Performance Analysis of IEEE 802.11 WLAN to Support Voice Services*

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Abstract

This paper studies the performance of the IEEE 802.11 standard MAC protocol for integrated data and voice transmission with the DCF (Distributed Coordination Function) and the PCF (Point Coordination Function). By simulation, we evaluate the network performance for various protocol parameters, especially, the delay jitter for voice traffic. The main factor to influence delay jitter is given. Numerical results show that it is important to choose appropriate parameters and compromise the number of voice stations and the data traffic throughput to get the enhanced performance of IEEE 802.11. The performance of protocol in theory is derived and is verified by the simulation results.

1. Introduction

Wireless local area network (WLAN) provides a resolution to realize mobile Internet and many products of WLAN have been commercially available at present. The most effective standard of WLAN is IEEE 802.11 protocols[1]. The performance of the IEEE 802.11 DCF has already been studied by many researchers[2,3]. However, the performance of combining DCF and PCF to operate in a common repetition interval is not analyzed. Regarding the data traffic, we can find performance evaluation results by taking into account throughput and average packet delay in [2~6], but the real-time services such as packet voice are not considered in these papers. Many studies on the capacity of WLAN to support voice services have been reported[8~10]. In [8], the effect of data traffic on the performance of packet voice has not been researched. Furthermore, performance evaluation to integrate voice and data can be found in [9,10] though the access method is traditional CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance). Integration of delay-sensitive services and non-delay-sensitive services has been discussed extensively in the literature [7~13]. Most of these studies are concerned with average delay but not analyze the delay jitter, i.e. delay variance, though it's so important to QoS of time-bounded services.

This paper studies the performance of the IEEE 802.11 standard MAC protocol by taking into account both data transmission with the DCF and voice transmission with the PCF. By simulation, we evaluate the performance of the protocol in terms of throughput and average MPDU (MAC Protocol Data Unit) delay for various values of CFP maximum duration. It is worth noting that the delay jitter performance of packet voice is discussed in this paper.

The rest of this paper is organized as follows. Section 2 describes the IEEE 802.11 protocol briefly. Our simulation model and the analysis results are presented in section 3. Finally in section 4 the conclusions are drawn.

2. Overview of the IEEE 802.11 MAC protocols

The IEEE 802.11 MAC layer supports two fundamentally different MAC schemes, i.e. DCF and PCF.

The DCF based on CSMA/CA is the fundamental access method used to support asynchronous data transfer on a best effort basis. The DCF supports contention services which imply that each station with an MSDU (MAC Service Data Unit) queued for transmission must contend for access to the channel. After the MSDU is transmitted, it must recontend for access to the channel for all the subsequent frames. Contention services promote fair access to the channel for all stations.

The advantage of this channel access method is that it promotes fairness among stations, but its weakness is that it probably could not support time-bounded services. Fairness is maintained because each station must recontend for the channel after every transmission of an MSDU. All stations have equal probability of gaining access to the channel after each DIFS interval. For time-bounded services such as packet voice or video, delay must be maintained with a specified maximum threshold.

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With DCF, there is no mechanism to guarantee maximum delay for stations to support time-bounded services.

The PCF is an optional function providing contention free (CF) frame transfer. Fig. 1 is a sketch of the CFP repetition interval, illustrating the coexistence of the PCF and DCF.

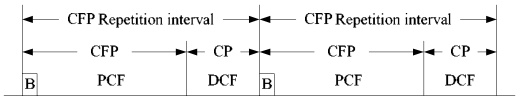

Figure 1. Coexistence of the PCF and DCF

The CFP repetition interval (*CFP_Rep*) determines the frequency with which the PCF occurs. Within a repetition interval, a portion of the time is allotted to contention-free traffic, and the remainder is provided for contention-based traffic. The maximum size of the CFP is determined by the parameter of CFP maximum duration (*CFP_Max*).

During a CFP, the AP polls voice stations on its polling list and enables them to transmit voice MPDUs without contentions. The polling scheme for the PCF is not defined in IEEE 802.11 specifications. In this paper we adopt a polling scheme based on [7].

At last in this section let us analyze the saturation throughput of the PCF.

Assume that the input load is heavy enough and the channel has no error, therefore, the utilization of PCF can be thought as 1, so the saturation throughput of PCF (say S) can be thought as the ratio of the average payload of a voice MPDU to the time needed to transmit the MPDU. Therefore, we can derive S as follows:

$$S = \frac{E[P]}{H + T[P] + \tau + SIFS} \tag{1}$$

where $E[P]$ is the mean value of payload in one MPDU, $T[P]$ the time to transmit the average payload, H is the time to transmit the head of MPDU (including MAC layer head and physical layer head), τ the propagation delay and $SIFS$ is the protocol parameter of short interframe space.

