Using Dynamic PCF to Improve the Capacity for VoIP Traffic in IEEE 802.11 Networks

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Abstract— **With the deployment of IEEE 802.11 networks, supporting real-time traffic with stringent Quality of Service (QoS) requirements on these networks becomes a critical issue. In this paper we propose two new media access schemes, namely Dynamic Point Coordination Function (DPCF) and modified DPCF (DPCF2). These can improve the capacity for Voice over IP (VoIP) traffic by up to 20% in IEEE 802.11b networks. We will show how we can also achieve a drastic improvement in the end-to-end delay with mixed VoIP and data traffic. Delay is kept around 100 ms in heavily loaded traffic conditions with an average value under 60 ms in normal traffic conditions.**

I. Introduction

Internet telephony using Voice over IP (VoIP) technology is defined as the transport of telephone calls over the Internet. As many VoIP clients for mobile devices such as PDAs are becoming available, VoIP in IEEE 802.11 networks [1] will spread quickly. Because of packet header overhead and other factors, the capacity of IEEE 802.11 networks for VoIP is far below the nominal bit rate. We anticipate that public spaces such as airports, train stations and stadiums will need to support a large number of concurrent voice conversations, but increasing the capacity by adding APs is difficult since there are only three usable non-overlapping channels for IEEE 802.11b/g [2], [3]. These needs and constraints motivate us to investigate increasing the capacity in IEEE 802.11 networks for VoIP calls.

This paper is organized as follows: In Section II-A we give a general overview of the IEEE 802.11 and the IEEE 802.11e MAC protocols; in Section III we analyze the capacity when using DCF and PCF; in Section IV we describe our new algorithms; in Sections V and VI we describe our experiments and show our results; in Section VII we compare our approach with other approaches. Section VIII concludes the paper.

II. Background

In this section, we will give an overview of the IEEE 802.11 and the IEEE 802.11e MAC protocols.

A. IEEE 802.11 MAC Protocol

The IEEE 802.11 standard provides two different channel access mechanisms, namely the Distributed Coordination Function (DCF) and Point Coordination Function (PCF). Our new scheme will introduce enhancements to the PCF access scheme.

The PCF is based on a polling scheme. Each STA in the polling list is polled in turn. The Point Coordinator (PC) sends a CF-Poll frame to each pollable station (STA), i.e., each STA that can respond to a CF-Poll in the polling list. The STA responds by sending a Data frame if it has data to send or a Null packet if it has no data to send at that time.

Piggybacking is commonly used. If the PC has some data to send to a particular pollable STA, a Data + CF-Poll frame will be sent to this STA when its turn to be polled arrives and the STA will respond with a Data + CF-Ack frame if it has data to send or with CF-Ack (no data) if it does not have any data to send at that time.

In an infrastructure network, the AP acts as the PC. The PC will gain access to the medium with a higher priority than other STAs using DCF. When a PC is operating, the two access methods alternate, with a Contention Free Period (CFP) followed by a Contention Period (CP). The PCF controls frame transfers during a CFP, while the DCF controls frame transfers during a CP.

B. IEEE 802.11e MAC Enhancements

To support applications with Quality of Service (QoS) requirements on IEEE 802.11 networks, the IEEE 802.11e standard is currently under development [4]. It introduces the concept of Hybrid Coordination Function (HCF) for the MAC mechanism. HCF is backward compatible with DCF and PCF, and it provides QoS STAs with prioritized and parameterized QoS access to the wireless medium. The HCF uses both a contention-based channel access method, called the Enhanced Distributed Channel Access (EDCA) and a contention-free channel access method, called HCF Controlled Channel Access (HCCA). With the EDCA, QoS is supported by using four access categories (ACs), each corresponding to an individual prioritized output queue. A traffic class which requires lower transmission delay can use an AC with higher priority in its contention for the channel. With the HCCA, a hybrid coordinator (HC) allocates transmission opportunities (TXOPs) to wireless STAs by polling, so as to allow them contention-free transfers of data, based on QoS policies. An HC can generate an alternation of CFP and CP.

