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Efficient Voice Communications over IEEE802.11 WLANs Using Improved PCF Procedures

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Abstract

This paper presents a new dynamically adaptable polling scheme for efficient support of voice communications over different IEEE802.11 networks. The proposed polling scheme is simple to implement and does not require any modification on the existing access protocol. The analytical approach that models the proposed polling scheme, uses a discrete-time Markov chain and proves that the specific polling scheme, when silence detection is used at the wireless terminals, improves the capability of IEEE802.11wireless LANs for handling voice traffic efficiently.

Keywords

Wireless Networks, Centralized access control, Voice communication

1. Introduction

The IEEE802.11 standard (IEEE, 1997) for wireless local area networks (WLANs) covers the Medium Access Control (MAC) sub-layer and the physical layer of the Open System Interconnection (OSI) reference model. A group of wireless terminals under the control of a *Distributed Coordination Function* (DCF) or a *Point Coordination Function* (PCF) forms a *Basic Service Set* (BSS). A BSS can either be an independent network or an infrastructure network, in which an *Access Point* (AP) links the wireless terminals to a backbone network (like Asynchronous Transfer Mode, ATM), therefore extending their range to other BSSs via other APs. In a BSS, the wireless terminals and the AP can use either DCF for asynchronous data transmissions, or PCF for contention-free packet transmissions. In DCF mode, the network is in a *Contention Period* (CP) and the stations contend in order to gain access using the *Carrier Sense Multiple Access / Collision Avoidance* (CSMA/CA) access method. In PCF mode, the AP coordinates the medium usage and the network is in a *Contention-Free Period* (CFP). The medium can be alternated between CP and CFP according to the *Contention-Free Period Repetition* (CFPR) interval that is the reciprocal of the rate at which the AP initiates the CFP.

As the speed and capacity of WLANs increase, so does the demand for improving the quality of service (QoS) of real-time applications. (Zahedi and Pahlavan, 2000) used the DCF mode in order to provide real-time applications and showed that the performance is poor by considering the delay requirements. (Romans and Tourrilhes, 1998), (Deng and Chang, 1999) and (Liu and Wu, 2000) proposed modifications to the DCF mode in order to carry packet telephony traffic. (Sheu and Sheu, 2001) proposed a novel modified protocol to provide asynchronous and multimedia traffic over IEEE802.11 WLANs that is compliant with the IEEE802.11 standard. However, this protocol was not designed for real-time traffic exchange between stations belonging to different BSSs. For real-time traffic exchange between stations from different BSSs, all real-time packets have to be transferred to the AP of the BSS and thus a centralized access method like PCF is required. Although PCF introduces high overhead in each transaction, it can satisfy time-bounded requirements if it is properly adjusted. (Crow and *et al*, 1997) showed with simulations that polling is inefficient for handling on/off speech traffic. In (Ziouva and Antonakopoulos, 2001-1) we examined how PCF can efficiently be used to carry *Constant Bit Rate* (CBR) voice traffic between stations of the same BSS, while the DCF mode supports asynchronous data traffic. The number of voice users supported by PCF can be further increased by using silence detection at each voice user and a new management scheme on the AP's polling list, the so called Cyclic Shift polling process, that does not require any modifications on the wireless terminals access mechanism (Ziouva and Antonakopoulos, 2001-2).

In this work, we extend the Cyclic Shift polling process by adding a temporary removal procedure for stations that are in silent state. The AP's polling list does not remain constant during the lifetime of a conversation, but stations that enter silent state during their call are removed from the AP's polling list temporarily and are reinstalled on the polling list after a specific duration related to the *hangover* used by the silence detector. This dynamically adaptable management scheme implemented on the AP's polling list is called *Cyclic Shift and Station Removal Polling process* (CSSR). We consider voice activities between stations from different BSSs and we estimate an upper bound on the number of voice stations that a BSS can support, while keeping low voice packet delay and guaranteeing predetermined minimum bandwidth for data traffic. The CSSR polling process is evaluated by a discrete time Markov chain and is also compared with the Cyclic Shift polling process. Our studies were performed using 32 kbps *Adaptive Differential Pulse Code Modulation* (ADPCM) voice coding. For focusing on different aspects of PCF performance, we assumed an error-free channel.

