Hop-Based Priority Technique Using 802.11e for Multimedia Streaming

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Abstract — *Multimedia streaming is a key service provided by certain home appliances. Multimedia streaming services require sufficient bandwidth and delay variation to achieve good quality of service (QoS). Previous wireless ad hoc networks could not satisfy these requirements. Therefore, we propose a hop-based priority (HBP) technique using 802.11e for ensuring a good QoS of a multimedia streaming service. Multimedia streaming data packets are assigned a higher priority after every hop. This can minimize the contention between the previous packet and the next packet; further, the probability of channel access for the previous packet increases. We demonstrate a better performance by modeling and simulation*[1].

Index Terms — Wireless ad hoc network, Multimedia Streaming, 802.11e EDCF, Hop-based priority.

I. INTRODUCTION

A home network is a technology that connects information appliances like Internet TV and Internet refrigerator and receives services through a network. A home network helps to control home appliances, to connect the Internet, and to use data in a home server. In particular, multimedia streaming is the main theme of research. Users can be served various multimedia contents by a home network if the home server is deployed and serves multimedia streaming. A large bandwidth and real-time operation are required for multimedia streaming services that consist of a large amount of data and have to be played in real time [1].

Networks have evolved from wired to wireless forms. Wireless local area network (WLAN) supports scalability, flexibility, and ubiquity, and these features make WLAN more popular. However, a wireless network has certain drawbacks such as interference and limitation of the communication range. A wireless multi-hop network has overcome the limit of WLAN's communication range.

The applications and services of wired and wireless networks are not different. Services that need a good quality of service (QoS) like voice over IP (VoIP) and multimedia streaming can be provided through wireless networks. However, it is difficult for WLAN to guarantee a high QoS with respect to bandwidth and delay.

In order to solve this problem, the 802.11e standard [2] is designed. 802.11e has a hybrid coordination function (HCF) and an enhanced distributed coordination function (EDCF) to support QoS that could not be supported in the previous 802.11 standard. HCF extends the point coordination function (PCF) to support the immediate sending of data. An access point (AP)-based infrastructure mode is used in HCF. EDCF is designed to support a priority upon a distributed coordination function (DCF). EDCF can be used in both the ad hoc mode and the infrastructure mode. This technique just assigns priority in packets and it is not enough because of contention with previous and next packets.

In this paper, we propose a hop-based priority (HBP) technique to provide more bandwidth and constant delay by minimizing the contention in an 802.11e wireless ad hoc network with EDCF. Previous techniques assign priorities to packets from the source to the destination. However, the priority of a packet is changed after every hop in HBP. This causes a difference in priority between the previous and the next packets and minimizes contention.

This paper is organized as follows. Requirements of multimedia streaming are given in section 2. An overview of 802.11e EDCF is given in section 3. Related studies are discussed in section 4. Section 5 provides an overview of the HBP technique that we propose. We present the analysis and evaluation of the HBP technique by modeling and simulation in section 6. Finally, we conclude and state some future work in section 7.

II. REQUIREMENTS OF MULTIMEDIA STREAMING

Requirements of multimedia streaming are different from those of a general file transfer. Bandwidth is the most important factor in a file transfer. A large bandwidth leads to a short transfer time and guarantees a good QoS. However, streaming has variable requirements with respect to the type of streaming service.

Delay is a very important aspect in VoIP. Voice data size is not kept too large because of using codec. However, the real-time constraint is important, and the delay must be small. Nevertheless, the bandwidth must be considered in VoIP. A higher bandwidth results in a good quality of voice. However, if the other person's voice is recognizable, the quality of voice

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is sufficient. Delay must be so sufficiently small that no delay is felt, and real-time operation is required. Therefore, delay is more important than bandwidth in VoIP [3].

Bandwidth is important in multimedia streaming. Multimedia data size typically ranges from a few megabytes to a few gigabytes. These large amounts of data require sufficient bandwidth to minimize buffering. If sufficient bandwidth is not guaranteed, users who are served by a multimedia streaming service must have periodic buffering or download all multimedia data before play. Therefore, bandwidth is an important factor for multimedia streaming [4].

