

Performance Analysis of the Packetized Voice Transmission with PCF in an IEEE 802.11 Infrastructure Wireless LAN

Xiyan Ma, Yi Wu, Zhisheng Niu*, Tamio Saito**

*State Key Lab on Microwave and Digital Communications,

Dept. of Electronic Engineering, Tsinghua University, Beijing, China

**Fujitsu Research and Development Center Co., LTD., Beijing, China

Abstract—The IEEE 802.11 standard supports the coexistence of asynchronous and time-bounded traffic by use of the two modes of Medium Access Control (MAC) protocol termed as DCF (Distributed Coordination Function) and PCF (Point Coordination Function) respectively. The former is a random access scheme originally designed for delay-insensitive data applications, and the latter is based on polling mechanism which is more suitable for real-time services such as voice and video etc. In this paper, we focus on the performance of the packetized voice transmission using PCF mode in IEEE 802.11 Wireless LAN while the DCF mode is used to provide minimum bandwidth available for data transfers. The effect of fragmentation threshold, echo cancellation and burst errors on network performance is theoretically analyzed.

Keywords - IEEE 802.11; packetized voice; PCF; Wireless LAN

I. INTRODUCTION

The IEEE 802.11 Wireless LAN (WLAN) is designed for Internet-based applications such as World Wide Web, file transfer etc. at the first beginning. With the notable popularity and success gained nowadays for data application in campus network, university campuses and airports for instance, people are more and more interested in providing voice services with WLAN as a local alternative of cellular wireless network.

The IEEE 802.11 standard specifies the coexistence of Distributed Coordination Function (DCF) and Point Coordination Function (PCF) in the Medium Access Control (MAC) sub-layer architecture [1]. DCF uses a contention-based random access mechanism named Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) for all stations to share the wireless medium, which is designed for asynchronous data transmission. The optional PCF uses polling scheme to provide contention-free medium access through a Point Coordinator (PC: usually the Access Point of a Basic Service Set—BSS), which is more suitable for delay-sensitive traffic like voice and video.

However, many doubts have been arisen concerning that

whether the 802.11 PCF mode is well suitable for supporting real-time traffic. On the specific topic of supporting time-sensitive services with IEEE 802.11 MAC protocols, Kopsel and Wolisz [2] compare the suitability of the basic DCF and PCF protocols for the transmission of audio data in an interactive scenario through simulation. They show in the paper that the most suitable medium access mechanism for audio is PCF which reduces the mean channel access delay by avoiding contention phases during CFP (Contention Free Period) and reducing contention in CP (Contention Period). Paper [3] makes investigation of the performance of the IEEE 802.11 networks by simulating asynchronous data traffic in a 1 Mbps *ad hoc* network and packetized voice traffic in a 1 Mbps infrastructure network. The simulation results show that voice packets can be supported together with data packet but an echo canceller is required for packet voice systems. Veeraraghavan *et al* design and analyze a system that uses the polling mode of PCF for interactive voice traffic, showing the maximum number of voice calls that can be handled in a certain inter-poll period with a worst case delay value [4]. But as illustrated in [4], most of the voice delays exceed the delay requirement for voice typically limited to 25 ms without echo canceling, and the performance is not analyzed in the case of using echo cancellation. Moreover, the authors carry out an error analysis to show that voice packets can be expected to suffer a high packet error rate, but conversations number influenced by retransmission are not analyzed.

In this paper, we examine the performance that packetized voice traffic experiences when supported by the PCF access scheme of an IEEE 802.11 WLAN, while the DCF mode supports the data transfers. The performance is theoretically analyzed in terms of maximum number of conversations and the available bandwidth for data transmission, taking echo cancellation, fragmentation threshold and the burst error characteristic of wireless channel into account.