TABLE I

PARAMETERS IN IEEE 802.11B (11 MB/S)

Parameters	Time (μs)	Size (bytes)
PLCP[1] Preamble	144.00	18
PLCP Header	48.00	6
PLCP Header Service	192.00	24
MAC Header+CRC	24.73	36
IP+UDP+RTP	29.09	40
Voice	58.18	160
ACK	10.18	14
SIFS	10.00	
DIFS	50.00	
Slot	20.00	
CW_{MIN}	31 slots	

III. ANALYSIS OF VOIP CAPACITY

In this section, we analyze the capacity of Constant Bit Rate (CBR) VoIP numerically, as this capacity is an upper bound on the network capacity. Here, the capacity is the maximum number of calls that are allowed simultaneously for a certain channel bit rate; we assume that all the voice communications are full duplex (Refer to Section VI for the analysis results).

A. VoIP Capacity of DCF

A CBR VoIP client generates one VoIP packet every packetization interval. The packets should be transferred right after they are generated to avoid delay. Therefore, the number of packets that can be sent during one packetization interval is the maximum number of calls, and we can calculate the capacity of VoIP using the following equation: $N_{max} = T_p/(2T_t)$, where N_{max} is the maximum number of calls, T_p is the packetization interval, and T_t is the time for sending one voice packet. The reason for T_t being multiplied by 2 is that the voice communication is full duplex. To send one voice packet, the Distributed Interframe Space (DIFS), Short Interframe Space (SIFS), acknowledgment (ACK) and its own backoff time are required. So, we can get T_t using the following equation.

$$T_t = T_{DIFS} + T_{SIFS} + T_v + T_{ACK} + T_b$$

Here, T_v and T_{ACK} are the time for sending a voice packet and ACK, respectively, T_b is the backoff time, T_{DIFS} and T_{SIFS} are the lengths of DIFS and SIFS, respectively. The backoff time is Number of Backoff Slots $* T_{Slot}$ where T_{Slot} is a slot time, and Number of Backoff Slots has a uniform distribution over $(0, CW_{MIN})$ with an average of $(T_{Slot} * CW_{MIN}/2)$. Therefore, the capacity of VoIP can be expressed as:

$$N_{max} = \frac{T_p}{2(T_{DIFS}+T_{SIFS}+T_v+T_{ACK}+(T_{Slot}\times CW_{MIN}/2))}$$

In our analysis, a G.711 generates 160 byte packets every 20 ms. The other parameters are taken directly from the IEEE 802.11b standard. All the parameters used in our analysis are shown in the Table I.

B. VoIP Capacity of PCF

To avoid delay in PCF mode, VoIP STAs need to be polled every packetization interval, meaning that the CFP interval should be less than or equal to the packetization interval. So, we need to know how many STAs can be polled in one CFP. We can calculate the capacity of VoIP in PCF mode using the following equation.

$$N_{max} = (T_{CFP} - T_B - T_{CE} - T_{CP})/(2T_t)$$

where T_{CFP} is a CFP duration, T_B is the time needed for sending a Beacon frame, T_{CE} is the time for sending a CF-End frame and T_{CP} is a CP duration. The time required for sending one voice packet T_t can be obtained using the following equation, since we just need to turn on a bit in the voice packet to notify CF-Poll or CF-Ack: $T_t = T_v + T_{SIFS}$.

Therefore the final equation for calculating the capacity of VoIP is the following.

$$N_{max} = \frac{T_{CFP}-T_B-T_{CE}-T_{CP}}{2(T_v+T_{SIFS})}$$

To compute the capacity in Fig. 5, we used IEEE 802.11b as wireless network, 20 ms as CFP interval, and 1 ms as CP. The other parameters are listed in Table I.

IV. DYNAMIC POINT COORDINATION FUNCTION (DPCF)

While PCF is proposed to support real time traffic, it is not implemented in most of the wireless cards on the market because it has a lot of disadvantages. Here, we propose two modified PCF schemes, Dynamic Point Coordination Function (DPCF) and modified DPCF (DPCF2) that overcome the disadvantages of PCF without changing the standard.

A. Differentiation of Traffic Types

Even though PCF is meant to support real time traffic, it cannot guarantee QoS. So, in DPCF, we classify the traffic into two classes, VoIP traffic and best effort, to support QoS. Usually, VoIP packets are sent during CFP, but they can be sent also during CP if there are some pending packets. However, non-VoIP packets must be sent only in CP. In this way, we can give higher priority to VoIP packets and we can reduce the delay due to the other traffic types.