Another important requirement for multimedia streaming is the delay variation or jitter. Buffering for multimedia streaming is decided by the delay variation. When the delay variation is large, the buffering time increases considerably in order to minimize the effect of delay. Delay is determined upon the initialization of multimedia streaming. If a delay spike occurs after initialization, additional buffering is required and users experience a low quality of multimedia. Moreover, delay variation is related to bandwidth. Therefore, minimizing delay variation is important in order to maintain a constant bandwidth.

III. 802.11E ENHANCED DISTRIBUTED COORDINATION FUNCTION [1]

The 802.11 legacy MAC does not support the concept of differentiating frames with different priorities. Basically, the DCF is supposed to provide a channel access with equal probabilities to all stations contending for the channel access in a distributed manner. However, equal access probabilities are not desirable among stations with different priority frames. The emerging EDCF is designed to provide differentiated, distributed channel accesses for frames with eight different priorities (from 0 to 7) by enhancing the DCF. Distinct from the legacy DCF, the EDCF is not a separate coordination function. Rather, it is a part of a single coordination function, called the hybrid coordination function (HCF), of the 802.11e MAC. The HCF combines the aspects of both DCF and PCF. However, the detailed aspects of the HCF are beyond the scope of this paper as we focus on EDCF.

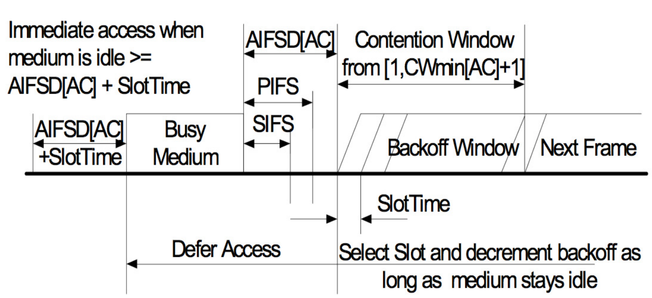

Fig. 1. IEEE 802.11e EDCF channel access

Each frame from the higher layer arrives at the MAC along with a specific priority value. Then, each QoS data frame carries its priority value in the MAC frame header. An 802.11e STA shall implement four access categories (ACs), where an AC is an enhanced variant of DCF 0.

Basically, an AC uses AIFSD[AC], CWmin[AC], and CWmax[AC], instead of DIFS, CWmin, and CWmax, of the DCF, respectively, for the contention process to transmit a frame belonging to the access category AC. AIFSD[AC] is determined by

$$AIFSD[AC] = SIFS + AIFS[AC] * SlotTime$$

where AIFS[AC] is an integer greater than zero. Moreover, the backoff counter is selected from [1,1+CW[AC]] instead of [0,CW] as in the DCF. Fig. 1 shows the timing diagram of the EDCF channel access.

The values of AIFS[AC], CWmin[AC], and CWmax[AC], which are referred to as the EDCF parameters, are announced by the AP via beacon frames. The AP can adapt these parameters dynamically depending on the network conditions. Basically, the smaller the AIFS[AC] and CWmin[AC] are, the shorter is the channel access delay for the corresponding priority, and hence, the higher is the capacity share for a given traffic condition. However, the probability of collisions increases in the case of a smaller CWmin[AC]. These parameters can be used in order to differentiate the channel access among traffic of different priorities.

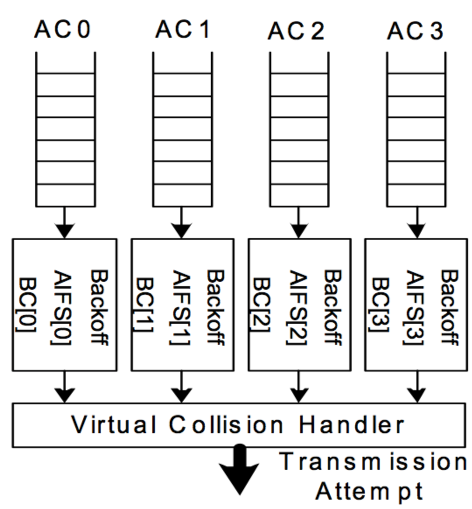

Fig. 2. Four access categories (ACs) for EDCF

Fig. 2 shows the 802.11e MAC with four transmission queues, where each queue behaves as a single enhanced DCF contending entity, i.e., an AC, where each queue has its own AIFS and maintains its own backoff counter BC. When there is more than one AC completing the backoff at the same time, the collision is handled in a virtual manner. That is, the highest priority frame among the colliding frames is chosen and transmitted, and the others perform a backoff with increased CW values.