The structure of the paper is as follows. Section II presents the system model that allows the analysis of performance achieved for voice traffic. Maximum number of conversations

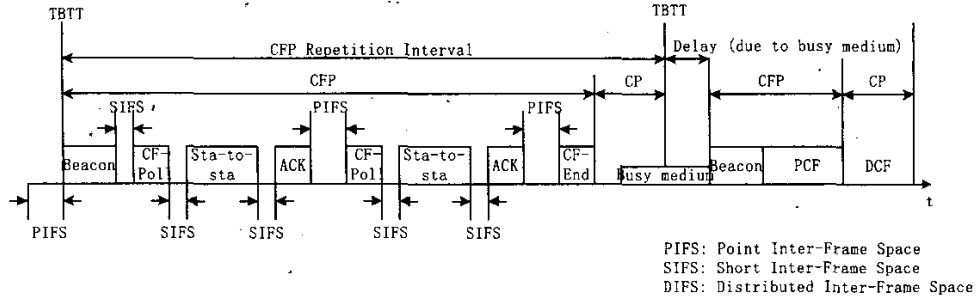


Figure 1. Voice packet transmission in CFP and CFP/CP alternation [5]

and bandwidth for data transmission are computed in the case of using/not using echo cancellation and burst error channel. Extensive numerical results are presented and analyzed in section III and finally section IV concludes the paper.

II. SYSTEM DESCRIPTION AND PERFORMANCE ANALYSIS

A. System description

In this paper, we consider the infrastructure network that can employ the combination of PCF and DCF. The network's architecture is as follows, similar with [5]:

- One PC that is the AP of the BSS,
- Data only stations that access the medium using DCF,
- Voice stations that use PCF for packetized voice transmission but also support data services, using DCF to establish connections through PC.

Both the two functions are employed during the network operation, using a time-sharing mechanism. All stations transmit their data packets in CP using CSMA/CA as specified in IEEE 802.11 standard, and voice stations are allowed to transmit the voice packets when polled by PC during the CFP. The PC controls the length of CFP interval according to the available traffic and the size of its polling list.

We assume the voice coding rate as 64 kbps, i.e., the voice stations generate traffic at 64 kbps constant rate (Constant Bit Rate: CBR traffic). And a voice packet is assumed to be generated every CFPR (Contention Free Period Repetition) interval. So the voice packet size depends on the duration of CFPR interval. Meanwhile, we assume that each voice station starts sampling every TBTT (Target Beacon Transmission Time) instant, ignoring the packetization delay. Moreover, all stations in the polling list are polled just once during each CFP, in a fixed sequence. The process of voice packet transmission in CFP and CFP/CP alternation are illustrated in Fig. 1. Seen from Figure 1, it may happen that a station begins to transmit a frame just before the end of CP, hence elongating the current CFP repetition interval and shorten the next CFP.

We call this as a "stretched" CP.

B. System performance analysis

a) Echo cancellation is used.

As in paper [5], let T_{CFPR} , T_{CFP} and T_{CP} denote the CFPR, CFP and CP intervals respectively. The time length T_{con} of a connection between stations for exchanging the voice packets during PCF is given by:

$$T_{con} = 2(T_{CF-Poll} + T_v + 2T_{SIFS} + T_{ACK} + T_{PIFS}) \quad (1)$$

where $T_{CF-POLL} = S_{CF-Poll} / R_c$ ($S_{CF-Poll}$ is the size of CF-Poll frame in bits, R_c is the channel bit rate) and T_v is the transmission time of a voice packet with sampling rate R_s . Notice that headers and beacons are transmitted at the rate of 1 Mbps (denoted with R) required in standard [1][4]. Use H_{PH} and H_{MAC} to denote the Physical layer and MAC layer headers respectively, hence

$$T_v = \frac{H_{PH} + H_{MAC}}{R} + \frac{T_{CFPR} R_s}{R_c} \quad (2)$$

Let N_{max} denote the maximum number of voice conversations that can be handled during the CFP, then