B. Dynamic Polling List

The biggest disadvantage of PCF is that a lot of bandwidth is wasted by sending CF-Polls and Null packets when STAs have no packets to send. Usually, in normal conversations, when one person is talking, the other one listens without talking. So, one STA is sending packets and the other STA is not sending but just receiving packets (if silence suppression is used) and a lot of bandwidth is wasted with a lot of unnecessary CF-Polls and Null packets. This waste of bandwidth significantly reduces the capacity of VoIP in wireless network. To minimize the waste of bandwidth, we introduce a dynamic polling list. The dynamic polling list maintains the active nodes, i.e., those sending data. The PC can avoid polling non-active nodes which are not sending any data.

1) Removing an STA from the Polling List: The PC needs to remove an STA when it stops sending packets. When the AP gets a Null packet from an STA, the PC removes the STA from the polling list. However, we should not remove the STA immediately because a packet can be lost or delayed; as a heuristic we remove the STA after the AP gets three consecutive Null packets from the same STA.

[1]Physical Layer Convergence Protocol

2) Adding an STA to the Polling List: We considered two schemes for adding an STA to the polling list.

Scheme 1: When an STA starts to talk, it sends the first VoIP packet in CP. When the AP gets a VoIP packet in CP, it adds the STA into the polling list and the STA is polled starting from the next CFP. The problem of this method is that if CP is very congested, the first packet of a talk-spurt could be delayed until the next CP. However, even if the first packet is delayed, the next packets are sent without significant delay because of the More Data field in DPCF (See Section IV-C).

Scheme 2: Another approach for adding an STA into the polling list is using statistical data of VoIP traffic. If we can estimate the duration of the next silence period precisely, then the STA can be added into the polling list before it starts to send voice packets. Ziouva et al. [5] proposed a scheme where an STA can be added into the polling list after k CFP intervals, with 1 and 2 as k values. We also tried adding an STA into the polling list using a statistical approach. We used ITU-T P.59 [6] for statistical data and used 500 ms as threshold value for adding. That is, an STA is added to the polling list 500 ms after it is removed.

Scheme 2 has the following problem. Since the talk spurts are statistically independent, it is very difficult to estimate the duration of the next silence period precisely. If an STA is added into the polling list too early, then CF-Polls are wasted until it starts to send voice packets. If an STA is added too late, then the data packets should be sent in CP until the STA is added to the polling list after 500 ms. We confirmed that scheme 2 has worse performance than scheme 1 by simulation. Therefore, we applied scheme 1 to DPCF and DPCF2.

C. Dynamic CFP Interval and the More Data Field

In PCF, the CFP interval should be less than or equal to the packetization interval to deliver packets without delay. However, VoIP clients differ in their packetization intervals. When more than one packetization interval is used in a wireless network, the choice of the CFP interval affects the capacity and quality of VoIP. For example, let's say STA A uses 10 ms and STA B uses 20 ms packetization intervals. In this case, when 10 ms is used as CFP interval, STA B is polled twice per packetization interval so one CF-Poll is wasted every 20 ms. When 20 ms is used as CFP interval, STA A generates 2 packets and only one packet can be sent in a CFP interval (20 ms). The other packet should be sent in CP or in the next CFP interval. This creates significant delay and such a delay is accumulated. To solve this problem, we use the *More Data field* in the Frame Control field. This field is defined in IEEE 802.11 standard and used in power-save mode to indicate that at least one additional buffered MSDU is present for the same STA. We use the largest packetization interval as CFP interval. If an STA needs to send more than one packet, it sets the More Data field in the first packet. When the AP detects the More Data field in a data packet, the AP polls the STA again until the More Data field is not set any more. In the above example, STA A can send two packets per 20 ms instead of sending only one packet. The first packet is delayed 10 ms but the delay is never accumulated and CF-Polls are not wasted. See Fig. 1.

