

# Supporting VBR VoIP Traffic in IEEE 802.11 WLAN in PCF Mode

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## Abstract

With fast deployment of wireless local area networks (WLANs), the ability of WLAN to support real time services with stringent quality of service (QoS) requirements is attracting ever-increasing research interest. In this paper, we evaluate the capability of the point coordination function (PCF), which is part of the medium access control (MAC) mechanisms specified in the IEEE 802.11 standard for WLANs to support voice over IP (VoIP) applications. The capability of PCF mode in support of variable bit rate (VBR) VoIP traffic is investigated, where the silence suppression technique is employed in voice codecs so that no packets are generated in silence period. The impact of voice traffic burstiness and load on the average end to end delay is also evaluated, and various voice models are used to compare the results.

Our simulation shows that under the PCF mode the end to end delay of VoIP traffic may be effectively reduced by using VBR mode for the VoIP traffic. Moreover, the end to end delay is not sensitive to the burstiness of the voice traffic. These features make the PCF channel access method a promising alternative for conveying VoIP traffic in present IEEE 802.11 WLANs.

## 1. Introduction

Wireless local area networks (WLAN) hold the promise of providing unprecedented mobility, flexibility and scalability than its wireline counterpart, and much research effort has been attracted in this field. As a result of rapid growth in wireless networking market, the IEEE 802.11 standard for wireless local area networks was published in 1997, which covers the media access control (MAC) layer and physical layer specifications for WLANs [1].

The IEEE 802.11 WLAN has two different channel accessing mechanisms, namely, the distributed coordination function (DCF) and point coordination function (PCF). DCF is based on the carrier sense multiple access with collision avoidance (CSMA/CA) channel accessing mechanism, while PCF is based on the polling technique. The DCF operation mode consists of two techniques for packet transmission. The default scheme is a two-way handshaking technique where a positive acknowledgment is transmitted by the destination station upon successful reception of a packet from the sender station. The reason for an ACK is that the sender cannot determine whether its transmission is successful only by

listening to its own transmission. Another scheme involves a four-way handshaking technique known as request-to-send/clear-to-send (RTS/CTS) mechanism. By this scheme, the sender first sends the RTS to reserve the channel before its transmission, and upon receiving CTS from the receiver, the normal packet transmission and the ACK response proceeds. On the other hand, for the PCF operation mode, stations are polled in turn, and the station with a packet pending for transmission sends out the packet upon being polled.

While the DCF mode works in a distributed manner and is the fundamental channel accessing method, it is hard to implement quality of service (QoS) assurance in this mode. The lack of QoS assurance will become a significant problem when the wireless network is to carry real time traffic, where stringent QoS requirements are posed. To make things worse, DCF is found to have poor performance under heavy and unbalanced traffic load conditions [3]. Consider the case when the WLAN is connected to the wireline networks, and all the real time traffic from the wireline networks are routed through the access point (the "bridge" between the WLAN and the wireline network) to the wireless part. When this happens, excess delay may be incurred at the heavily loaded AP [2]. As an example, it is found in the study of [9] that a typical WLAN with 11Mbps bandwidth could only support a very limited voice over IP (VoIP) connections in DCF mode. This makes the DCF scheme an infeasible option to support quality of service for real-time traffic.

Based on these considerations, it was pointed out in [4] that PCF mode may be an alternative way to convey realtime traffic over IEEE 802.11 WLANs. The idea is that by using a centralized controller, the network may exhibit a more "packet switched" nature, and it is easier to realize QoS assurance in the central controller. This concept is elaborated in [5], where the capability of PCF mode to support voice services is investigated, and theoretical results are derived for the PCF mode with constant bit rate (CBR) voice traffic. Along this line, [6] carried out a simulation approach to study the performance of various polling schemes in PCF mode.

We observe that above works on PCF operation mode more or less concentrate on constant bit rate (CBR) traffic, whose fixed length packets are generated at pre-determined time intervals. Telephony traffic, on the other hand, is known to

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exhibit alternating periods of talk spurts and silences, and is more like a variable bit rate (VBR) traffic. By taking advantage of this feature, the bandwidth efficiency of WLAN may be significantly improved.