$$T_{CFP} = T_{PIFS} + T_{beacon} + N_{max} T_{con} + T_{CF-END} \quad (3)$$

Here $T_{beacon} = S_{beacon} / R$, $T_{CF-END} = S_{CF-End} / R_c$ and T_{CFP} is the length of CFP interval considering a "stretched" CP. Using $T_{stretch}$ to describe the maximum delay at the start of CFP because of a "stretched" CP, then $T_{CFP} = T_{maxCFP} - T_{stretch}$. Unlike paper [5], we take fragmentation threshold f (bits) into consideration, so

$$T_{stretch} = T_{RTS} + T_{CTS} + T_{max} + T_{ACK} + 2T_{SIFS} \quad (4)$$

$$T_{max} = (m-1) \left(\left[\frac{f}{R_c} + \frac{H_{PH} + H_{MAC}}{R} \right] + T_{ACK} + 2T_{SIFS} \right) + T_{last} \quad (5)$$

where $T_{RTS} = S_{RTS} / R_c$, $T_{CTS} = S_{CTS} / R_c$, $T_{ACK} = S_{ACK} / R_c$ and $m = \lceil S_{maxSDU} / f \rceil$ ($\lceil x \rceil$ means the nearest integers less than or equal to x). T_{last} is given by:

$$T_{last} = \frac{S_{max_SDU} - f(m-1)}{R_c} + \frac{H_{PH} + H_{MAC}}{R} + T_{ACK} + 2T_{SIFS} \quad (6)$$

So $T_{CFPR} = T_{min_CP} + T_{max_CFP} = T_{min_CP} + T_{CFP} + T_{stretch}$. According to the standard [1],

$$T_{min_CP} = T_{max} + 2T_{SIFS} + 2T_{slotTime} + 8T_{ACK} + T_{DIFS} \quad (7)$$

Now we can get the expression of N_{max} (and the maximum number of voice stations: $2N_{max}$):

$$N_{max} = (T_{CFPR} - T_{min_CP} - T_{stretch} - T_{beacon} - T_{CF-End} - T_{PIFS}) / T_{con} \quad (8)$$

Because the MAC frame has a maximum size ($maxPayload$), the size of the voice packet must be up to this value. So the voice packet generation interval T_{CFPR} has an upper bound: $T_{max_CFPR} = maxPayload / R_s$. The value $maxPayload$ in an IEEE 802.11 WLAN is 2304×8 bits, so T_{max_CFPR} equals 288 ms for 64 kbps CBR packetized voice, which satisfies the QoS (Quality of Service) requirement for voice delay typically limited to 500 ms.

So the remaining percentage of bandwidth for asynchronous data transmissions is given by:

$$BW_d \% = (T_{CFPR} - T_{stretch} - T_{PIFS} - T_{beacon} - T_{SIFS} - N_{max} T_{con} - T_{CF-End}) 100 / T_{CFPR} \quad (9)$$

b) Echo cancellation is not used.

Considering the maximum delay requirement, the maximum number of conversations for different CFPR intervals without echo cancellation is given by [5]:

$$N_{max} = \begin{cases} N_{max}, & \text{if } D_{max,LS} \leq d_{max} \\ \frac{d_{max} - T_{stretch} - T_{beacon} - T_{SIFS}}{T_{con}}, & \text{otherwise} \end{cases} \quad (10)$$

where $D_{max,LS}$ is the maximum voice packet delay the last station on the polling list can suffer and d_{max} is the upper bound of 25 ms without echo cancellation. As for the bandwidth for data transfers, N_{max} in (9) is substituted with N_{max} .

