively. Failure happens when the transmitter does not receive ACK frame for the transmitted data packet because of collision or channel condition. Let $p_{success}^{LBP}$, and $p_{success,I}^{RMM}$ be the success probabilities of LBP and I packet of RMM protocol, respectively. In LBP, a transmitter receives ACK frame from a leader for all types of packets. However, in the RMM protocol, the transmitter receives ACK frame only for packets belonging to an I-frame. Failure probabilities can be written as:

$$p_{failure}^{LBP} = p_{collision} + p_e = 1 - p_{success}^{LBP}, \tag{3}$$

$$p_{failure,I}^{RMM} = p_{packet,I}\,(p_{collision} + p_e) = 1 - p_{success,I}^{RMM}, \tag{4}$$

$$p_{failure,P}^{RMM} = p_{packet,P}\,(p_{collision} + p_e), \tag{5}$$

$$p_{failure,B}^{RMM} = p_{packet,B}\,(p_{collision} + p_e), \tag{6}$$

where $p_{packet,I}$, $p_{packet,P}$, and $p_{packet,B}$ are the probabilities that a packet belongs to an I-frame, a P-frame and a B-frame, respectively. $p_{collision}$ is assumed to be a constant value, independent of the number of retransmission that have occurred. It is sufficient to note that probability $p_{collision}$ that a transmitted packet encounters a collision 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 τ for each protocol, and $p_{collision}$ is equal to

$$p_{collision} = 1 - (1 - \tau)^{N-1}. \tag{7}$$

Probability that a node is found in backoff stage i is given as:

$$\Pr(b = i) = \tau \frac{\Pr(b=i\,|X)}{\Pr(X\,|b=i)},\ i \in (\,0, \ldots, B). \tag{8}$$

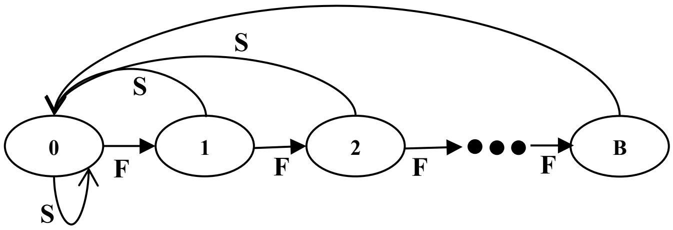


Fig. 4. Markov chain model for the backoff stage.

By summing all the values of i, we get

$$\sum_{i=0}^{B} \Pr(b = i) = \tau \sum_{i=0}^{B} \frac{\Pr(b=i\,|X)}{\Pr(X|b=i)}. \tag{9}$$

τ can be calculated as

$$\tau = \frac{1}{\sum_{i=0}^{B} \frac{\Pr(b=i\,|X)}{\Pr(X|b=i)}}. \tag{10}$$

A Markov Chain model for the backoff stage is depicted in **Fig. 4**, and the transition probabilities of the backoff stage are given as

$$\Pr\{b(t+1) = i \mid b(t) = i-1\} = p_{failure,I}^{RMM},\ i = 1, \dots, B, \tag{11}$$

$$\Pr\{b(t+1) = 0 \mid b(t) = i\} = 1 - p_{failure,I}^{RMM},\ i = 1, \dots, B-1, \tag{12}$$

$$\Pr\{b(t+1) = 0 \mid b(t) = B\} = 1 \tag{13}$$

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

$$\Pr(b = i|X) = \frac{(1-p_{failure,I})p_{failure,I}^{i}}{\left(1-p_{failure,I}^{B+1}\right)},\ \ i \in (0, \dots, B). \tag{14}$$

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 (backoff and transmission cycle in backoff stage i). Because a time slot corresponds to a backoff counter decrement,

$$\Pr(X|b = i) = \frac{1}{1+E[c_i]},\ \ i \in (0, \dots, B), \tag{15}$$

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 $(0,W_i)$.

Using (10) and (14)-(15), we get

$$\tau_{LBP} = \frac{1}{1+\frac{1-p_{failure}}{1-p_{failure}^{B+1}}\sum_{i=0}^{B} p_{failure}^{i} E[c_i]}. \quad (16)$$

$$\tau_{RMM} = \frac{1}{1+\frac{1-p_{failure,I}}{1-p_{failure,I}^{B+1}}\sum_{i=0}^{B} p_{failure,I}^{i} E[c_i]}. \quad (17)$$

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

$$\Pr(b = i) = \tau \frac{(1-p_{failure,I})p_{failure,I}^{i}}{1-p_{failure,I}^{B+1}}(1 + E[c_i]). \quad (18)$$

Note that (7) and (16)-(17) represent a nonlinear system with two unknown τ and $p_{failure}$, which can be solved by using numerical techniques.