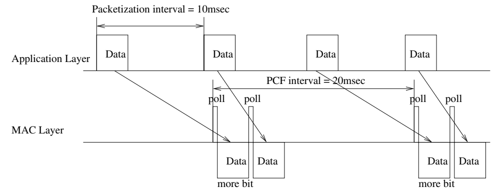

Fig. 1. Packet transfer in DPCF mode with 10 ms packetization interval

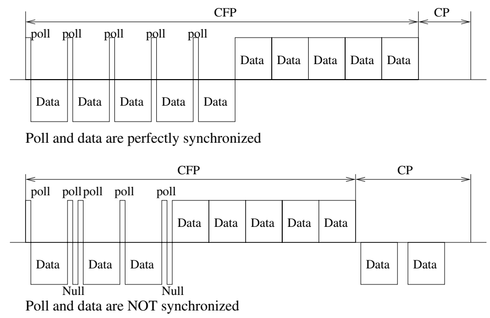

Fig. 2. Synchronization problem for PCF in an AP

D. DPCF2; Modifying to Avoid CFP-to-CP Migration

We cannot eliminate wasted CF-Polls and Null packets completely even if we use the dynamic polling list, a dynamic CFP interval and the More Data field. PCF has the following critical disadvantage. Theoretically, only one CF-Poll is required per packetization interval in VoIP, and the CF-Poll is not wasted. However, if the STA is polled before a VoIP packet is created, the CF-Poll is wasted and the data packet will be sent in CP or in the next CFP interval. If CP is not very congested, the packet will be sent in CP, the next CF-Poll will be wasted and the same problem will occur over and over. Eventually, most of the CF-polls are wasted and most of the packets are sent in CP. This is a synchronization problem between CF-Polls and data. If this synchronization problem happens in a lot of STAs, CFP is shortened and CP is increased. During CP there is enough time to send the packets missed in CFP, and the vicious cycle is repeated, as depicted in Fig. 2. The ideal solution for this problem is synchronizing CF-Polls and data. In reality, it is very difficult to synchronize CF-Polls and data because the PC cannot poll an STA with the exact same interval, and at the application layer, packets cannot be created at exactly the same rate. We proposed a practical solution, DPCF2, which prevents STAs from sending VoIP packets in CP when there is only one VoIP packet in their queue so that the packet can be sent in the next CFP. That is, VoIP packets can be sent in CP only when there is more than one packet in the queue. In this way, we can avoid wasting CF-Polls, and the number of packets sent in CP is decreased, optimizing the CP for the non-VoIP packets.

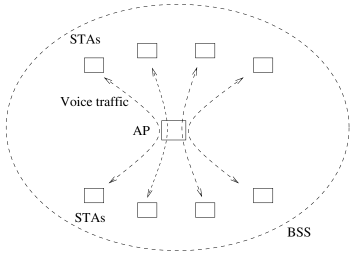

Fig. 3. Network topology

TABLE II
SIMULATION PARAMETERS

Transmission rate (Mb/s)	1	2	5.5, 11
RTS threshold (bytes)	1500	1500	1500
Retransmission limit	7	7	7
CFP interval (ms)	60	50	40
CFP duration (ms)	40	40	35

V. SIMULATION

Usually it is very difficult to precisely measure the capacity of VoIP in a real environment because we cannot completely eliminate physical obstacles, microwave interference and multi-path effects. In order to evaluate the capacity of VoIP, we implemented DPCF and DPCF2 in the QualNet simulator [7].

A. Network Topology

Fig. 3 shows the network model we used in this paper. We considered a Basic Service Set (BSS). We assumed that the BSS includes one AP and some wireless STAs which exchange VoIP packets via the AP.

B. Wireless Setup

In this paper, we adopted the system parameters shown in Table I, which are specified in the IEEE 802.11b standard [2]. We also set other parameters as shown in Table II. Note that we used 40 ms and 35 ms at 5.5 Mb/s and 11 Mb/s, 60 ms and 40 ms at 1 Mb/s, and 50 ms and 40 ms at 2 Mb/s as CFP interval and CFP duration, respectively, so that there is sufficient time to send at least one data frame during the CP.

C. Traffic Characteristics

We utilized for our simulation the G.711 codec with a payload of 160 bytes and a packetization interval of 20 ms. We considered VoIP traffic both without silence suppression (CBR) and with silence suppression (Variable Bit Rate: VBR). We used the conversational speech model described in ITU-T P.59 for VBR. In order to simplify our simulation, we removed the double talk in the model and used the mean values of 0.9 sec and 1.5 sec for exponentially distributed duration of talk-spurt and pause, respectively, so that the ratio of mutual silence period is the same as in ITU-T P.59. CBR sources send voice packets at the voice codec rate, while VBR sources send them at the voice codec rate during the talk-spurt and do not send them during the pause.