c) Error analysis

The IEEE 802.11 MAC protocol supports retransmission

TABLE I. MAXIMUM NUMBER OF CONVERSATIONS WITHOUT ECHO CANCELLATION

		Fragmentation threshold (bits)					
		300	500	1000	1500	2000	2304
CFPR interval (ms)	50	-5	2	6	7	7	7
	100	-2	2	4	5	6	6
	150	-2	1	4	4	5	5
	200	-2	1	3	4	4	4
	250	-2	1	3	3	3	3

to handle transmission errors in both PCF and DCF. However, retransmission is typically avoided for real-time traffic due to delay constraint. Here, we examine the influence of burst error channel characteristic on the performance when voice packets are transferred by PCF. Using the same expression of maximum total packet error probability P_{e_max} in [4], the probability that a voice packet undergoes i times transmission ($i-1$ times retransmission) before correctly received is as follows:

$$P_i = P_{e_max}^{i-1} (1 - P_{e_max}) \quad (11)$$

So the average number of transmission n is given by $n = \sum_{i=1}^{\infty} iP_i = 1/(1 - P_{e_max})$. The average transmission time T of

a voice packet is: $T = \sum_{i=1}^{\infty} iT_v P_i = T_v / (1 - P_{e_max})$. Substitute the T_v in equation (1) with T , then the system performance under burst error conditions can be evaluated.

III. NUMERICAL RESULTS

In this section, we present and discuss numerical results showing the performance of the IEEE 802.11 BSS infrastructure network with coexistent voice and data traffic. Parameters for numerical computation are: $H_{MAC}=34 \times 8$ bits, $H_{PH}=16 \times 8$ bits, $S_{ACK}=30 \times 8$ bits, $S_{RTS}=36 \times 8$ bits, $S_{CTS}=30 \times 8$ bits, $S_{CF-Poll}=50 \times 8$ bits, $S_{CF-End}=36 \times 8$ bits, $S_{beacon}=106 \times 8$ bits, $T_{SIFS}=10$ usec, $T_{PIFS}=20$ usec, $T_{DIFS}=50$ usec, $T_{slotTime}=20$ usec, $channel\ bit\ rate=5.5$ and 11 Mbps.

Taking the independent rate of headers and beacons into account, Figure 2 depicts the maximum number of conversations with different fragmentation thresholds at 11 Mbps channel bit rate, in the case of using echo canceller. The results of not using echo canceling are listed in TABLE 1. Obviously, the number of conversations supported is much greater with echo cancellation than without. As illustrated in Figure 2, with the increase of fragmentation threshold, the

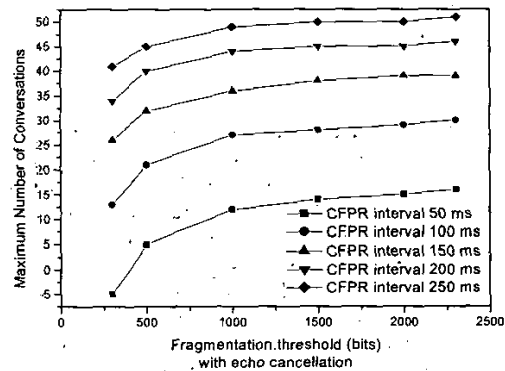


Figure 2. Maximum of conversations supported

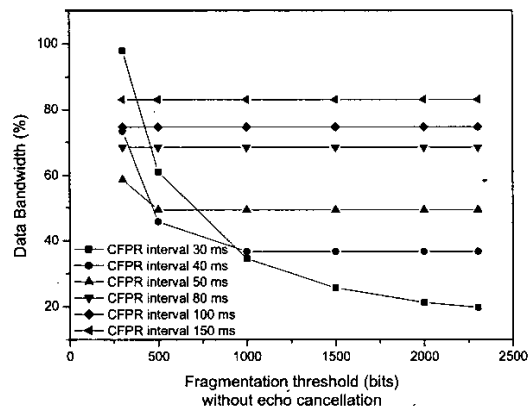
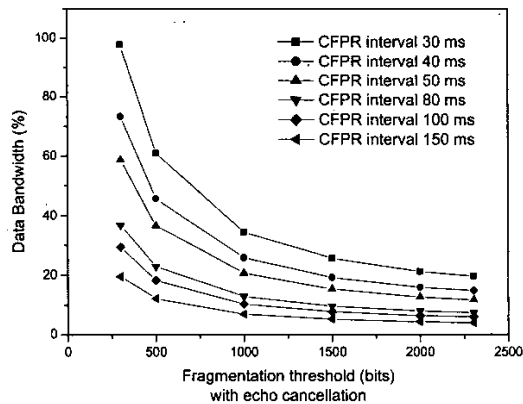