3.3 Packet Drop Probability

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

$$p_{d,I} = \sum_{i=0}^{B} \Pr(\text{drop}|b = i) \Pr(b = i). \quad (19)$$

A packet in backoff stage i will be dropped if it reaches maximum backoff stage B (i.e., it collides for $B - i$ times) and it collides during the last transmission attempt. The $\Pr(\text{drop}|b = i)$ of LBP and RMM protocol is given as

$$\Pr^{LBP}(\text{drop}|b = i) = p_{failure}^{B+1-i}, \quad (20)$$

$$\Pr^{RMM}(\text{drop}|b = i) = p_{failure,I}^{B+1-i}. \quad (21)$$

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. Failure of P and B packets are considered as droped and are represented as $\mathrm{p_{d,P}}$ and $\mathrm{p_{d,B}}$, respectively

3.4. Decodable Frame Rate

The decodable frame rate is a metric used to evaluate the quality of video stream, and has been used in earlier work [19]. In this section, we present the analytical estimation of the decodable frame rate. In GOP, an I-frame is successfully decodable only if all the packets belonging to the tagged 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 belonging 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 consist the tagged B-frame have been successfully received. Therefore, the expected numbers of successfully decodable I, P and B-frames for the whole video are [19]

$$N_{dec-I} = \left(1 - P_{d,I}\right)^{C_I} \alpha, \tag{22}$$
$$N_{dec-p} = (1 - P_{d,I})^{C_I} \sum_{i=1}^{N_P} (1 - p_{d,P})^{iC_P} \alpha, \tag{23}$$

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

where C_I is the average number of packets in one I-frame, α is the number of GOP, 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 previous section, the respective number of successfully decodable frames can be analytically estimated.

4. Numerical Results

4.1 Transmission and Failure Probability as a Function of the Number of Nodes

Fig. 5 shows an example of finding the failure probability and transmission probability. The intersection points are the values of failure probability and transmission probability with the fixed number of nodes. The transmission probability of the RMM protocol is higher than the LBP because RMM protocol transmits ACK frame only for packets belonging to the I-frames and, there is no ACK frame for packets of P and B-frames. The average packet drop probability is shown in **Fig. 6**. To calculate the drop probabilities, first we need to calculate the transmission and failure probabilities as a function of number of nodes. In RMM and LBP protocols, packets are dropped because of retry limit exhaustion. The drop probability increases as the number of nodes increases because the failure probability increases. The drop probabilities of P and B-frames are higher than I-frames because there is no retransmission for P and B-frames and failure of P and B-frames are considered as drop.

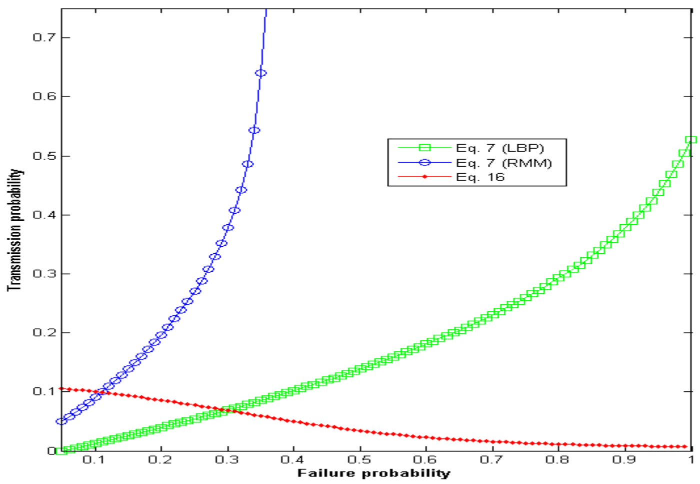


Fig. 5. Failure probability and Transmission Probability when the number of nodes is 5.

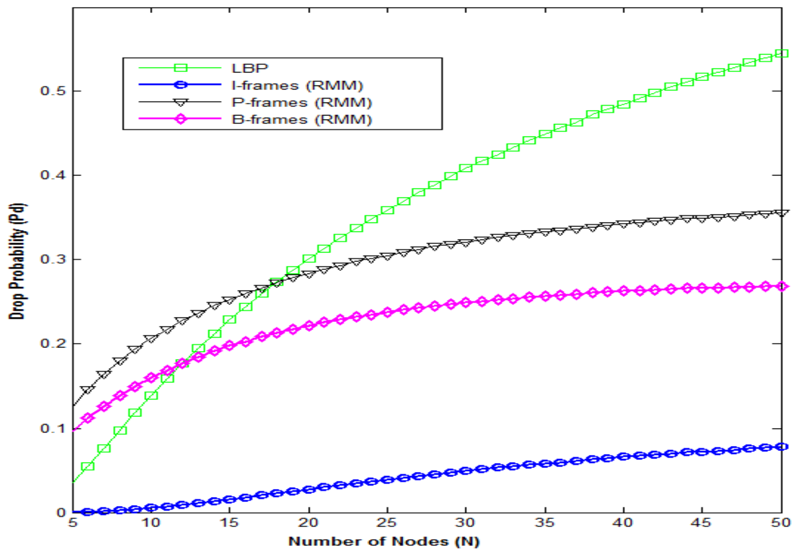


Fig. 6. Average drop probability as a function of the number of nodes.