In this paper, we thus extend the work in [9], which studies the CBR VoIP traffic in DCF mode in IEEE 802.11 WLAN, by considering the VBR VoIP traffic supported by PCF mode of IEEE 802.11 WLANs. Our work is also an extension to the work of [5], which presented the theoretical results for CBR VoIP traffic under PCF mode. From a simulation approach, the focus of our study is on the determination of system design parameters based on the traffic patterns, as well as the effect of traffic burstiness on wireless network capacity. Such analysis may either provide a good reference to either system capacity design, or help determining optimal resource management algorithms.

This paper is organized as follows: Section 2 briefly introduces the operation of the PCF mode of an 802.11 LAN, in Section 3 we describe the setup of our simulation, in Section 4 the system performance is evaluated under various scenarios, and in Section 5 some issues on using PCF to support VoIP is addressed. Finally, Section 6 concludes the paper.

## 2. Background

The media access control protocol specified by 802.11 LAN includes two modes, namely, the DCF mode and the PCF mode. The DCF mode is the fundamental access method of the 802.11 MAC sublayer, while the PCF mode is optional. An 802.11 LAN may either work solely in DCF mode, or work alternatively in DCF mode and PCF mode. In the latter case, a point coordinator (PC) is required as the major control station to poll each mobile station. The PC may be any station in the network, but when the system is configured as an infrastructure network where access point (AP) is present, the AP normally takes up the responsibility to act as the PC. The period in which the system operates in PCF mode is called *contention free period* (CFP); while the period in which the system operates in DCF mode is called *contention period* (CP). A "superframe" is defined as the sum of these two periods.

The channel accessing priority of various frames is controlled by the length of time interval between frames, known as the *Inter-Frame Spaces* (IFS). There are five different kinds of IFS's: Short IFS (SIFS), Distributed cooperation function IFS (DIFS), Point coordination function IFS (PIFS), and Extended IFS (EIFS). SIFS is used by acknowledge packets to successfully received data or control packets, or subsequent packets to a previously sent data packet due to segmentation; DIFS is used by data packets to defer before contending for the channel; PIFS is used by the PC to take control of the channel and start a CFP; and EIFS is used by re-transmission of an erroneously transmitted data packet in DCF mode.

To start the CFP, the PC gains control of the channel after sensing the channel idle for a time period equivalent to PIFS.

During the CFP, the PC polls each CFP-Pollable stations (stations that can respond to polls). Upon receiving a poll a station either responds with an **Ack** packet, if it has no data to send; or respond with **Data+Ack** packet, if it has data to send. At the end of CFP, the PC sends out a **CF-End** packet to inform other stations of the end of CFP. This procedure is shown in Figure 1.

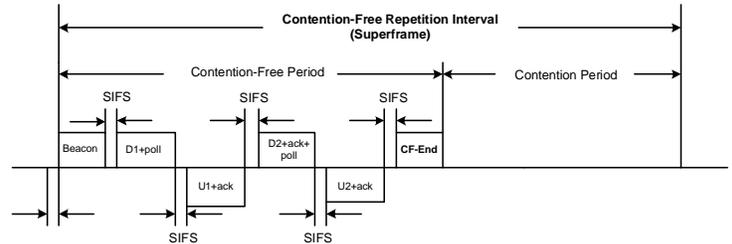


Figure 1 PCF Packet Transfer

## 3. Simulation Setup

### System description

The system framework is shown in Figure 2.