Figure 3. Available bandwidth for data transmissions

maximum number of conversations increases, though the increasing rate decreases when the threshold becomes large. When the fragmentation threshold is too small, much time is wasted in transmitting ACK frames and SIFS between fragments in the “stretched” CP. So the efficient time for voice packets transmission are decremented and the number of conversations will decrease. But the CFPR interval’s influence is different in the two cases. Using echo canceller, with the increase of CFPR interval, the number of conversations keeps increasing. On the other hand, the larger of CFPR interval, the smaller of the conversations number without echo cancellation, as long as the fragment size exceeds 500 bits, showed in TABLE I. The trend conforms to results in [5] since the CFPR interval is larger than the specific value, beyond which the maximum number of conversations decreases with the increase of CFPR interval. Notice that if fragment size is smaller than 500 bits (for example 300 bits), the system cannot support any conversations.

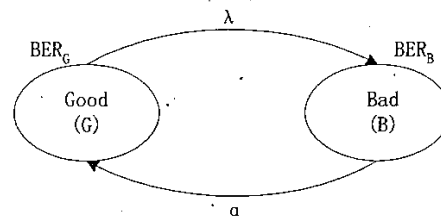


Figure 4. Markov chain model of wireless channel

TABLE II. PARAMETERS FOR BURST ERROR MODELS

Case	BER _G	BER _B	α	λ
1	10 ⁻¹⁰	10 ⁻⁵	10/sec	30/sec
2	10 ⁻⁸	10 ⁻⁴	10/sec	30/sec
3	10 ⁻⁶	10 ⁻³	10/sec	30/sec

Figure 3 shows that the percentage of bandwidth of the CFPR interval decreases with the increase of fragmentation threshold and CFPR interval when using echo canceller, since the number of conversation increases. However, without echo canceller, the fragmentation only shows its influence when CFPR interval is smaller than 80 ms. If CFPR interval grows larger, the percentage does not change with fragment size. Compromised performance for both voice and data traffic is achieved when the fragmentation threshold is around 1000 bits.

Figure 5 illustrates the performance of our system under the assumption of ideal and non-ideal channel. We model the burst error channel with two-state continuous time Markov chains [6] in three cases different in the Bit Error Rate (BER) parameters as listed in TABLE II. The Markov model is showed in Figure 4 and the fragmentation threshold equals 1000 bits.

In the case of ideal channel, the number of conversations increases as the CFPR interval and the channel bit rate increases using echo canceller. But without echo cancellation, the conversations number reaches a maximum value at a specific CFPR interval. Beyond that value, the number decreases while the CFPR interval increases.

However, the maximum number of conversations that can be supported decreases sharply when the channel’s BER increases, no matter whether the echo cancellation is used or not. For instance, in case 3 of burst error channel, the conversations number drops to zero with both 5.5 and 11 Mbps channel bit rate. Besides, notice that in the case of using echo canceller, there is an “optimal” value of CFPR interval that supports the maximum number of conversations when channel, which is quite different with ideal channel case. The reason is that with the increase of CFPR, the generated voice packet size increases, which makes it easier to suffer from error and cause more retransmission, hence decrement the conversations number. Moreover, the specific value decreases

when the channel condition deteriorates. Similarly, the value of CFPR interval that supports the maximum number of conversations also decreases when the BERs increase without echo cancellation. The results can give instruction on selecting best value of CFPR interval during system operation.

