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Rate-Adaptive Multimedia Multicasting over IEEE 802.11 Wireless LANs

Youngsam Park*, Yongho Seok†, Nakjung Choi†, Yanghee Choi† and Jean-Marie BONNIN‡

* Wibro Technology Lab., Telecommunication R&D Center
SAMSUNG ELECTRONICS CO., LTD, Suwon, Korea
Email: youngsam.park@samsung.com

† School of Computer Science and Engineering
Seoul National University, Seoul, Korea
Email: {yspark, yhseok, fomula, yhchoi}@mmlab.snu.ac.kr

‡ Network and Multimedia Department
ENST Bretagne, Rennes, France
Email: jm.bonnin@enst-bretagne.fr

***Abstract*— Multimedia services are a major application for IEEE 802.11 WLANs but the current standard does not utilize any rate adaptation technique for multicasting. SARM (SNR-based auto rate for multicast) is a new rate adaptation mechanism for multicasting multimedia content in IEEE 802.11 WLAN environments. Since there is no RTS/CTS or MAC ACK frame in multicast packet transmissions, SARM uses an auxiliary signaling method to obtain the channel quality for each mobile node. By adapting a transmission rate based on the SNR of the node experiencing the worst air-channel, SARM achieves the best quality of service for the prevailing conditions. This reduces the proportion of wireless channel resources assigned to multicasting, allowing more bandwidth for unicast flows.**

I. Introduction

IEEE 802.11 Wireless LANs enable people to stay connected to the Internet while they are away from their home or office. Currently, wireless LAN service providers are broadening hotspot areas to improve Internet accessibility. One of the major applications driving wireless LAN services is multimedia streaming, which is based on the ability to multicast the same multimedia contents to a group of users simultaneously, thus reducing the bandwidth requirement. Compared to traditional applications such as Web services, multimedia streaming requires a higher QoS (Quality of Service) level, which is hard to meet in a wireless LAN because of its relatively low bandwidth and unstable wireless link conditions, even though the IEEE 802.11b standard supports transmission rates up to 11Mbps and IEEE 802.11a/g supports 54Mbps.

A component which is critical to the performance of a wireless link is the transmission rate adaptation mechanism. To improve throughput in wireless LANs, an access point (AP) needs to estimate the channel state so that it can select an appropriate transmission rate. In a unicast environment, an AP can determine the channel state of the receiving mobile node through a feedback mechanism such as RTS/CTS or the MAC ACK frame, and then adapt its transmission rate appropriately [3][4][5]. But multicast has no feedback mechanism because it is usually an unreliable one-to-many communication scenario. Therefore, unicast transmission rate adaptation mechanisms, such as ARF (auto rate fallback) [3] and RBAR (receiver based auto rate) [4], cannot be directly applied to multicast since they are based on the estimation of individual channel states. Even through IEEE 802.11a/b/g supports transmission rates up to 11Mbps or 54Mbps, mutlicast packets are often transmitted at a configurable basic rate (e.g., 1 or 2Mbps in the case of 802.11b). Such transmissions are a significant waste of wireless channel resources with a negative impact on the whole network [1].

In this paper, we propose a novel multicast transmission rate adaptation mechanism that aims to maintain an adequate quality of multimedia streaming while using minimum wireless channel resources. We call this approach SARM (SNR-based auto rate for multicast).

The rest of this paper is organized as follows. Section II gives our motivation. We describe our transmission rate adaptation mechanism, SARM, in Section III, and then present experimental results in Section IV. Finally, we conclude our work in Section V.

II. Motivation

Rate adaptation mechanisms are not standardized in IEEE 802.11 WLANs, and each manufacturer chooses its own. A small number of rate adaptation mechanisms have been implemented and deployed for unicast transmission. However, due to the absence of link-level feedback signals in multicast packet transmissions, no rate adaptation mechanism for multicasting has been proposed. Most commercial APs use a fixed and relatively very low transmission rate for multicast, although more recent APs have included a manual configuration facility which enables an administrator to select a transmission rate. Other APs have only one transmission rate, which is fixed by the manufacturer, usually at one of the basic rates (1Mbps or 2Mbps).