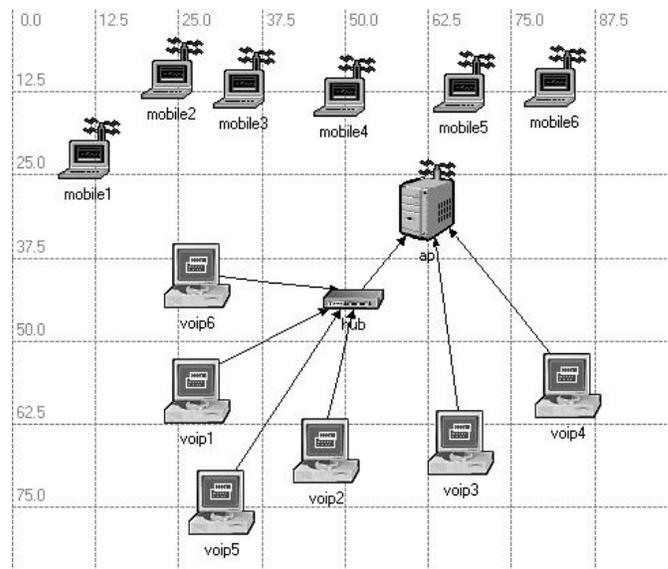


Figure 2 OPNET WLAN Model

There are four different kinds of nodes in the simulation system:

**VoIP traffic stations in wireline network**, labeled in Figure 2 as *voip1* to *voip6*. Since we only consider telephony traffic in this paper, and the range of a WLAN is in general small (in the order of several hundred meters), we assume all the VoIP traffic will go through the AP to the wireline network. For this purpose, we use VoIP stations on wireline network as one end of the VoIP connection. The VoIP node model is shown in the following figure. The VoIP packets are generated in the *traffic\_source* node, and the packet head information (source address and destination address, packet type, etc) are attached in the *wlan\_interface* node.

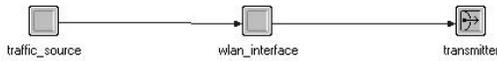


Figure 3 Node model for the VoIP traffic station nodes

**Access points.** The access point will act as the point coordinator during the CFP period to send polling packets to each CFP-Pollable mobile station (a mobile station that can respond to a polling packet) AP maintains a queue for each incoming traffic from VoIP. Upon polling a specific mobile station, the AP will first determine whether there is any packets to be sent to the polled mobile station. If there is, the AP will send out a **Data + CF-Poll** packet. Otherwise, the AP will send a **CF-Poll** packet.

The node model for the AP is shown below. Each packet arrival from the wireline network is stored in the *queue* node. The *proc* node takes charge of polling each mobile station, and it accesses the *queue* node to obtain the packet to be delivered to each mobile station being polled. The statistic wire from node *proc* to node *queue* carries the information of the station address currently being polled.

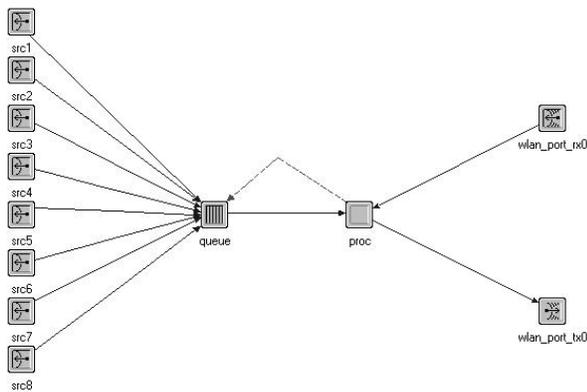


Figure 4 Node model for the access point

**Hubs.** The hubs are used to aggregate VoIP traffic and forward the packets from VoIP sources to AP. The node model for the hub is shown in Figure 5.

**Mobile stations,** labeled as *mobile1* to *mobile6* in Figure 2. The mobile station emulates the behavior of IEEE 802.11 mobile stations during CFP period. A local queue is maintained at the mobile station for its own generated traffic. Upon being polled, the mobile station either send out a **Data + ack** packet or an **Ack** packet, depending on whether the queue is empty.

