Improving VoIP Capacity in Multi-Rate IEEE 802.11e WLANs via Relay

Jeonggyun Yu,[1] Munhwan Choi,[2] Sunghyun Choi,[2] and Daji Qiao[3]
[1]Samsung Electronics Co., LTD, Suwon, Korea
[2]School of Electrical Engineering and INMC, Seoul National University, Seoul 151-744, Korea
[3]Department of Electrical and Computer Engineering, Iowa State University, Ames, IA 50011, USA
Email: jeonggyun.yu@samsung.com, mhchoi@mwnl.snu.ac.kr, schoi@snu.ac.kr, daji@iastate.edu

Abstract - ***In order to improve the Voice over IP (VoIP) capacity of a Wireless Local Area Network (WLAN), we develop a relay-aided solution, which includes a relay-aided admission control scheme, a packet aggregation scheme, and a protocol for packet relay without sacrificing the voice quality. Basically, our admission control scheme utilizes relay stations to admit as many VoIP sessions as possible. When a new VoIP session requests admission, the admission controller checks whether it can be admitted by either using relay or communicating with the AP directly. Moreover, to overcome the overhead increased due to relaying small-size VoIP packets, we employ a packet aggregation scheme for VoIP packet transmissions between the AP and relay stations. Simulation results demonstrate that the proposed relay-aided VoIP provisioning scheme can significantly improve the VoIP capacity without compromising the service quality of the admitted VoIP sessions.***

I. Introduction

Voice over IP (VoIP) is one of the fastest growing applications over the Internet today. At the same time, the deployment of the Wireless Local Area Network (WLAN) based on the IEEE 802.11 standard [1] is fast increasing. Naturally, VoIP over WLAN is emerging as a very attractive and popular application. However, it is critical that the VoIP capacity of a WLAN is very limited due to various inherent header and protocol overheads of the 802.11 Medium Access Control (MAC). Moreover, in a multi-rate WLAN,[1] low-rate stations can severely decrease the VoIP capacity because they cause long channel occupancy time for their transmission attempts [2]. In a given network topology, an effective method to increase the network capacity is to increase transmission rates of low-rate stations. Therefore, using relays could be a good approach to improve the VoIP capacity of a WLAN.

So far, existing relay schemes such as rDCF [3], [4] and RAMA [4] have mainly focused on throughput enhancement in ad-hoc networks when the packet size is large (i.e, larger than or equal to 1000 bytes). Note that our interest is to increase the VoIP capacity in an infrastructure-based WLAN. Moreover, both rDCF and RAMA mandate the use of RTS/CTS handshake before the sender starts a data packet transmission, which makes them not appropriate for VoIP services due to the large RTS/CTS overhead compared to the small size of voice packets.

[1] The 802.11 physical (PHY) layers support multiple transmission rates. For example, the 802.11b PHY supports rates of 1, 2, 5.5 and 11 Mbps.

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In order to improve the VoIP capacity of a WLAN, we develop a relay-aided solution, which includes a relay-aided admission control scheme, a packet aggregation scheme, and a protocol for packet relay. Basically, when a new VoIP session requests its admission, our admission controller located at the AP checks whether it can be admitted by either using relay (i.e., two-hop architecture) or communicating with the AP directly (i.e., one-hop architecture). With the option of using a relay, a station may now be admitted to communicate with the AP via a relay station though it cannot be admitted under the conventional one-hop architecture due to its low transmission rate. Moreover, to overcome the overhead increased due to relaying small-size VoIP packets, we employ a packet aggregation scheme for VoIP packet transmissions between the AP and relay stations. In order to consider the Quality of Service (QoS) requirement of each VoIP session, our solution has an admission control scheme which is based on an IEEE 802.11e Enhanced Distributed Channel Access (EDCA) admission control scheme proposed in our previous work in [5]. Different from the existing schemes [3], [4], our goal is to design an effective admission control scheme that considers the option of using relays, so as to admit as many new VoIP sessions as possible while providing the guaranteed QoS for the admitted VoIP sessions.

The rest of the paper is organized as follows. Section II presents the system model, and the proposed relay-aided VoIP provisioning scheme is described in Section III. Section IV presents the simulation results, and the paper concludes in Section V.

II. SYSTEM MODEL

We consider an infrastructure-based WLAN based on the IEEE 802.11e EDCA. It consists of an AP and multiple stations communicating with the AP. The AP is wire-connected to the Internet and serves as the gateway for all the stations in the WLAN. All the VoIP stations are stationary and capable of using multiple transmission rates. An example scenario is to support VoIP services in an office room where the VoIP phones are mounted at fixed positions such as desks. All the VoIP stations are always associated with the AP but not every one of them always has an active voice session at a given time. We assume that the transmission power is always fixed. The wireless channel between a sender and a receiver is assumed to be symmetric. Moreover, we assume that all the stations can sense the transmissions of each other (i.e., there is no hidden station).