Generally, a small amount of distortion, caused by a few incorrect bits, in multimedia content are considered tolerable by service subscribers, a perfectly error-free transmission is not necessary. Some video encoders such as MPEG4/AVC and H.264 provide FEC (forward error correction) in the application layer to make services more error-resilient. Provided that erroneous multimedia packets are not discarded in the link layer, they can be recovered by FEC. QoS is usually improved by making the best of erroneous multicast frames instead of dropping them [2], which must be considered in our mechanism.

III. Multicast Transmission Rate Adaptation Mechanism

In existing unicast transmission rate adaptation mechanisms, an AP collects channel state information, such as SNR values or the number of successful/failed transmissions between the AP and each mobile node. Based on this information, the AP dynamically determines a transmission rate for each mobile node, which can be between 1Mbps and 11Mbps in IEEE 802.11b, and up to 54Mbps in IEEE 802.11a/g. However, due to the absence of link-level feedback signals in multicasting, an AP cannot use the same method and collect channel state information from each mobile node. In SARM, a supplementary link-level signaling method has been introduced, which allows an AP to collect channel state information for each mobile node participating in a multicast group. We suggest that the AP should use the SNR values received from each mobile node as the criterion for rate selection. Using this feedback, the AP can collect SNR values for all the mobile nodes participating in multicast groups and then determine the transmission rate for each group. These SNR-based rate adaptation mechanisms have already been shown to be feasible [4][5].

A. Channel Probing Mechanism in SARM

The key to our rate adaptation mechanism is that each mobile node measures the channel quality and feeds back the information to the AP. Algorithm 1 shows how the necessary signaling between an AP and each mobile node is introduced.

Algorithm 1 Channel Probing Mechanism

Mobile node joined in multicast group

if $Beacon\ Contains\ No\ Multicast\ Group\ Option$ **then**
  $//InitializationPhase$
  Measure Received SNR;
  Send Feedback Signal;
else
  $//NormalPhase$
  Measure Received SNR;
  if $PrevMinSNR > RecvSNR$ **then**
    Send Feedback Signal;
  else if $CurrentNode == PrevMinSNRNode$ **then**
    Send Feedback Signal;
  end if
end if

AP

Update $Multicast\ Group\ Table$ when IGMP is received;
Update $Channel\ States\ Table$ when Feedback is received;
if $CurrentTime == TBTT$ **then**
  while $Each\ Multicast\ Group\ in\ MGT$ **do**
    Find Mobile Nodes(MNs) in Multicast Group;
    Find $PrevMinSNR$ and $PrevMinSNRNode$ among MNs in CST;
    Attach $Multicast\ Group\ Option$ in Beacon Frame
  end while
  Send Beacon;
end if

In the initialization phase, all mobile nodes which are participating in multicast groups estimate their channel state by measuring the received SNR (rSNR) values of beacons periodically broadcast by an AP. If the received beacon frame contains no additional information about multicast transmission rates related to the multicast groups that it has joined, the mobile node sends a feedback signal which contains the rSNR value. To minimize the possibility of collisions between feedback signals from different nodes, each mobile node adjusts its backoff time depending on the rSNR value. A mobile node with a low rSNR receives a higher priority than one with a high rSNR. The following Equation 1 and 2 show how to determine the backoff time when mobile nodes are trying to transmit these feedback signals.

$$Backoff\ Time = I(rSNR) \cdot CW_{\min} \qquad (1)$$

$$I(rSNR) = \frac{rSNR - MinSNR_{1Mbps}}{MinSNR_{11Mbps} - MinSNR_{1Mbps}}, \quad MinSNR_{1Mbps} \le rSNR \le MinSNR_{11Mbps} \qquad (2)$$

The constants $MinSNR_{1Mbps}$ and $MinSNR_{11Mbps}$ are the minimum rSNR values required for accurate reception of frames transmitted at 1Mbps and 11Mbps respectively. If mobile nodes overhear the feedback signals transmitted from other mobile node in the same multicast group, they cancel I transmission of their own feedback signal. For this purpose, we need to know the identifier and rSNR value of the mobile node with the worst channel quality among all nodes in the multicast group.