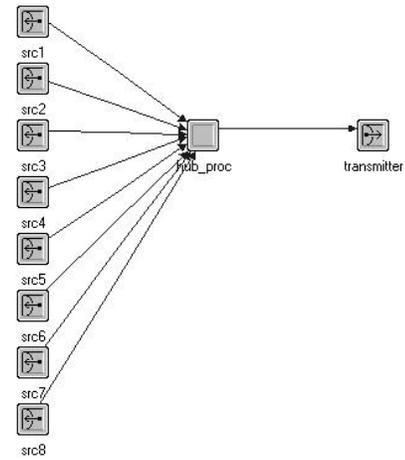


Figure 5 Node model for the hub nodes

The mobile node model is shown in figure 6. Packets generated by the traffic source (node *src*) are stored in node *queue*. Upon reception of a polling packet from the AP, the node *proc* will access the queue node and send a **Data** packet if the queue is not empty, or an **Ack** packet if the queue is empty.

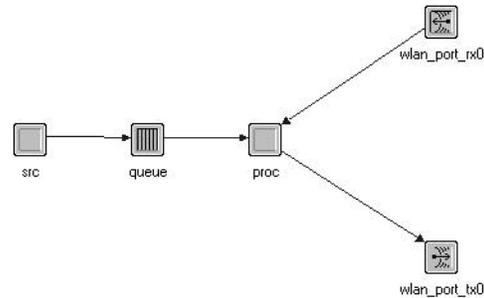


Figure 6 Node model for the mobile nodes

The following assumptions/simplifications are made in this paper:

- We assume that the VoIP stations in wireline networks are connected to the AP via point to point links with negligible link delay. Since we are only interested in the performance of the wireless part, this assumption is reasonable.
- We assume the channel is error free, and there is no channel outages for packet transmission. In our future study, we will extend our study on the effect of channel errors on PCF mode.
- We have neglected the *stretching* of CP [5] caused by a frame transmission at the end of CP, and assumed that the CP and CFP duration are deterministic variables. This simplification will not cause much deviations in the final result when the CFP duration is much longer than the data frame length. However, when the CFP duration is

comparative to the frame length, this factor cannot be neglected.

### Simulation parameters

The parameters for the simulation are chosen as in Table 1, according to the IEEE 802.11 protocol specifications.

WLAN Parameters	Values
PHY layer specification	DSSS
Transmission rate	11Mbps/sec
ACK packet size	14 bytes
SIFS interval	10ms
CF-end packet size	20 bytes

Table 1: WLAN Parameters

Assume using the ITU G711 a-Law codec for the voice calls. Then, during the ON period, frames are generated every 10ms with size 80 bytes. However, the actual frame at the physical layer has a much larger size than this, with RTP, UDP, IP, MAC and PHY layer overheads included. Table 2 shows the traffic parameters we have chosen for the simulation, which results in an equivalent packet size of 422 bytes.

VoIP Traffic Parameters	Values
Packet interarrival time during ON state	10ms
Voice packet length	80 bytes
RTP layer overhead	12 bytes
UDP layer overhead	8 bytes
IP layer overhead	20 bytes
MAC layer overhead	34 bytes
PHY layer overhead	24 bytes

Table 2: Traffic parameters

The Weibull distribution is used to model the ON and OFF periods of the source traffic. By proper choice of its parameters, the Weibull distribution may represent certain levels of burstiness in the traffic. The probability distribution function of Weibull distribution is

$$P\{X \leq x\} = F(x) = 1 - e^{-(x/\beta)^\alpha},$$

with its mean given by

$$E(X) = \frac{\beta}{\alpha} \Gamma\left(\frac{1}{\alpha}\right). \quad (1)$$

### Performance measures

An important measure for the quality of voice transmission is the end to end delay. This delay consists of two parts: One part is the queuing delay, incurred by the local queue in the mobile station and the access queue in the AP; and the other part is the MAC access delay, incurred by the MAC layer delays in waiting for the station to be polled.

Another interesting metric might be the amount of dropped packets either due to the limited queue capacity, or due to the need to limit the overall end to end delay. When the packet waiting time in the queue exceeds a certain threshold, it is generally better to drop the packet to guarantee the overall voice quality.