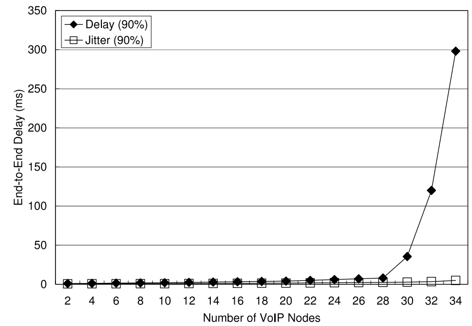

Fig. 4. End-to-end delay for VoIP at 11 Mb/s in DCF

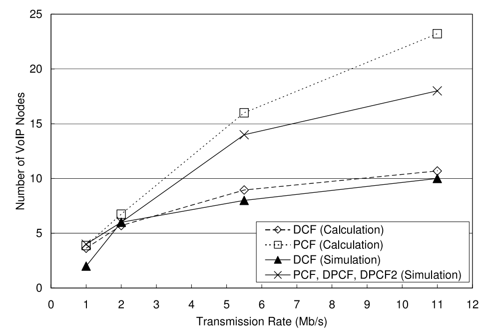

Fig. 5. Capacity of VoIP without silence suppression

Usually some wireless STAs in the BSS will use not only VoIP but also HTTP, FTP, E-mail etc. We also investigated the case when VoIP traffic and file transfer traffic coexist.

VI. RESULTS AND ANALYSIS

A. Capacity of VoIP

The one-way end-to-end delay of voice packets is supposed to be less than 150 ms [8]. We assumed the codec delay to be about 30-40 ms at both sender and receiver, and backbone network delay to be about 20 ms. Therefore, the wireless networks should contribute less than about 60 ms delay.

We measured the 90th percentile of the end-to-end delay of voice packets at each STA with a varying number of wireless STAs, and defined the capacity of VoIP as the maximum number of wireless STAs so that the average of the 90th percentile of the one-way end-to-end delay does not exceed 60 ms. We also measured the jitter of voice packets. In this paper, we defined jitter as the variation of the difference in two consecutive packet-receiving intervals. We observed no packet loss caused by path loss or expiration of frame retransmission limit in our simulation.

In Fig. 4, we plot the average of the 90th percentile of the end-to-end delay and jitter of VBR voice packets against the number of wireless VoIP STAs. Here, the bit rate is 11 Mb/s. We can see that the end-to-end delay increases slowly as the number of wireless STAs increases, and it increases sharply

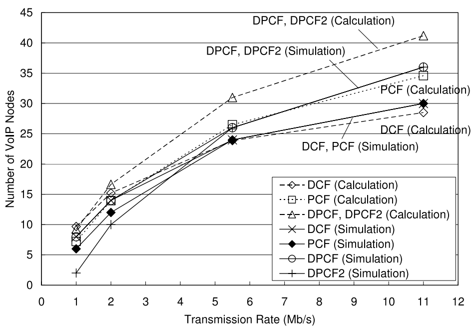

Fig. 6. Capacity of VoIP with silence suppression

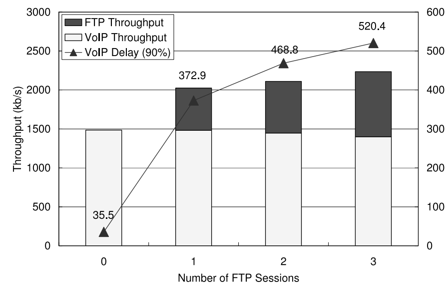

Fig. 7. End-to-end delay of VoIP and throughput of FTP and VoIP for DCF, 30 VoIP nodes

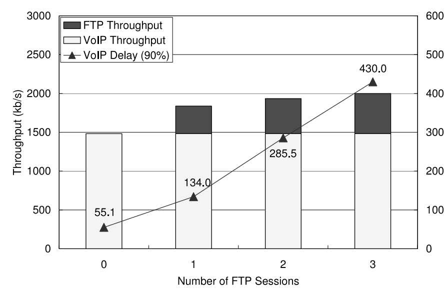

Fig. 8. End-to-end delay of VoIP and throughput of FTP and VoIP for PCF, 30 VoIP nodes

when the number of STAs exceeds some value. In this figure, the capacity for VoIP traffic is 30 calls.