By means of this mechanism, a transmission rate is determined, based on the lowest rSNR value in a multicast group. On receiving feedback signals from mobile nodes which are participating in multicast groups, the AP constructs a CST (channel states table) for each mobile node. An entry in the CST has a list of hardware addresses and the node's rSNR value. To make the AP aware of the presence of multicast groups, it needs to snoop on the IGMP messages from mobile nodes. From information gathered in this way, the AP creates a MGT (multicast group table) that consists of multicast group addresses and a list of the identifiers of the mobile nodes participating in each multicast group. When the AP transmits a multicast packet, it first searches the MGT to identify the mobile nodes that are expecting to receive the packet. Then, the AP finds the mobile node that has the minimum rSNR among the identified nodes, and then changes the transmission rate to correspond to this rSNR value.

After the initialization phase, it is necessary for the AP to inform all mobile nodes in the multicast group of the minimum rSNR value ($PrevMinSNR$) and the hardware address ($PrevMinSNRNode$) of the mobile node that reported the minimum rSNR value to all mobile nodes. This information is piggybacked in the original beacons sent by the AP. Figure 1 shows the modified beacon frame format. After receiving the beacon each mobile node periodically measures the channel state and compares the current rSNR value with the minimum rSNR announced by the beacon. If the currently measured rSNR value is greater than the minimum SNR in the beacon frame, the mobile node does not send feedback signals to the AP. Otherwise, the mobile node needs to update the minimum rSNR value with the current rSNR, and then send feedback to the AP. If the AP does not receive feedback signals from the mobile node with the minimum rSNR value within a specified duration (e.g., 3 beacon intervals) the AP concludes that the node has no connection and restarts the initialization process.

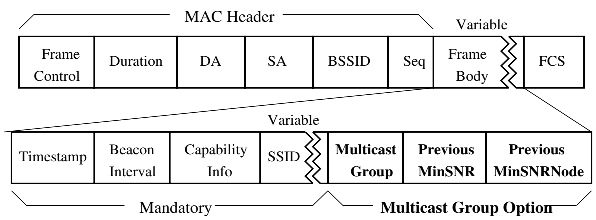

Fig. 1. New beacon frame format

B. Transmission Rate Selection Criterion in SARM

When an AP selects a multicast transmission rate based on rSNR values from mobile nodes, it needs to be aiming to meet some specific criterion. We are aiming to maintain an adequate quality of multimedia content while using the minimum portion of wireless resources for multicasting. We therefore introduce the concept of PSNR (Peak Signal-to-Noise Ratio) as the criterion for rate selection. In related works [6][7] focusing on the effect of bit errors on video quality, the PSNR metric has been generally used as an estimation of video quality. Results from this work show that the PSNR should be maintained above 30 to guarantee an adequate quality of multimedia content. SARM aims to meet the same criterion. In order to achieve this, it is necessary to construct a database which represents the relationship between SNR and PSNR for each transmission rate between 1Mbps and 11Mbps. We conducted preliminary experiments to derive this SNR-PSNR relationship. The details will be presented in section IV-A and experimental results are shown in Figure 2.

C. Quality Enhancement for Multimedia Contents

We also propose a simple method to enhance the quality of multimedia content and the utilization of wireless channel resources. Due to the absence of a retransmission mechanism in multicasting, dropping erroneous frames at mobile nodes wastes wireless channel resources and also degrades the quality of multimedia content. In SARM, the mobile nodes participating in multicast groups do not always drop erroneous multicast frames. That is made possible by turning off FCS (Frame Check Sequence) for the MAC layer and unsetting the UDP checksum for the transport layer. As a result, our rate adaptation mechanism can use a higher transmission rate for a given channel condition. Table I shows the relationship between SNR and transmission rate. The details will be presented in section IV-A.

TABLE I
SNR-TRANSMISSION RATE MAPPING TABLE

	required SNR for PSNR larger than 30	
	SARM with FCS-OFF	SARM with FCS-ON
11Mbps	26	30
5.5Mbps	21	24.5
2Mbps	17.5	21

IV. EXPERIMENTAL RESULTS

A. Determining the SNR-PSNR relationship

We implemented a simplified multicast test-bed which delivers a MPEG2 stream from a multicast server to mobile nodes. This experiment has two primary goals. The first is to create a database which shows the relationship between SNR and PSNR values for each transmission rate. The second is to analyze the effect on the PSNR of the received multimedia content when mobile nodes do not drop multicast frames that contain errors.