All the VoIP stations associated with the AP are able to relay other VoIP stations' traffic. Therefore, voice packets may be delivered via one or two hops between the AP and a VoIP station. Although it is possible to have an n-hop ($n > 2$) relay architecture, the incurred control overheads (e.g., duplicated packet transmissions, coordination among stations, etc.) make it less attractive and less practical. Thus, we focus on the two-hop MAC-layer relay in this paper.

III. RELAY-AIDED VOIP SUPPORT

The basic Traffic Stream (TS) setup procedure and the admission control framework in the IEEE 802.11e standard are used in our scheme. When a VoIP station requests the admission of its VoIP session to the AP, the AP evaluates whether the available bandwidth can accommodate the newly-requested VoIP session via either a two-hop relay or the one-hop direction communication. If the AP can accommodate the VoIP session with either option, it is admitted; otherwise, it is rejected.

A. Collecting Rate Information

In order to form a relay topology, the AP needs to have the transmission rate information of all the VoIP stations including itself. Therefore, each station needs to report the transmission rates between itself and its neighboring stations including the AP. To estimate the transmission rates in a similar manner to the existing work in [3], [4], [6], we employ a receiver-based channel condition measurement. When a station overhears a packet, it measures the Signal-to-Noise Ratio (SNR) and extracts the sender's MAC address from the packet if the packet can be decoded correctly. Then, it selects the proper transmission rate based on the measured SNR of the packet. Finally, the station caches the sender's MAC address and the estimated transmission rate in a table. The AP periodically collect the estimated transmission rates of each VoIP station to neighboring stations.

B. Relay Setup

When a VoIP station requests the admission of its VoIP session to the AP, the station might serve as a relay station for other stations or use one of the other stations as its relay station based on the transmission rates between the station and the AP. There are two cases:

- If the transmission rate between the station and the AP is a low rate such as 1 or 2 Mbps of the 802.11b PHY, it may use one of the other stations as its relay station instead of serving as a relay for other stations with active VoIP sessions.
- Otherwise, it may serve as a relay station for other stations with active VoIP sessions.

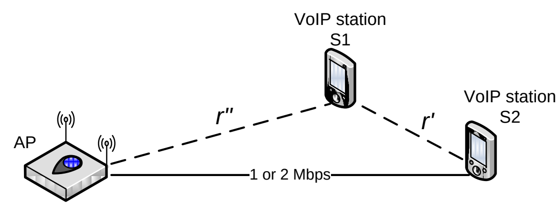


Figure 1: An example relay topology.

Table I

PRIORITY LEVELS OF RELAY SELECTION

	Priority Condition		
	r' (Mbps)	r'' (Mbps)	Relay
Priority Level 1	11	11	*On*
Priority Level 2	11	5.5	*On*
Priority Level 3	5.5	11	*On*
Priority Level 4	5.5	5.5	*On*
Priority Level 5	11	11	*Off*
Priority Level 6	5.5	11	*Off*
Priority Level 7	11	5.5	*Off*
Priority Level 8	5.5	5.5	*Off*

Station S2 in Fig. 1 is an example admission-requesting station corresponding to the first case. Using the transmission rate table, the AP tries to select a proper relay station among all candidate stations for S2, i.e., all stations with active VoIP sessions. For example, S1 could be a candidate station if it carries an active VoIP session. Table I lists the eight priority levels for relay selection, assuming that all candidate stations are equipped with the 802.11b PHY. As shown in the table, the priority levels are decided by three factors: (1) r' – the transmission rate between S2 and the candidate station; (2) r'' – the transmission rate between the candidate station and the AP; and (3) whether the candidate station is currently relaying for other stations: *"On"* if so and *"Off"* if not. In general, a candidate station with higher r' and r'' has higher priority than others since the channel occupation time is smaller. Moreover, a candidate station which is already relaying for others may be a more appropriate choice as the packets of the new VoIP session may be aggregated together with packets from others, thus reducing the channel occupation time for relaying these packets. If no relay station can be found for S2 that belongs to any one of the eight priority levels, the AP assumes that the only possible way of admitting the VoIP session of S2 is via the one-hop direct communication with the AP.

Station S1 in Fig. 1 is an example admission-requesting station that corresponds to the second case. In this case, the

AP uses a table similar to Table I to find all the low-rate stations that may use S1 as the relay station.

After the admission decision of a new VoIP session has been made, the corresponding admission-requesting station will be informed of the decision. If the new VoIP session is admitted, the AP adds an entry to its relay table and informs the admission-requesting station and the relay station of the relay topology.