IEEE 802.11e defines a transmission opportunity (TXOP) as the interval of time when a particular STA has the right to initiate transmissions. Along with the EDCF parameters of AIFS[AC], CWmin[AC], and CWmax[AP], the AP determines and announces the limit of an EDCF TXOP interval for each AC, i.e., TXOPLimit[AC], in beacon frames. During an EDCF TXOP, an STA is allowed to transmit multiple MPDUs from the same AC with a SIFS time gap between an ACK and the subsequent frame transmission.

IV. RELATED STUDIES

Many studies have been conducted for streaming multimedia data through wireless networks. Most studies use a cross-layer approach to notify the network state to the encoder, and the encoder encodes according to the network state. After 802.11e that supports QoS was published, researches have been focused on using H.264 SVC with the 802.11e priority.

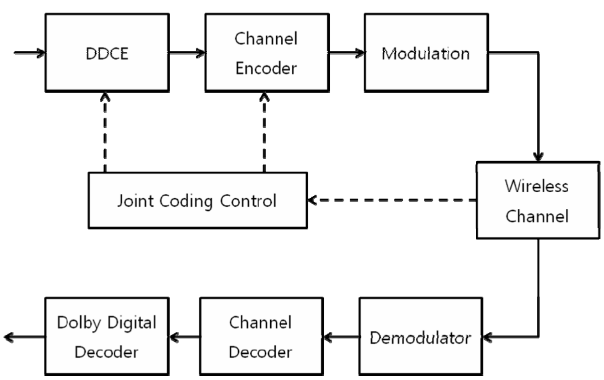

Fig. 3. Functional block diagram of wireless communication system for multichannel audio streaming

H. Ghasravi and K. Ban used ad hoc routing protocol control messages to optimize multimedia data [5]. In ad hoc routing protocols such as ad hoc on-demand distance vector (AODV) and dynamic source routing (DSR), each node maintains a routing table for an entry (destination) with a hop count (number of hops from source to destination) and a sequence number. This information can be used by the application to control the transmission rate in accordance with the hop count. In the case of AODV, the hop count can be extracted from the routing table information. If a route change is the consequence of a link breakage, any intermediate node (between the source and the destination) detecting the link breakage (to the next hop) will send the route error (RERR) message back to the source node. Therefore, the source node may use the reception of RERR as an indication of a link breakage. As soon as a new route is established, the application layer, upon receiving the hop-count information from the routing layer, can adjust its bit rates in accordance with the permissible transmission rate. In the case of video communications, the bit rates can be adjusted by changing the value of the quantization parameter (QP). This parameter has been specifically defined in the syntax structure by all video coding standards as a means to control the video transmission rate. Its value, which can have a direct bearing on the video quality, is selected as a two-way compromise between the average transmission rate and the video quality. Here, we have considered a new video-coding standard known as H.264 AVC.