The devices used in this experiment are an AP, a multicast server and mobile nodes. The multicast server is a Pentium 4 desktop PC running RedHat Linux 7.3. This server encodes MPEG2 data from the original YUV format at a rate of 1Mbps, and transmits it at 1Mbps. The AP (Orinoco AP-1000) can monitor received SNR values for a specified mobile node by sending and receiving probing packets. The administrator can also set up a fixed multicast transmission rate. Each mobile node is a Pentium 3 Laptop PC running RedHat Linux 7.3, and incorporating a wireless interface card based on an Intersil chipset [8]. While receiving MPEG2 data from the AP, a mobile node disables checksum mechanisms, such as FCS (Frame Check Sequence) in the MAC layer or UDP Checksum, so that it can utilize erroneous frames.

The experiment was conducted as follows. We located each mobile node so as to maintain a specified received SNR. Actual SNR values are monitored and logged by the AP. The multicast server sends MPEG2 data to receivers via the AP at a multicast transmission rate of 2Mbps, 5.5Mbps or 11Mbps. After receiving the data, the receivers decode the MPEG2 data to YUV format and calculate a PSNR value by comparing it with the original data, also in YUY format. In all these experiments, no multicast process taking place above the MAC layer, such as joining a multicast group, is allowed, for simplicity. By changing the locations of the mobile nodes and we can obtain an SNR-PSNR relationship for each transmission rate (Figure. 2).

Based on these results (which show a similar form to the well-known SNR-BER relationship) we can determine appropriate SNR values to keep the PSNR over 30 at each transmission rate and a database of mapping between SNR and transmission rate can then be built, which allows the AP automatically to determine the transmission rate to be used for multicasting. Table I shows this database. We see that SARM with FCS-OFF almost always permits a transmission speed that in one level higher than that achieved by FCS-ON, for the same channel condition.

B. NS simulation - Protocol implementation

We implemented the proposed protocol using the NS-2 network simulator, and evaluated the performance of SARM compared to a fixed-rate multicast protocol. All the experiments use Ricean propagation [9] as a channel propagation model for IEEE 802.11 PHY, and the NOAH ns-extension [10] as an infrastructure mode. The received power thresholds for different transmission rates were based on the distance ranges specified in the Orinoco 802.11b card data sheet. We also used the channel error model derived in Section IV-A to simulate an IEEE 802.11b channel. Figure 3(a) shows the measured packet error rate (PER) as a function of SNR for each transmission rate. The cumulative density function (CDF) of bit errors in each frame transmitted from the AP is shown in Figure 3(b), which suggests that the bit errors in each frame can be modeled by a log-normal distribution, in which the most corrupt packets have several error bits.

Based on these results, we were able to model channel errors in an IEEE 802.11 WLAN. The simulated topology consists of a multicast server, an AP in a wired network, and three mobile nodes connected to the AP in a wireless network. The multicast server sends MPEG2 video data at 1Mbps and the packet size is 1220 bytes, as in the previous section.

Figure 4 shows the service quality which SARM is capable of providing for different wireless channel states, with FCS-ON and FCS-OFF. Using a fixed-rate multicast protocol, transmitting at 2Mbps, SNR values over 26 correspond to PSNR values over 50. The probability of an error in a packet is very low because the packet is always transmitted at a low transmission rate irrespectively of wireless channel states. However, the PSNR declines as the channel state gets worse. SARM uses a higher transmission rate if the channel state is good and only transmits more slowly if the service quality of multimedia contents falls below the specified threshold (PSNR=30 in our experiments). As shown in Figure 4(b), SARM with FCS-OFF uses a rate of 11Mbps for values of SNR between 34 and 26. This response follows the SNR-Transmission rate mapping of Table I. In general, a low SNR value means a lower PSNR but, if the SNR value falls below 26, SARM chooses a lower transmission rate which keeps the PSNR value over 30, as shown in Figure 4. Setting FCS-OFF enables SARM to achieve a better service quality when the air channel is bad, which confirms that it is better to use erroneous frames rather than to drop them.