### 4. Simulation results

In Figure 3 we have shown the end to end delay perceived at the mobile end, with CBR traffic load on the channel (which means that the traffic source is always in ON state). Here we assume no packet will be dropped due to excess waiting time in the queue. From this figure, we observe that the average end to end delay remains a constant for 6 and 15 VoIP connections, but it increases linearly with time with 17 VoIP connections case. This means that the number of VoIP connections has exceeded the system capacity. Without any packet drops, more and more packets accumulated at the local queue of each station and result in an increasing end to end delay.

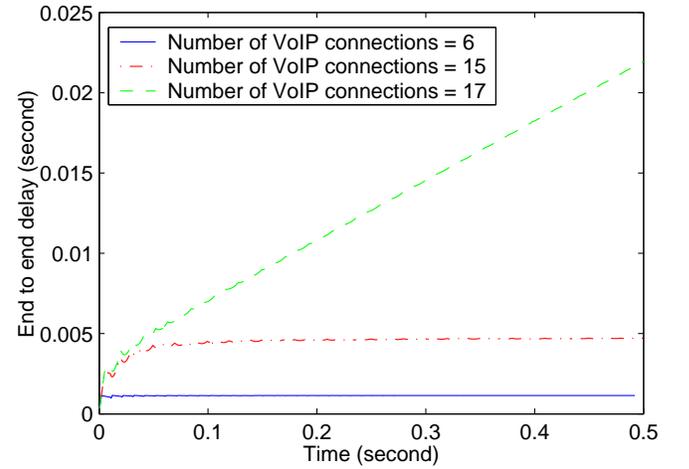


Figure 7 End to end delay of CBR VoIP connections.

As a brief calculation to find the number of CBR VoIP calls that the network could support at the same time, we note that the time spent for each polling of one mobile station is

$$T_{poll} = 2(T_{data} + SIFS) \approx 614\mu s$$

where  $T_{data}$  is the time spend to transmit a **Data+CFP-Poll** frame or a **Data + CFP-Ack** packet.

Suppose there are  $n$  VoIP connections in the network. Recall that the voice packet inter-arrival time is 10ms, the following condition to avoid queue build up at each node the following condition need to be satisfied:

$$nT_{poll} + T_{stretch} + T_{CFPend} \leq 10ms$$

where  $T_{stretch}$  is the stretching time, and  $T_{CFPend}$  is the time to send a **CFP-End** packet. Ignoring the stretching time, the number of supported VoIP connections thus satisfies

$$n \leq 15$$

which means that the WLAN can support a maximum of 15 VoIP connections with CBR traffic.

To observe the capacity improvement by adopting VBR voice coding, assume the silence suppression technique is used in the VoIP sources in another set of experiments. We assume the average ON time is 352ms and the average OFF time is 650ms, as is used in [5]. We also assume the ON state time and the OFF state time follow exponential distribution. The system end to end delay obtained under this scenario is shown in Figure 8. From this figure, we observe that significant improvements are achieved by adopting VBR coding. As an example, the average end to end delay for 15 VoIP connections drops from 5ms to 0.5ms. Even for 20 VoIP connections, the average end to end delay is still below 1ms.