Fig. 5 shows the capacity of VoIP without silence suppression (CBR) for different bit rates. At 11 Mb/s, the capacity in DCF is 10 calls, while in PCF, DPCF and DPCF2 it is 18 calls. With CBR traffic, DPCF and DPCF2 perform in the same way as PCF because STAs have no silence periods, i.e., DPCF and DPCF2 cannot save CF-Polls and Null packets. The capacity in PCF, DPCF and DPCF2 is larger than in DCF because packets will collide with each other more in the CP as the number of STAs increases, while using a polling mechanism will reduce the number of collisions.

Fig. 6 shows the capacity of VoIP with silence suppression (VBR) for different bit rates. At 11 Mb/s, the capacity in DCF, PCF, DPCF and DPCF2 is 30, 30, 36 and 36 calls, respectively. In this case, there is no big difference in capacity between DCF and PCF. The reason is that, in PCF, the PC polls STAs even if they do not have packets to send (silence period). In other words, the PCF scheme wastes a lot of CF-Polls and Null packets. On the other hand, DPCF and DPCF2 reduce the number of wasted CF-Polls and Null packets which results in improving the capacity by 20%.

B. Capacity of VoIP with FTP Traffic

Next, we simulated the case when VoIP traffic and FTP (TCP connection that runs at the maximum sustainable rate) traffic coexist under the same BSS to see how much VoIP and FTP interfere with each other in terms of throughput. We assumed VBR for VoIP traffic. We considered 30 VoIP STAs, which is the capacity of VoIP in DCF and PCF, and 1 to 3 FTP STAs in DCF, PCF, DPCF and DPCF2. We also simulated DPCF and DPCF2 with 36 VoIP STAs, which is the capacity of VoIP in DPCF and DPCF2, and 1 to 3 FTP STAs.

In the IEEE 802.11 standard, packet-header overheads, management and control frames occupy large part of the bandwidth. Fig. 13 shows an analysis of usage of the bandwidth in our simulation when there are 30 VoIP STAs and 1 to 3 FTP STAs with DPCF. In this case, the combined throughput of VoIP and FTP is less than 3 Mb/s although the channel bandwidth is 11 Mb/s.

Figs. 7 through 12 show the average of the 90th percentile of the end-to-end delay of voice packets and throughput of VoIP and FTP in each case. We can see in Fig. 7 and 8 that the end-to-end delay of voice packets in DCF and PCF increases dramatically when some FTP traffic is added. On the other hand, the delay in DPCF and DPCF2 increases only slightly as FTP traffic increases. Furthermore, the figures show that the FTP throughput is larger in DPCF and DPCF2 than in DCF and PCF. In the case of 30 VoIP STAs and 3 FTP STAs, the FTP throughput is 1012 kb/s and 1309 kb/s in DPCF and DPCF2, respectively, while it is less than 900 kb/s in DCF and PCF. In the case of 36 VoIP STAs and 3 FTP STAs, the FTP throughput in DPCF and DPCF2 is 482 kb/s and 662 kb/s, respectively. The end-to-end delay of VoIP in DCF is not acceptable while it is still less than 100 ms in DPCF and DPCF2. DPCF2 tries to put voice packets into CFP as much as possible to reduce the number of CF-Polls and Null packets. This reduces the number of packets in CP, which allows other traffic such as FTP to be transmitted during CP.