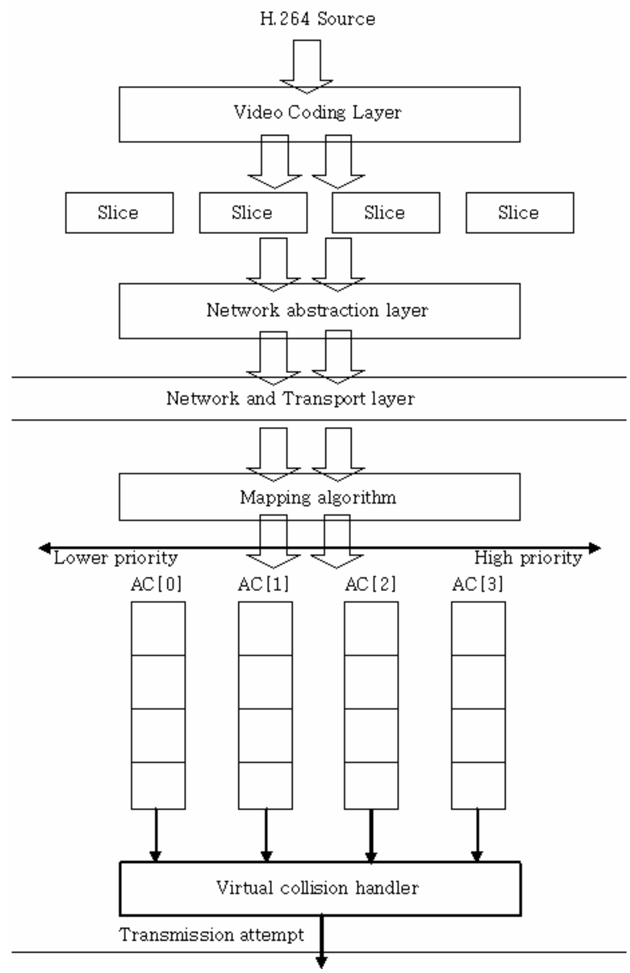

Fig. 4. Cross-layer architecture for H.264 video transmission

Jelena Kovacevic, et al. proposed an optimization algorithm in a source-channel domain [6]. Since the early days of digital communications, channel coding and modulation have been separated from source coding. According to information obtained from theoretical results, optimal source and channel coding can be performed separately for a stationary additive white Gaussian noise channel (AWGN) when no latency constraint is imposed. The result is known as the separation theorem. Audio and video streaming (including broadcast and unicast services) over time-varying wireless channels and finite block data lengths (implicitly latency constrained) cannot be optimally transmitted when the source and the channel coding are separated. Fig. 3 denotes a functional block diagram for multichannel audio streaming.

Adlen Ksentini, et al. proposed a cross-layer architecture based on two main interactions with 802.11e [7]. First, a top-down cross-layer interaction allows the H.264 network adaptation layer (NAL) video delivery module to transmit the QoS information related to the video fragment priority to the network layer. Second, a second top-down cross-layer interaction allows the network layer, in turn, to express the same QoS exigencies to an EDCA-based MAC layer. Fig. 4 depicts the cross-layer architecture for improving H.264 video transmission.

V. Hop-Based Priority

The previous studies that we have explained cannot satisfy the requirements of multimedia streaming. WLANs using 802.11 communicate via a shared medium. This characteristic of 802.11 requires an ad hoc wireless network with 802.11 to have a contention between the previous and next hops. This leads to delay fluctuation and additional back-off time due to collision. These, in turn, decrease the network bandwidth. The HBP technique using 802.11e EDCF is used for minimizing the contention between hops.

A previous packet or an early sent packet contends with a next packet or a later sent packet when the packets try to transmit to a destination through a route. The previous packet and the next packet have the same priority and contend with each other by a fair condition. It brings the probability of the gain of each packet in the channel to 50%. The effect that the next packet cannot get the channel by contention with the previous packet is small. If the next packet wins the previous packet and transmits one more hop, it cannot transmit more hops until the previous packet transmits. In other words, winning of the previous packet from the next packet causes the delay to decrease and the bandwidth to increase. Therefore, the previous packet has a higher priority than the next packet every time.

We propose an HBP technique to avoid the contention between the previous and the next packets. In other techniques, every packet has a fixed priority, and the previous packet and the next packet contend with each other. In HBP, each packet increases the priority after every hop and minimizes the contention between the packets. In other words, a priority is assigned to each hop, and a packet is assigned a higher priority after each hop.

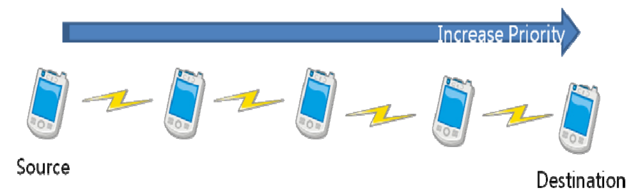

Fig. 5. Hop-by-hop increase in priority

802.11e EDCF can control the channel gain on the basis of priority; therefore, the HBP technique uses 802.11e EDCF. 802.11e EDCF supports eight priorities, which implies that the HBP technique has a limit of eight hops.