3. Simulations

In this paper, simulations are made by modeling the IEEE 802.11 network closely and carefully, including beacon, polling, DCF, PCF, and etc.

3.1. Simulation Model

Parameters used in the simulation obey the specifications in the IEEE 802.11 and are tabulated in Table 1. Besides, some assumptions are made for the simulation as follows:

•Propagation delay τ=1μs.

•The effect of bit errors in the channel and interference from the neighboring BSSs are both neglected.

•There is no "hidden terminal" and "capture" problem between users.

•No stations are operating in the "power saving" mode.

The voice stream is modeled using an ON/OFF process where the mean value of the silence (OFF) period is 1.35s and that of the talk spurt (ON) period is 1s. The voice transmission rate in the ON state is 64kb/s. The number of data traffic stations is 10. The arrival of data traffic packets to the MAC from higher layers is modeled as a Poisson process and the rate of arrival at each data station is assumed to be the same value of 7.5. The data packet length has been modeled as a truncated geometric distribution with a maximum length of 2312 octets.

Table1. Simulation parameters

Parameter	Default Value	Parameter	Default Value
Average Data MSDU length	1000 octets	Voice MSDU payload	200 octets
Channel Rate	2 Mbps	Slot Time	20 μs
Short Retry Limit	4	SIFS Time	10 μs
Long Retry Limit	7	PIFS Time	30 μs
CWmin	31	DIFS Time	50 μs
CWmax	1023	MAC header	28 octets
Beacon length	160 octets	PHY header	24 octets

3.2. Simulation results

In this section we evaluate the performance of combining voice transmission with the PCF and data transmission with the DCF. The parameters to influence the protocol performance are CFP repetition interval (*CFP_Rep*) and CFP maximum duration (*CFP_Max*). To analyze the influence of CFP duration conveniently, we define *CFP_Ratio* as the ratio of the average duration of a CFP to the duration of a *CFP_Rep*.

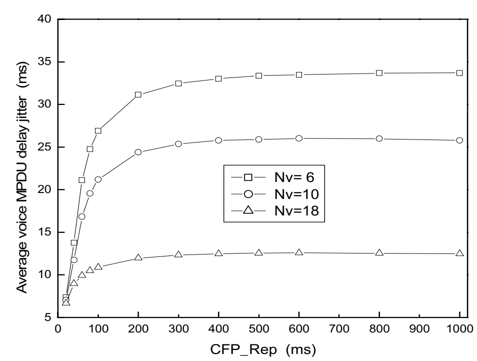

Figure 2. Average voice MPDU delay jitter versus *CFP_Rep*

In Fig. 2 and Fig. 3 where *CFP_Max*=0.8*CFP_Rep*, we plot the average voice MPDU delay jitter and the *CFP_Ratio* versus *CFP_Rep*. *Nv* is the number of voice stations. We see in Fig. 2 that the average voice MPDU delay jitter increases as *CFP_Rep* increases, or as *Nv* decreases. We find in Fig. 3 that *CFP_Ratio* increases when *CFP_Rep* decreases or *Nv* increases; that is, the CP

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duration in one *CFP_Rep* decreases relatively. Therefore, the voice MPDU delay vary slowly because the time inserted between two successive CFPs becomes shorter. This is just the reason why delay jitter decreases. We can say that, in the same case of *CFP_Rep* or *Nv*, the greater the *CFP_Ratio* is, the smaller voice traffic delay jitter becomes.

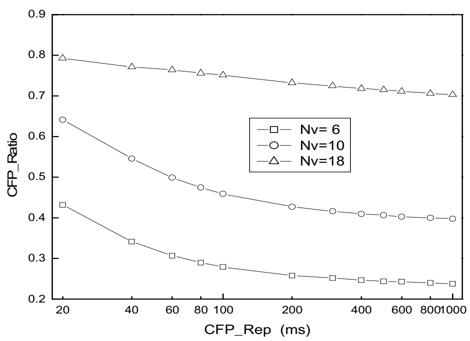

Figure 3. *CFP_Ratio* versus *CFP_Rep*

Now, we examine the effect of the CFP maximum duration on the performance. Fig. 4 depicts the average MPDU delay as a function of *Nv*, where we set *CFP_Rep*=100ms, and examine three cases of *CFP_Max*, i.e. *CFP_Max*/*CFP_Rep*=0.4, 0.6, and 0.8, respectively.