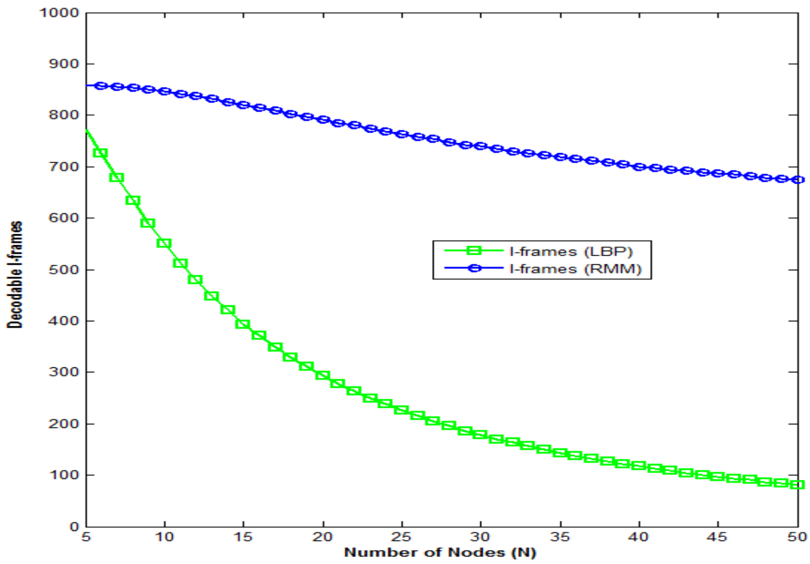


Fig. 7. Decodable I-frames as a function of the number of nodes.

4.2. Decodable Frame Rate as a Function of the Number of Nodes

Fig. 7 shows the decodable number of I-frames in RMM and LBP protocol as a function of number of nodes. The number of decodable I-frames in RMM protocol is higher than LBP. Even though it is not shown in the figure, the successful reception of I-frames also increases the decodable P and B-frames.

Fig. 8. Erroneous I-frame No: 71 from the "car-phone_qcif".

Fig. 9. Decodable I-frame No: 71 from the "car-phone_qcif".

4.3 Effect of I-frame retransmission on Video Quality

Fig. 8 shows the impact of the packets loss belonging to the I-frame on video quality. It can be seen that due to non-decodable I-frame, the video quality is not good. **Fig. 9** shows the successfully received or fully decodable I-frame. If packets belonging to the I-frame fail to be received successfully by the leader node or any multicast receiver, a NACK frame is transmitted. As a result, sender node retransmits the packet. This retransmission can increase the number of decodable I-frame as shown in **Fig. 7** and also enhanced the video quality as shown in **Fig. 9**.

5. Conclusion

Multicasting is an efficient way compared to unicasting in supporting multimedia applications. Therefore, in this paper, we propose a leader-based reliable multicast MAC layer protocol for multimedia applications. Results show that proposed protocol can enhance the video quality by increasing the number of decodable I-frames.

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Muhammad Khalil Afzal received his B.S and M.S degrees in Computer Science from COMSATS Institute of Information Technology, Wah Campus, Pakistan in 2004 and 2007, respectively. He has served as lecturer from January 2008 to November 2009 in Bahauddin Zakariya University Multan Pakistan, and from December 2009 to June 2011 in King Khalid University Abha, Saudi Arabia. Currently, he is PhD candidate in Wireless Information Networking Lab at department of information and Communication Engineering, Yeungnam University, Gyeongsangbuk-do Korea. His current research interest includes reliability in multicasting, and Cooperative Networks.

Byung-Seo Kim received his B.S. degree in Electrical Engineering from In-Ha University, In-Chon, Korea in 1998 and his M.S. and Ph.D. degrees in Electrical and Computer Engineering from the University of Florida in 2001 and 2004, respectively. His Ph.D. study was supervised by Dr. Fang. Between 1997 and 1999, he worked for Motorola Korea Ltd., PaJu, Korea as a Computer Integrated Manufacturing (CIM) Engineer in Advanced Technology Research and Development (ATR&D). From January 2005 to August 2007, he worked for Motorola Inc., Schaumburg Illinois, as a Senior Software Engineer in Networks and Enterprises. His research focuses in Motorola Inc. were designing protocol and network architecture of wireless mission critical communications. He is currently an Associate Professor at the Department of Computer and Information Communication Engineering in Hongik University, Korea. His research interests include the design and development of efficient link-adaptable MAC protocols, cross layer architectures, multi-MAC structures, and resource allocation algorithms for wireless networks.

Sung Won Kim received his B.S. and M.S. degrees from the Department of Control and Instrumentation Engineering, Seoul National University, Korea, in 1990 and 1992, respectively, and his Ph.D. degree from the School of Electrical Engineering and Computer Sciences, Seoul National University, Korea, in August 2002. From January 1992 to August 2001, he was a Researcher at the Research and Development Center of LG Electronics, Korea. From August 2001 to August 2003, he was a Researcher at the Research and Development Center of AL Tech, Korea. From August 2003 to February 2005, he was a Postdoctoral Researcher in the Department of Electrical and Computer Engineering, University of Florida, Gainesville, USA. In March 2005, he joined the Department of information and Communication Engineering, Yeungnam University, Gyeongsangbuk-do, Korea, where he is currently an Associate Professor. His research interests include resource management, wireless networks, mobile networks, performance evaluation, and embedded systems.