VII. Related Work

A number of previous studies have evaluated the capacity in IEEE 802.11 networks for voice traffic and real time traffic in general. Hole et al. [9] provides an analytical upper bound value of the capacity for VoIP applications in IEEE 802.11b networks. A wide range of scenarios was evaluated including different delay constraints, channel conditions and voice encoding schemes with an analytical method. In that

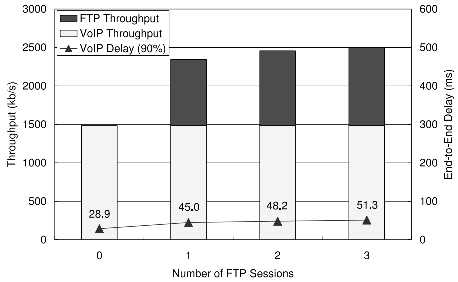

Fig. 9. End-to-end delay of VoIP and throughput of FTP and VoIP for DPCF, 30 VoIP nodes

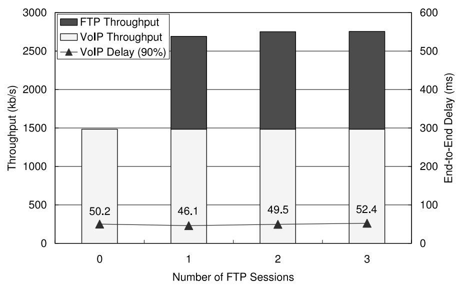

Fig. 10. End-to-end delay of VoIP and throughput of FTP and VoIP for DPCF2, 30 VoIP nodes

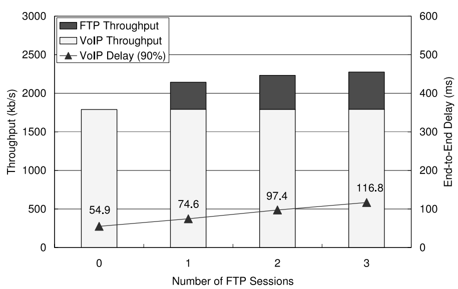

Fig. 11. End-to-end delay of VoIP and throughput of FTP and VoIP for DPCF, 36 VoIP nodes

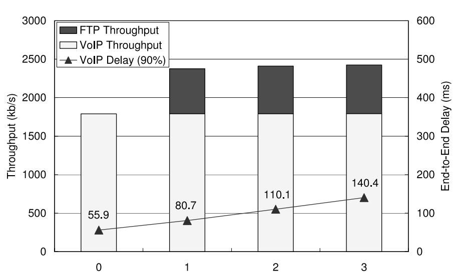

Fig. 12. End-to-end delay of VoIP and throughput of FTP and VoIP for DPCF2, 36 VoIP nodes

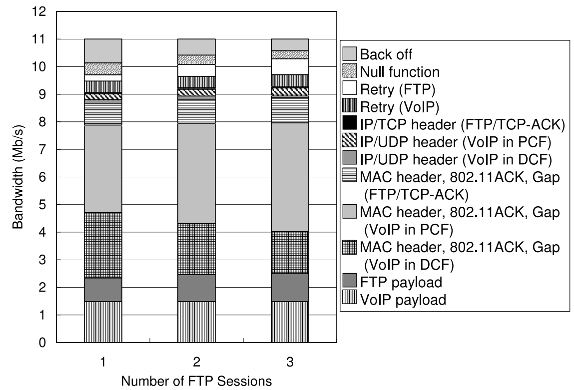

Fig. 13. Bandwidth usage of VoIP and FTP traffic for DPCF, 30 VoIP nodes

paper, the capacity of CBR VoIP with 20 ms packetization interval using G.711 was 12 calls. We confirmed this result with our analysis. In [10], the capacity of a system that uses the PCF for CBR and VBR voice traffic was analyzed, using Brady's model [11] and May and Zebo's model [12] for VBR voice traffic. In their analysis, they used a value of 75 ms and 90 ms as CFP interval, which causes a delay that is not acceptable for VoIP. The capacity for VoIP with a 90 ms CFP interval was 26 voice calls, but the maximum delay was 303 ms. In [13], the capacity of VoIP with IEEE 802.11e Enhanced DCF (EDCF) and Enhanced PCF (EPCF) was evaluated. They used G.711, G.729 and G.723.1 as voice codecs and assumed CBR traffic. IEEE 802.11e provides low end-to-end delay for voice packets even if mixed with best effort traffic. However, EDCF does not have mechanisms for reducing the number of collisions among the same priority traffics, which still remains a critical point, furthermore no effort is put in reducing the number of Null packets when using EPCF.