C. Relay Transmission

Apparently, the transmission rates of low-rate stations can be increased indirectly by using relay. However, the overheads (i.e, MAC header, PHY, ACK transmission, and contention overheads) also increase due to duplicated transmissions by relay stations. Moreover, the portion of the overheads can be large due to the small size of voice packets.

A popular technique to reduce these overheads is packet aggregation. The basic idea of packet aggregation is to combine several small packets together at the AP and relay stations, and then forward them with one MAC and PHY header. In the IEEE 802.11n standard [7], two packet aggregation schemes (i.e., Aggregate MAC Service Data Unit (A-MSDU) and Aggregate MAC Protocol Data Unit (A-MPDU)) are specified. In our solution, we use packet aggregation at the MSDU level (i.e., A-MSDU) which performs particularly well for small-size packets such as voice packets.

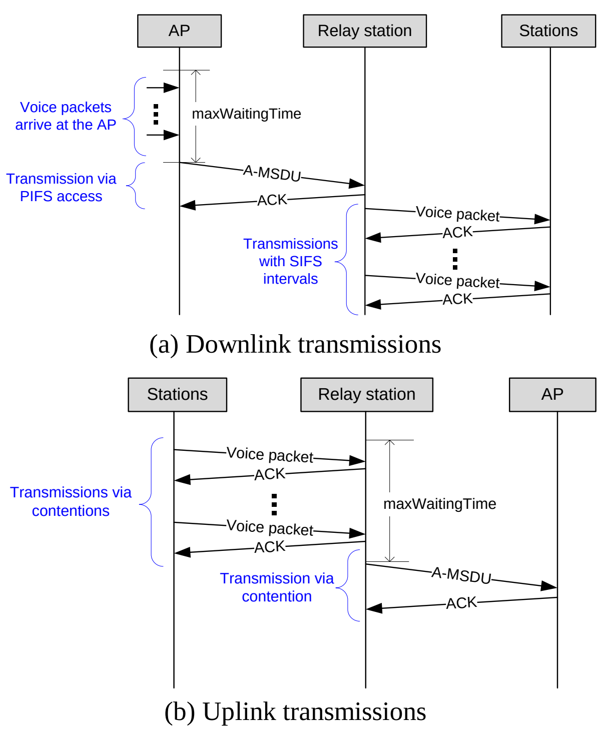


Figure 2: Transmission procedures using relay.

As shown in Fig. 2, in order to mitigate the bottleneck effect of the AP that limits the VoIP capacity [8], the AP transmits the downlink VoIP packets, including the aggregated frame, by using a simple contention-free access called Point Coordination Function Interframe Space (PIFS) access, which allows the AP to transmit a pending voice packet after a PIFS idle time without any contention (i.e., backoff). The AP aggregates the downlink packets that it has received during *maxWaitingTime* for the same relay station into one frame and transmits it to the corresponding relay station. The relay station in turn forwards the voice packets contained in the received aggregated frame to their destination VoIP stations with SIFS (Short Interframe Space) time intervals between an ACK reception and the next voice packet transmission. For uplink transmissions, relay stations and VoIP stations transmit their packets via contention to the AP and relay stations, respectively. Each relay station also aggregates the voice packets received from all the stations in its relay list during *maxWaitingTime* and transmits an A-MSDU frame to the AP. In this paper, we set *maxWaitingTime* to be the same as the packetization interval T_v of the VoIP session[1] to minimize the delay of the relay station's own voice packets.

D. Admission Decision

Packet aggregation can reduce the amount of overheads. However, it also increases the packet delay, which may reduce its suitability for VoIP services. To deal with this issue, the AP in our system aggregates and transmits an A-MSDU frame to each relay station every voice packetization interval T_v. Moreover, each relay station also aggregates and transmits an A-MSDU frame to the AP every T_v interval. As a result, in the worst-case scenario, the delay of uplink/downlink VoIP packets via relay stations and the delay of VoIP packets exchanged directly with the AP are $2T_v$ and T_v, respectively. Therefore, depending on the maximum tolerable wireless delay of the new admission-requesting VoIP session, which can be calculated using the information such as the wireline delay between the remote voice gateway and the AP and the maximum mouth-to-ear (m2e) delay that corresponds to the R value of 80,[2] the AP acts differently:

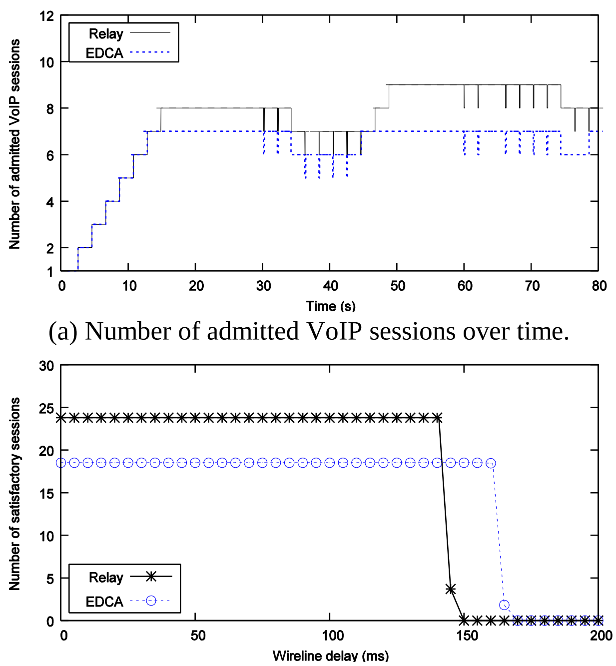
- If the maximum tolerable wireless delay is less than T_v, the AP denies the request since it cannot guarantee that the delay of voice packets of the new VoIP session is less than T_v due to packet aggregation;
- If the maximum tolerable wireless delay is between T_v and $2T_v$, the AP makes the decision based on whether the available bandwidth can accommodate the one-hop direct communication between the station and the AP, i.e., the option of using relay is not considered;
- If the maximum tolerable wireless delay is greater than $2T_v$, the AP checks all possible two-hop relay options as well as the one-hop direct communication, and then makes the admission decision.

IV. PERFORMANCE EVALUATION

[1] We assume that the VoIP traffic is generated with Constant Bit Rate (CBR).

[2] In this paper, we use the ITU E-model [9] to assess the quality of mouth-to-ear (m2e) voice communication. It gives an overall rating R to the quality of a phone call where $0 \le R \le 100$. Acceptable user satisfaction levels correspond to R values larger than or equal to 80.

We evaluate the effectiveness of the proposed scheme using the ns-2 simulator [10]. Multiple static VoIP stations are placed inside a square region with 160 meters on the diagonal. VoIP stations communicate with the remote voice gateway via the AP which sits at the center of the square region. VoIP stations only transmit and receive voice packets. Each station carries a single traffic flow. The IEEE 802.11b PHY is used in our simulation. We assume an AWGN (Additive White Gaussian Noise) wireless channel and the background noise level is set to -96 dBm. VoIP traffic is modeled as a two-way CBR session with 208 byte MSDU size and 20 ms packetization interval (i.e., T_v = 20 ms) which is set according to the G.711 voice codec [11].



(a) Number of admitted VoIP sessions over time.

(b) Total number of satisfactory VoIP sessions during simulation.

Figure 3: Comparison of relay-aided and EDCA-based solutions.

We simulate 40 VoIP sessions (i.e., 40 VoIP stations) which start successively every 2 seconds from the beginning of a simulation. The call duration of each VoIP session is 30 seconds. We simulate 50 random scenarios with different positions of VoIP stations. The total simulation time is 100 seconds for each random scenario. Simulation results are shown in Fig. 3 where each point is averaged over 50 random scenarios. We compare two schemes: the Relay scheme and the EDCA scheme. The Relay scheme is the one proposed in this paper, while the EDCA scheme is proposed in our previous work in [5] which is the same as the Relay scheme except that it only considers one-hop direct communications with the AP when making admission decisions.

Fig. 3(a) shows that the Relay scheme can serve more VoIP sessions than the EDCA scheme due to the gain from relay usage. Fig. 3(b) compares the two schemes in terms of the total number of satisfactory VoIP sessions during the 100-second simulation. An VoIP session with $R \geq 80$ is called a *satisfactory VoIP session*. In Fig. 3(b), the x-axis is the wireline delay, which is the delay that voice packets experience in the wired network between the remote voice gateway and the AP. As the wireline delay increases, a shorter wireless delay is desired so that the overall end-to-end delay is small enough to guarantee a satisfactory VoIP quality. As shown in the figure, the Relay scheme outperforms the EDCA scheme under most scenarios for a wide range of wireline delays between 0 and 140 ms. However, as the wireline delay surpasses 140 ms, the number of satisfactory sessions with the Relay scheme drops drastically while the number of satisfactory sessions with the EDCA scheme remains high until the wireline delay increases to greater than 160 ms. This is due to the additional aggregation delay introduced in the Relay scheme, i.e., *maxWaitingTime* for packet aggregation in the AP and relay station.

IV. CONCLUSION

In this paper, we propose a relay-aided VoIP provisioning scheme to increase the capacity of a WLAN in terms of the total number of admitted VoIP sessions with satisfactory service quality. Our admission control scheme utilizes high-rate VoIP stations as relay stations for low-rate VoIP stations to admit as many VoIP sessions as possible. Moreover, we employ a packet aggregation scheme to ameliorate the increased overheads caused by duplicated packet transmissions by the relay stations. Simulation results demonstrate that the proposed relay-aided scheme improves the VoIP capacity significantly without compromising the service quality of admitted VoIP sessions.

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