Figure 5 shows the average throughput of background TCP flows during multimedia streaming. These results show how much bandwidth the multimedia stream occupies. Higher background throughput was achieved using SARM than with the fixed-rate multicast protocol. For example, if the SNR is 27, SARM chooses a transmission rate of 11Mbps for multimedia streaming if FCS is OFF, and 5.5Mbps if FCS is ON. These rates are much higher than the 2Mbps of the fixed-rate multicast protocol. So, background TCP flows can achieve higher average throughput when using SARM.

V. Conclusion

SARM is a novel MAC-layer multicast mechanism with a multi-rate transmission capability. By changing multicast transmission rate on the basis of SNR values reported by mobile nodes, the wireless channel is used much more efficiently. SARM can maintain the quality of multimedia content under changing air-channel conditions. We verified its performance using a test-bed and the ns-2 simulator. Both experimental and simulation results show that SARM outperforms a fixed-rate multicast protocol. We also show that the service quality for multimedia content is improved by using erroneous frames in the MAC layer.

VI. Acknowledgment

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References

[1] M. Heusse, F. Rousseau, G. Berger-Sabbatel and A. Duda, "Performance anomaly of 802.11b," Proc. IEEE INFOCOM 2003, April 2003.

[2] L-A. Larzon, M. Degermark, S. Pink, L-E. Jonsson and G. Fairhurst, "The lightweight user datagram protocol (UDP-Lite)," IETF, RFC 3828, July, 2004.

[3] A. Kamerman and L. Monteban, "WaveLAN-II: A high-performance wireless LAN for the unlicensed band," *Bell Labs Technical Journal*, pp. 118-133, Summer 1997.

[4] G. Holland, N. Vaidya and P. Bahl, "A rate-adaptive MAC protocol for multi-hop wireless networks," Proc. ACM MOBICOM, Rome, July 2001.

[5] B. Sadeghi, V. Kanodia, A. Sabharwal and E. Knightly, "OAR: An opportunistic auto-rate media access protocol for ad hoc networks," Proc. ACM MOBICOM'02, Atlanta, GA, Sep 2002.

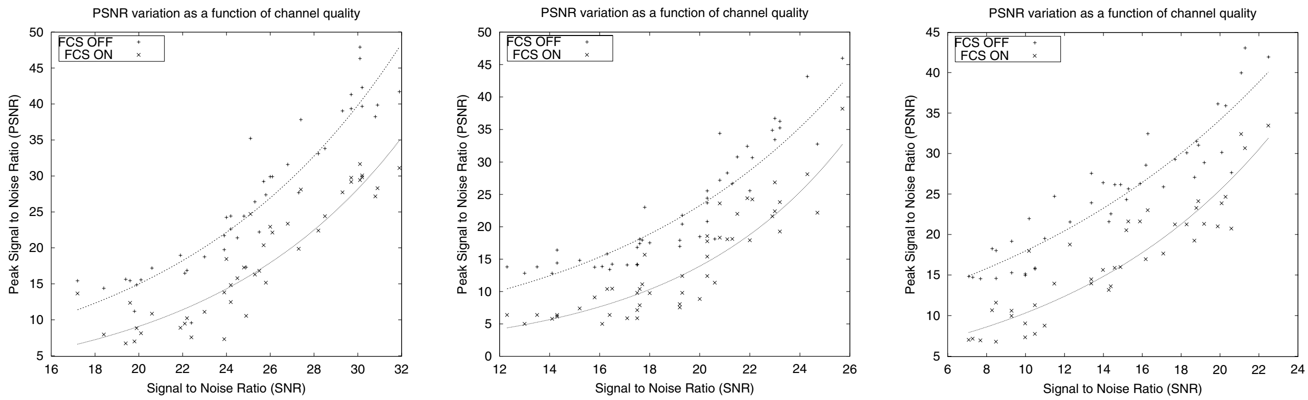
[6] S. Gringeri, R. Egorov, K. Shuaib, A. Lewis and B. Basch, "Robust compression and transmission of MPEG-4 video," Proc. the Seventh ACM International Conference on Multimedia (Part 1), 1999.