This technique has another limitation. This limitation is related to the mesh network. When there are many sources and destinations, starvation by priority can occur in nodes that are located in the route cross. The route that is far away from the source node has a higher priority than the other routes.

However, the limitation of priority distinction affects few people with a home network. A home network is composed of a home server and home appliances. The home server saves data and serves data to home appliances. Most of the stream services are served by the home server to the home appliances. Most of the data toward the home server do not have a time constraint. Therefore, the stream data in the home network are suitable for the HBP technique.

VI. Analysis and Simulation

A. Analysis

We analyze the delay in 802.11 WLAN. The delay in 1-hop transmission is denoted as

$$T_{trans} = T_{bas} + T_{cont} + T_{col} \tag{1}$$

The parameter T_{trans} denotes the time of 1-hop transmission. T_{bas} is the basic transmission time. T_{cont} and T_{col} are the additional delay caused by the loosening from contention and collision.

T_{bas} is the total sum of the DCF interframe space (difs, T_{difs}), random backoff (T_{ran}), transmission data (T_{data}), short interframe space (sifs, T_{sifs}), and transmission acknowledgment (T_{ack}) time.

$$T_{bas} = T_{difs} + T_{ran} + T_{data} + T_{sifs} + T_{ack} \tag{2}$$

T_{ran} is based on the size of the contention windows (CWs) and can be calculated.

$$Tran = (CW\max + CW\min) * slot\_time / 2 \tag{3}$$

T_{cont} has a probability related to the loosening for contention (P_{loose}).

$$T_{cont} = P_{loose} \times \left(T_{difs} + T_{data} + T_{sifs} + T_{ack}\right) \quad (4)$$

The last parameter is T_{col}. When a collision occurs, 802.11 operates the random backoff and attempts to transmit the data. T_{col} is denoted as

$$T_{col} = \sum_{i=1}^{n\_retry} P_{col\_i} \times \left[\begin{array}{l} T_{difs} + T_{ran\_i} \\ +P_{loose\_i} \times \left(T_{difs} + T_{data} + T_{sifs} + T_{ack}\right) \end{array}\right] \quad (5)$$

P_{col_i} is the probability of collision at the i^{th} attempt. Similarly, T_{ran_i} and P_{loose_i} are the random backoff time and the probability of loss at the i^{th} attempt. n_retry is the number of retry limits, generally 3.

The total delay can be calculated by T_{tran}*(number of hops). In 802.11e with HBP, the total delay has to be calculated because the priority changes at every hop. We have calculated the total delay related to HBP by considering each 1-hop delay and summarized it.

TABLE I
TIME OF DIFS, SIFS, SLOT, AIFS, AND CW SIZE IN 802.11B AND 802.11E

Variables	802.11b	802.11e			
		Pri 0	Pri 1	Pri 2	Pri 3
difs	50 μs				
sifs	10 μs				
Slot time	20 μs				
aifs	NA	40 μs	80 μs	140 μs	140 μs
CW_{min}	31	7	10	15	31

In 802.11e, there is one more variable, time of arbitration interframe space (aifs), and it can be used in difs. Each variable is different in 802.11, 802.11a, and 802.11b. We use 802.11b to evaluate HBP. Values of variables that we simulate are presented in Table I.

TABLE II
TRANSMISSION TIME AND BANDWIDTH OF 802.11B AND 802.11E

Time	802.11b	802.11e			
		hop 1	hop 2	hop 3	hop 4
1-hop trans. time	1135 μs	838 μs	764 μs	713 μs	660 μs
Total trans. time	4540 μs	2975 μs			
Bandwidth	965 Kbps	1.377 Mbps			

Table II denotes the 1-hop transmission time, total (4 hops) transmission time, and the estimated bandwidth in 802.11b and 802.11e when the data size is 512 bytes with an 11-Mbps link.

The HBP technique using 802.11e has a shorter delay and more bandwidth than that using 802.11b.