Several papers have proposed schemes for improving the capacity of VoIP. In [14], the capacity of VoIP in IEEE 802.11a was improved by using automatic rate selection instead of a fixed 6 Mb/s link rate. Hiraguri et al. [15] gave higher priority to VoIP traffic than other traffic types by using a shorter inter frame space and shorter backoff time for VoIP in IEEE 802.11a DCF mode. They showed that their priority scheme decreased the average delay and packet loss probability of VoIP packets compared to DCF, when VoIP and data traffic coexist. Using a shorter backoff time, however, increases collisions of VoIP packets as the number of VoIP nodes increases, and therefore does not improve the capacity for VoIP traffic. In [16], Suzuki et al. proposed a multiple polling lists scheme in which VoIP terminals are listed in the high-priority list. In their scheme, the PC polls terminals in the high-priority polling list first. They used a two-state Markov on/off model for VoIP traffic with exponentially distributed talk-spurts and silence periods. Their scheme can reduce the packet loss probability of VoIP when VoIP and other traffic coexist, however, they did not consider reducing the number of Null packets.

Ziouva et al. [5] presented a new polling scheme called CSSR for efficient support of VoIP over IEEE 802.11 networks. One similarity with our scheme is the use of an "active polling list". Only active nodes in the active polling list will be polled by the AP. However, there are many differences. First, the polling list management scheme is different. In the CSSR polling scheme, an STA is removed from the polling list when the start of a silence period is detected and it is added to the polling list k polling cycles after it is removed. In our DPCF, an STA is removed when the AP detects three consecutive Null packets, and an STA is added when the AP gets a packet from the STA during CP (Refer to Section IV-B.2). Secondly, the CSSR polling scheme uses a cyclic shifting of the position of the STAs in the polling list, in order to guarantee a uniformly distributed packet loss among the nodes. This packet loss is due to the fact that in the CSSR polling scheme, for each STA, if a new packet is generated before the previous packet has been transmitted, the older packet is discarded. In our DPCF, when an STA has more than one packet in its queue, all the pending packets are sent using the More Data field (Refer to Section IV-C) without introducing an additional packet loss. This makes our polling list management scheme much simpler than the one in [5], not requiring any cyclic shift process. In the CSSR polling scheme, there are two additional packet loss sources. 1. A STA has some packets to send, but it is not in the active polling list and therefore is not polled. 2. STA A is talking to STA B. STA A (in the active polling list of AP A) sends some packets to STA B, but STA B is not in the active polling list of AP B and therefore it will not be polled, which means that STA B will not receive the packets sent by STA A. In our approach these two components of packet loss are not present. When an STA has some packets to send, it can send them in CP, and the STA will be added into the polling list. Even if STA B is not in the polling list, packets for STA B are delivered after all STAs are polled. This is called "delivery and polling" in the standard PCF access scheme. Also, the CSSR scheme does not differentiate classes of traffic (real-time and best effort); our DPCF differentiates between real-time and best effort traffic to support QoS of VoIP traffic (Section IV).

VIII. Conclusions

In this paper, we proposed a modified PCF scheme, called DPCF, which resolves most of the disadvantages of the current PCF scheme with regards to VoIP, without changing the IEEE 802.11 standard. In DPCF, we use a dynamic polling list to reduce unnecessary CF-Polls and Null packets. We use a dynamic CFP interval and the More Data field to handle different audio packetization intervals. We categorize the network traffic into VoIP traffic and best effort, giving higher priority to VoIP traffic by allowing only VoIP traffic to be sent during the CFP. We propose a modified DPCF scheme which allows VoIP packets to be sent during CP only when there is more than one packet in an STA's queue. We have implemented our new schemes using the QualNet simulator and verified their performance. We improved the capacity of VoIP by 20%, while maintaining a lower end-to-end delay than in DCF and PCF. We also showed how our schemes perform better when FTP traffic is added to the VoIP traffic.

Our DPCF scheme is somewhat similar to IEEE 802.11e in that it categorizes the traffic and gives higher priority to VoIP traffic. Although IEEE 802.11e is not efficient to improve the capacity for VoIP traffic since no effort is put in reducing collisions among the same priority traffics in CP and Null packets in CFP, it can support QoS not only for VoIP traffic but also for other multimedia traffics. So, combining our scheme with IEEE 802.11e will be helpful for both improving the capacity and supporting multimedia traffics. In the future, we will analyze the capacity of IEEE 802.11e with our VoIP traffic model, and we will compare the results with the ones obtained using our approach.

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