[7] I. Bouazizi, "Size-distortion optimized proxy caching for robust transmission of MPEG-4 video," LNCS 2899, Proc. International Workshop on Multimedia Interactive Protocols and Systems, November 18-21, 2003, Napoli, Italy.

[8] "HFA3861B; Direct sequence spread spectrum baseband processor," January 2000.

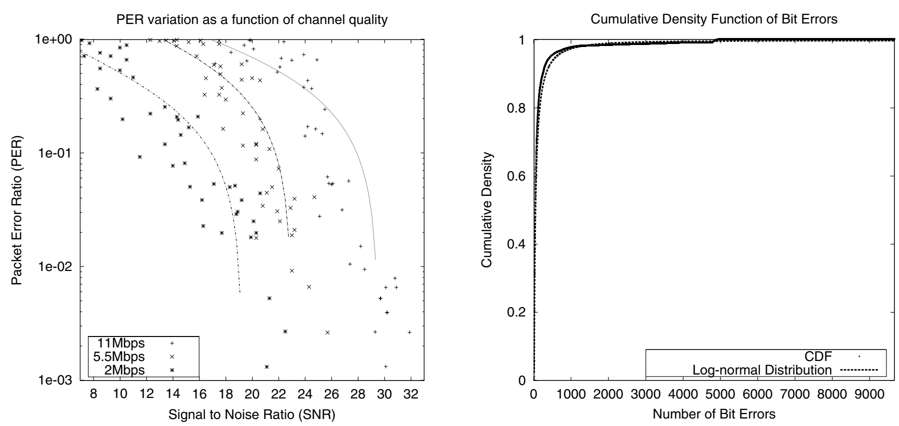
[9] R. Punnose, P. Nikitin and D. Stancil, "Efficient simulation of Ricean fading within a paket simulator," Proc. IEEE Vehicular Technology Conference, pages 764-767, 2000.

[10] J. Widmer, "Extensions to the ns network simulator (http://www.informatik.uni-mannheim.de/pi4/projects/MobileIP/ns-extension/)".



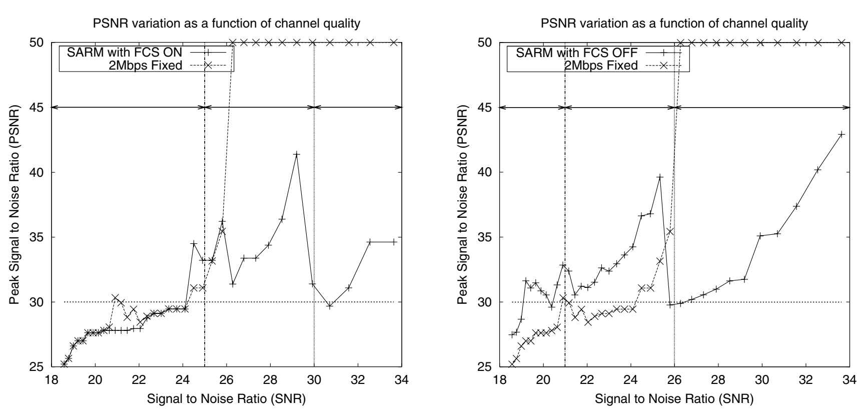
(a) Transmission rate of 11Mbps (b) Transmission rate of 5.5Mbps (c) Transmission rate of 2Mbps

Fig. 2. Relationship between SNR and PSNR for different transmission rates



(a) Packet error rate (b) Cumulative density function

Fig. 3. Bit error in each transmitted frame



(a) SARM with FCS-ON (b) SARM with FCS-OFF

Fig. 4. Video quality

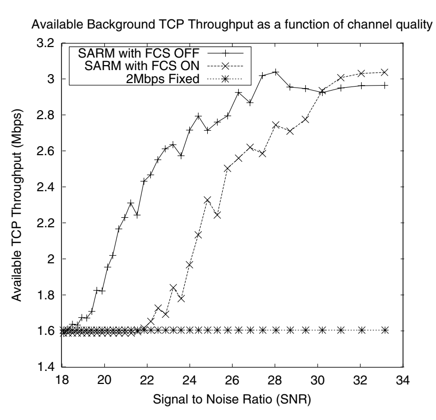


Fig. 5. Background TCP throughput