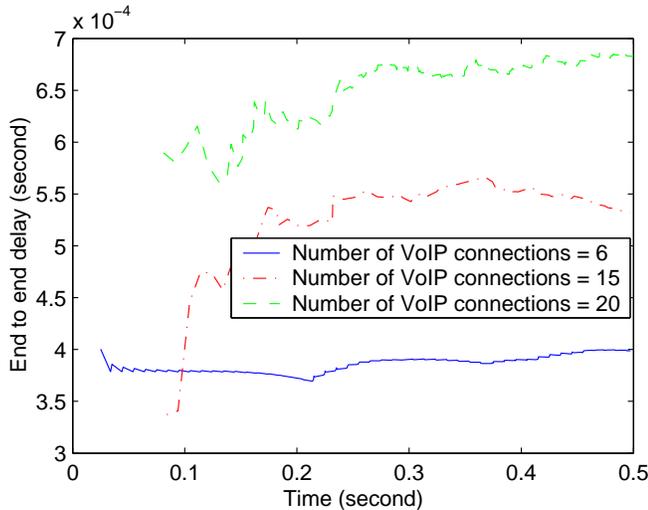


Figure 8 End to end delay of VBR VoIP connections

In our next experiment, we evaluate the effect of burstiness of VoIP packets on the performance. The value of  $\alpha$  is chosen as 0.65, 0.70 and 0.88. Note that smaller  $\alpha$  values correspond to higher bursty traffic. The value of  $\beta$  is computed from Equation (1). Figure 9 shows the simulation result for 20 VoIP connections. We observe that the burstiness does not have significant impact on the average end to end delay. In fact, the delay for the less bursty traffic ( $\alpha = 0.88$ ) even exceeds that of a more bursty traffic ( $\alpha = 0.65$ ). This indicates that under PCF mode the system performance is not sensitive to the traffic burstiness.

We also compare the end to end delay of a system of 20 VoIP connections with different voice models. Since as shown in Figure 9, the VoIP traffic burstiness does not have significant impact on the system performance, in this experiment we assume the system ON state time and the system OFF state time follow exponential distributions. The voice models employed in this experiment is shown below:

Model	Mean ON period	Mean OFF period
Brady's model [7]	1 second	1.35 second
May and Zebo model [8]	352ms	650ms

Table 3: Voice Models

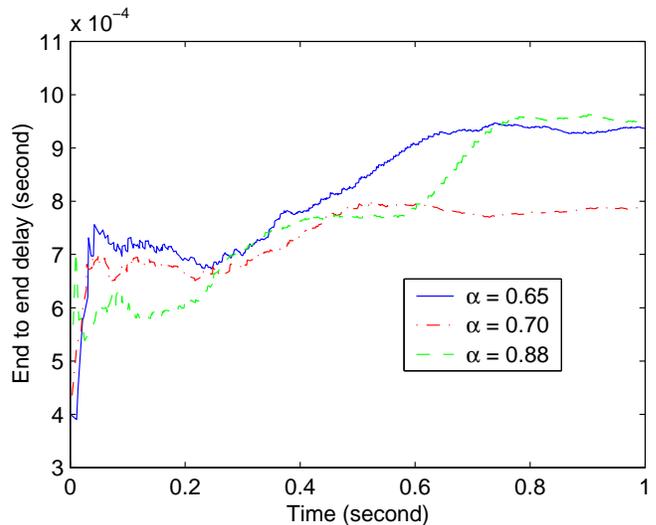


Figure 9 Effect of VoIP traffic burstiness on the average end to end delay (with 20 VoIP connections)

The result is shown Figure 10. The May's model generates higher end to end delay than the Brady's model since the average ON time for May's model is higher than the Brady's model. For both of these cases, a low delay may be achieved.

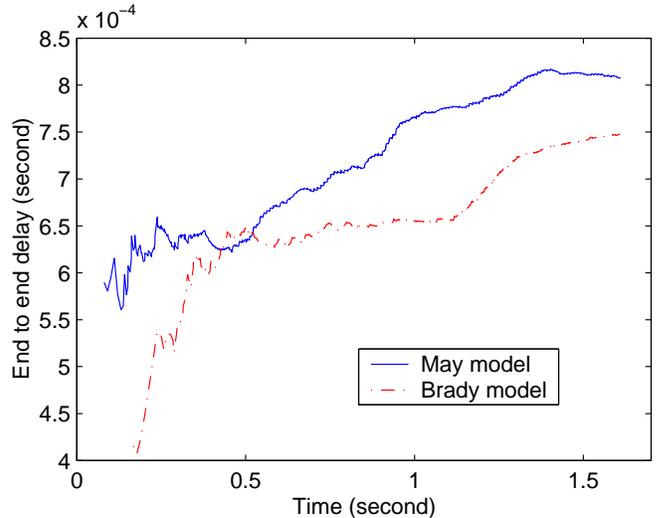


Figure 10 End to end delay of different voice models (20 VoIP connections)