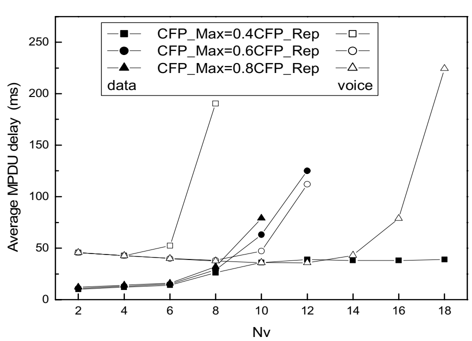

Figure 4. Average MPDU delay versus *Nv*

We here focus on the maximum number of voice stations (*Nvmax*) that can share the channel under the condition that the average voice MPDU delay is limited to the maximum allowable value (say 200ms[9,10]). From Fig. 4, we see that the value of *Nvmax* is 8, 12, and 16 for *CFP_Max*/*CFP_Rep*=0.4, 0.6, and 0.8, respectively. Consequently, we can say that as *CFP_Max* increases, *Nvmax* also becomes larger; that is, the system can accommodate a larger number of voice stations. We also find that the average voice MPDU delay does not change drastically as *Nv* increases if *Nv* does not exceed the value of *Nvmax*. This is because an increasing number of voice stations only leads to an increase of CFP duration which can be explained in Fig. 5 where we plot the *CFP_Ratio* versus *Nv*. If *Nv* is beyond *Nvmax*, the average voice MPDU delay increases rapidly because the CFP can not be extended any more and goes into a congested state.

From the perspective of the average voice MPDU delay, it is better to select a larger value of *CFP_Max*/*CFP_Rep*. It is clear for the larger the value of *CFP_Max*/*CFP_Rep*, the shorter the CP duration inserted between two successive CFPs, therefore, the smaller the average voice MPDU delay will be.

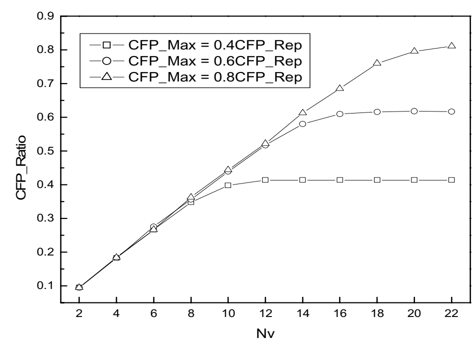

Figure 5. *CFP_Ratio* versus *Nv*

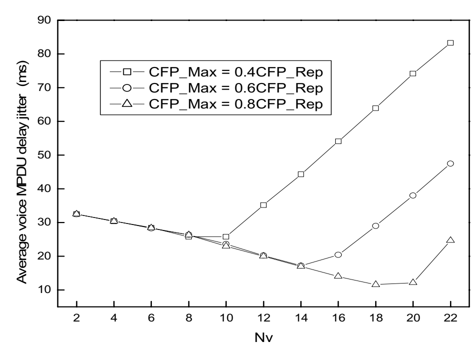

Figure 6. Average voice MPDU delay jitter versus *Nv*

Fig. 6 plots the average voice MPDU delay jitter against *Nv*. In Fig. 6 we find that there exists a threshold value of the number of voice stations (say *Nt*) in each case of *CFP_Max*/*CFP_Rep*. As *Nv* increases, the average jitter decreases for *Nv*<*Nt* while it increases fast for *Nv*>*Nt*. The value of *Nt* is 10, 14, and 18 for *CFP_Max*/*CFP_Rep* =0.4, 0.6, and 0.8, respectively. The interesting thing is that *Nt* is slightly larger than *Nvmax*. If we analyze Fig. 5 now, a conclusion can be drawn that if *Nv*<*Nt*, the CFP duration still has space to be lengthened, so *CFP_Ratio* is the main factor to influence the jitter; on the contrary, if *Nv*>*Nt*, CFP traffic is saturated and delay performance deteriorates drastically as *Nv* increases, therefore, delay increases fast and so does jitter.

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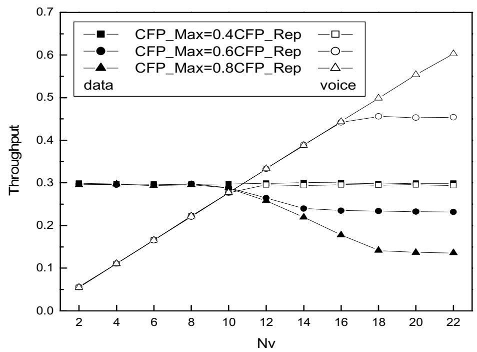

Figure 7. Throughput versus *Nv*

Next let us discuss the data performance using Fig.4 and Fig. 7 where we plot the throughput versus *Nv*. It should be noted that the data throughput decreases and the average data MPDU delay increases fast if the number of voice stations is greater than 10 for *CFP_Max*/*CFP_Rep* =0.6 and 0.8. In the case of *CFP_Max*/*CFP_Rep*=0.4, data traffic performance is fine where the value of the data throughput is very close to 0.3 and the average delay is not beyond 40ms. This is because CP duration is longer relatively and so there will be more data stations can access into channel with contention. The CP duration is shortened if *CFP_Max*/*CFP_Rep*=0.6 and 0.8, which makes the contention more serious. As a result, the data throughput is lower and the delay performance deteriorates especially when *Nv*>12.