B. Simulation

We use NS2 version 2.33 [8] to simulate the HBP technique. In order to support 802.11e, a TKN EDCA patch [9] is used. There are five nodes on the same line, and each node can communicate with the neighboring node. The node at one end sends data to the node at the other side with constant bit rates.

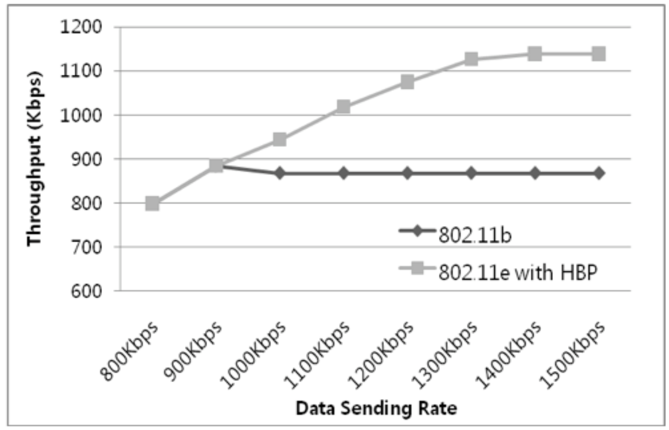

Fig. 6. Throughput of 802.11b and 802.11e with HBP from 800 Kbps to 1500 Kbps

Fig. 6 shows the throughput of 802.11b and 802.11e with HBP from 800 Kbps to 1.5 Mbps. The lower bit rate result is meaningless because the network is not saturated and very little contention occurs. The network is saturated at approximately 900 Kbps in 802.11b, and the throughput is steady after 900 Kbps. However, in the case of 802.11e with HBP, the throughput is increased until 1400 Kbps. Finally, 802.11b's throughput is 865 Kbps, and the throughput of 802.11e with HBP is 1138 Kbps. This is the reason for a smaller bandwidth than that of our model since our model does not consider the necessary operations of MAC, such as HELLO message and ARP. The MAC operation causes a contention of the channel and decreases the throughput.

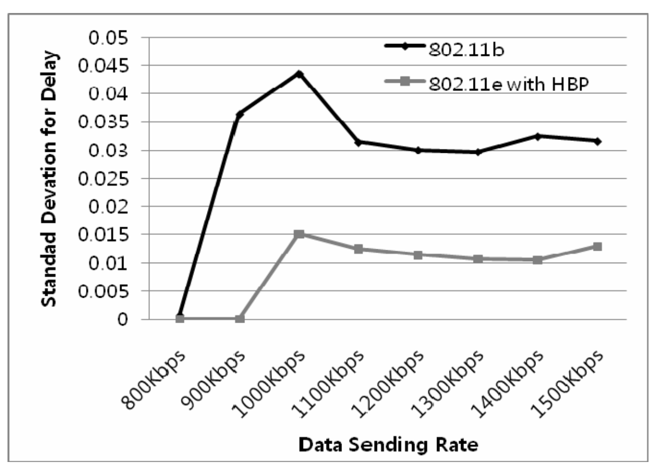

Fig. 7. Standard deviation of delay of 802.11b and 802.11e with HBP from 800 Kbps to 1500 Kbps

Fig. 7 shows the standard deviation of the delay. When the data sending rate is lower than 800 Kbps, the delay variation is low because there is little contention during the transmission like throughput. With a higher rate than 900 Kbps, a contention occurs and the delay variation increases in the case of 802.11b. However, we can see a low standard deviation of delay in 802.11e with HBP because of the minimized contention. This implies that the delay variation is low and less buffering is required.

VII. CONCLUSION

We propose an HBP technique using 802.11e for wireless multihop networks. Analysis and simulation results show that the HBP technique using 802.11e performs 31% better than that using 802.11b. In addition, the inter-packet delay is constant in HBP.

The HBP technique using 802.11e can be used in a wireless multihop network but not in an ad hoc mesh network. When many routes randomly exist, some routes have a lower priority than other routes at the cross-route node and cannot obtain the channel. It is difficult to solve this problem using 802.11e, and we intend to find a solution to this problem in the future.

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BIOGRAPHIES

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