Above simulations are based on the assumption that the network always works in PCF mode. However, in IEEE 802.11 networks the DCF mode is the fundamental channel access method and the co-existence of these two schemes are required, which results in the alternation of CP and CFP's. To study the impact of the existence of CP on the performance, we simulate the scenario with 20 VoIP connections with VBR traffic following May's model, with ON time and OFF time exponentially distributed. Figure 11 shows the result with different CP duration, with CFP duration as 50ms (we assume that the VoIP traffic can only be sent in PCF mode).

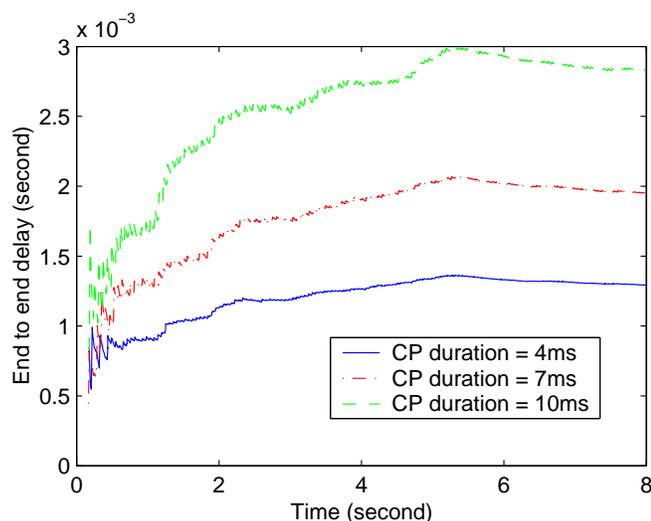


Figure 11 The end to end delay with alternation of CP and CFP (20 VoIP connections, Brady's voice model, CFP duration = 50ms)

By increasing the CP duration from 4ms to 10ms, the average end to end delay is increased from around 1ms to 3ms. Thus the choice of CP duration is important to meet the QoS requirement posed by the VoIP calls. We also need to keep in mind that the CP duration cannot exceed 10ms, or queues may be built up at the VoIP stations since the interarrival time of VoIP packets used in our simulation is 10ms.

## 5. Discussion

In this paper, we have assumed that the AP keeps polling each VoIP station regardless of whether it has data for transmission. This may result in considerable polling overhead when the number of stations is large while only a few of these stations need to send data. In this case, the overhead may be reduced by maintaining a dynamic polling list at the AP. A station that has not sent any packets for a certain amount of polls may be removed from the polling list. When packets arrive at that station, the station may send a polling request to the AP so that the AP may add this station back to its polling list. By this scheme, the stations that are not in the AP's polling list need to find some way to send their polling requests. In this sense, the DCF mechanism is still needed in addition to PCF, and this requires alternation of CP and CFP intervals.

However, one remaining problem for this mechanism is that the length of CP cannot exceed the minimum polling interval requested by the QoS traffic. When the number of VoIP connections is large, it may be hard for a VoIP station to contend the channel and send its polling request to the AP in limited CP duration. To make things worse, the amount of best effort traffic load in DCF mode will also contend with the VoIP stations. By this consideration, keeping a static

polling list at the AP may reduce the possible excessive delays caused by the heavy load in DCF mode.

We also observe the possibility to use some dynamic reservation based polling scheme where some channel resources are reserved for the VoIP stations to send their polling requests in VBR mode [10]. Our future work thus includes the performance evaluation of the enhanced MAC schemes in support of VBR QoS traffics.

## 6. Conclusion

In this paper we have presented the simulation results of the performance of PCF to support VBR VoIP applications on WLAN. Our results show that the end to end delay for VBR traffic is significantly reduced compared with CBR VoIP traffic, and the system capacity is also increased. Our results also show that the performance of PCF is not sensitive to traffic load, burstiness and voice patterns.

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