From the results we can conclude that the value of *CFP_Max*/*CFP_Rep* should be a trade-off between the maximum number of voice stations the system can support and the data traffic performance. A small value of *CFP_Max*/*CFP_Rep* is helpful to the data traffic performance while a large value can make the maximum number of voice stations increase.

Finally, we verify the simulation results from the equation (1).

We set $E[P]$=200bytes, τ=1μs and *SIFS*=10μs as in (1). Therefore, we get the normalized saturation throughput of PCF as S=0.785.

Then, we can get the value of saturation throughput of PCF is 0.4S=31.4%, 0.6S=47.1%, and 0.8S=62.8% for *CFP_Max*/*CFP_Rep*=0.4, 0.6, and 0.8, respectively. The values are all a little larger than our numerical results in Fig. 4 if *Nv* is beyond 12, 16, and 22, due to the overhead of such control frames as beacon, CF_Poll and CF_End, which proves the simulation results in this paper correct.

4. Conclusions

This paper studied the performance of the IEEE 802.11 standard MAC protocol for integrated voice and data transmission. By simulation, we evaluated the throughput and average MPDU delay for various values of the CFP maximum duration. The simulation model we established included both data transmission with the DCF and voice transmission with the PCF. Numerical results showed that with the lengthen of the CFP maximum duration, the maximum number of voice stations supported by the protocol increases, while the performance of the data traffic deteriorates.

Especially, packet voice is sensitive to delay jitter. If the voice load is light, *CFP_Ratio* is the main factor to affect the jitter performance; that is, as *CFP_Ratio* increases due to *CFP_Rep* decreases or *Nv* increases, the jitter of packet voice decreases. Once the CFP become saturated, the delay jitter increases quickly which causes the performance to deteriorate drastically.

References

[1] IEEE 802.11 WG, Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specification, 1999.

[2] Bianchi G., "Performance analysis of the IEEE 802.11 distributed coordination function", *IEEE Journal on Selected Areas in Communications*, 18(3), 2000, pp. 535-547.

[3] Khurana S., Kahol A., Gupta S. K. S., et al., "Performance evaluation of distributed co-ordination function for IEEE 802.11 wireless LAN protocol in presence of mobile and hidden terminals", *Modeling, Analysis and Simulation of Computer and Telecommunication Systems, Proceedings*, 1999, pp. 40-47.

[4] Tien-Shin Ho, and Kwang-Cheng Chen, "Performance analysis of IEEE 802.11 CSMA/CA medium access control protocol", *PIMRC'96*, Vol 2, 1996, pp. 407-411.

[5] C. H. Foh, and Zukerman M., "Performance evaluation of IEEE 802.11", *VTC 2001*, Vol 2, 2001, pp. 841-845.

[6] Bruno R., Conti M., and Gregori E., "IEEE 802.11 optimal performances: RTS/CTS mechanism vs. basic access", *PIMRC 2002*, Vol 4, 2002, pp. 1747-1751.

[7] B. P. Crow, I. Widjaja, J. G. Kim, and P. T. Sakai, "IEEE 802.11 wireless local area networks", *IEEE Communications Magazine*, 35(9), 1997, pp. 116-126.

[8] M. Veeraraghavan, N. Cocker, and T. Moors, "Support of voice services in IEEE 802.11 wireless LANs", *INFOCOM 2001*, Vol 1, 2001, pp. 488-497.

[9] A. Zahedi, and K. Pahlavan, "Capacity of a Wireless LAN with Voice and Data Services", *IEEE Transactions on Communications*, 48(7), 2000, pp. 1160-1170.

[10] A. Zahedi, and K. Pahlavan, "Voice and data integration on TCP/IP wireless networks", *PIMRC'97*, Vol 2, 1997, pp. 678-682.

[11] H. H. Liu, and J. L. C. Wu, "Packet telephony support for the IEEE 802.11 wireless LAN", *IEEE Communications Letters*, 4(9), 2000, pp. 286-288.

[12] T. Suzuki, and S. Tasaka, "Performance evaluation of integrated video and data transmission with the IEEE 802.11 standard MAC protocol", *GLOBECOM'99*, Vol 1B, 1999, pp. 580-586.

[13] M. A. Visser, and M. E. Zarki, "Voice and data transmission over an 802.11 wireless network", *PIMRC'95*, Vol 2, 1995, pp. 648-652.

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