IV. CONCLUSIONS

In this paper, we investigate the packetized voice transmission with minimum bandwidth allocated to data traffic over an IEEE 802.11 BSS infrastructure network. Voice packets are transferred by use of PCF mode specified in standard MAC sub-layer protocols, while data services are fulfilled using DCF. Performance of the network is theoretically analyzed in terms of maximum number of supported conversations and available bandwidth of data transmissions, taking echo cancellation, fragmentation threshold and burst error channel into account.

According to our analysis, PCF mode can well support packetized voice transmission using echo canceller, while the performance degrades quickly without echo cancellation especially when the CFPR interval value increases. When payloads are fragmented into small packets, large fragment size such as 2000 or 2304 bits does favor to voice traffic at the expense of available bandwidth for data transmissions. Compromised performance for both voice and data traffic is achieved when the fragmentation threshold is around 1000 bits. Besides, burst error characteristic of wireless channel is considered, which shows that when channel condition deteriorates, less number of voice conversations can be supported. There is a specific value of CFPR interval that can support maximum number of conversations no matter if the echo canceller is used or not. Moreover, the value decreases when the bit error rate of channel increases.

REFERENCES

[1] IEEE Standard for Wireless LAN Medium Access Control (MAC) and Physical (PHY) Layer Specifications, *IEEE Std. 802.11*, New York, Nov. 1997.
 [2] A. Kopsel, A. Wolisz, "Voice Transmission in an IEEE 802.11 WLAN Based Access Network," in *Proc. ACM International Workshop on Wireless Mobile Multimedia*, July 2001.
 [3] B. P. Crow, I. Widjaja, J. G. Kim, and P. Sakai, "Investigation of the IEEE 802.11 Medium Access Control (MAC) Sublayer Functions," in *Proc. IEEE INFOCOM'97*, vol. 1, pp. 126-133, Apr. 7-12 1997.

[4] T. M. Veeraraghavan, N. Cocker, T. Moors, "Support of Voice Services in IEEE 802.11 Wireless LANs," in *Proc. IEEE INFOCOM 2001*, Vol. 1, pp. 488-497.
 [5] E. Ziouva, T. Antonakopoulos, "CBR Packetized Voice Transmission in IEEE 802.11 Networks," in *Proc. Sixth IEEE Symposium on Computers and Communications 2001*, pp. 392-398, 2001.
 [6] E. Gilbert, "Capacity of a Burst Noise Channel," *Bell Syst. Tech. J.*, Vol.39, pp. 1253-66, Sept. 1960.

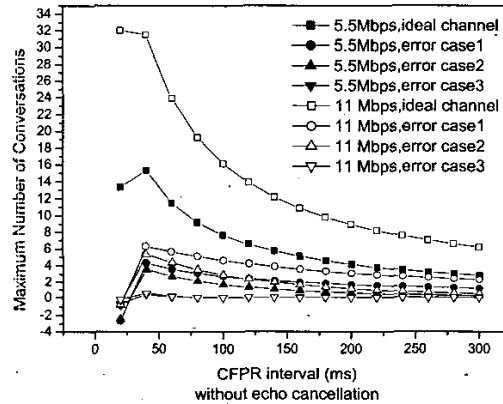
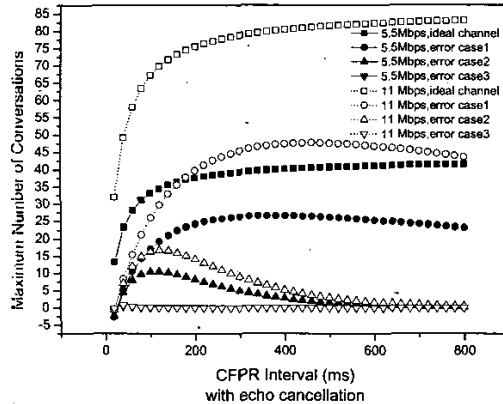


Figure 5. Maximum number of conversations in burst error channel