Section 2 briefly describes the PCF operation for handling voice traffic while Section 3 presents the CSSR polling scheme and its Markov model. In Section 4, we evaluate the performance of the proposed scheme deriving the maximum number of voice stations handled by PCF. Finally, Section 5 discusses some numerical results.

2. Voice Traffic Management

We consider a number of BSSs interconnected via a backbone network. The AP initiates the CFP with the transmission of a Beacon frame and terminates it with the transmission of a CF-END frame. The CFP has a maximum duration that defines the maximum number of voice stations that can be supported. Due to the DCF traffic, the CFP may be stretched. Since the actual duration of CFP and CP may vary, the IEEE802.11 standard defines a maximum value of time that the CFP is stretched and a minimum duration for CP. All timing parameters that determine the coexistence of PCF and DCF are contained in each Beacon frame. Each voice station uses an ADPCM coder at 32 kbps and a silence detector. We exploit the silent periods of a voice source in order to increase the number of calls but at the expense of voice packets dropping rate. The used voice model is the well-known model proposed in (Brady, 1969). The durations of voice talk spurts and silence periods follow the exponential distribution with average d_t equal to 400 ms and d_s equal to 600 ms respectively (Sriram and *et al*, 1999), (Onvural, 1994). The silence detectors use a technique to avoid sudden end-clipping of speech and to bridge short speech gaps such as those due to stop consonants, the so called *hangover*. This technique results to silent gaps larger than the hangover duration. A voice packet is generated every CFPR interval when the CFP is scheduled to begin. A voice packet is transmitted over the network each time the station is being polled by the AP. If a new packet is generated before the previous packet has been transmitted, the older packet is discarded. All stations on the polling list are polled once during each CFP. The voice traffic management guarantees that the voice packet delay (Ziouva and Antonakopoulos, 2001-1) is less than the CFPR interval used by the AP. A station in talk spurt generates a voice packet (called *talk packet*) that is transmitted when the AP polls the station. A station in silent period does not generate any voice packets and transmits a NULL packet (called *silent packet*) when it is polled. When the AP takes control of the medium, it starts polling according to Figure 1.

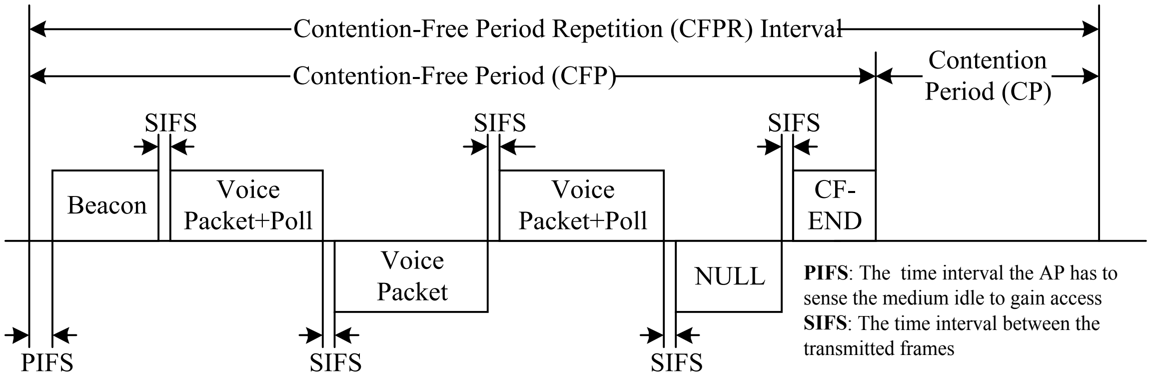

Figure 1. Voice transmissions in PCF mode

3. The CSSR polling process

The basic functionality of the dynamically adaptable polling scheme is the following: At the beginning of each CFP round, the AP cyclically shifts the stations on its polling list, so the first station at the previous round becomes the last station at the current round and all other stations advance one position towards the start of polling. At the beginning of each CFP round, the polling starts from the beginning of the shifted polling list. When the start of a silence period is detected for a station, the AP does not poll this station for a few PCF rounds. The PC maintains two polling lists, a *main polling list* that contains all stations having established connections, irrespective of their state and an *active polling list* (a subset of the main polling list) which contains only the stations that have to be polled at the next PCF round. Both polling lists are cyclically shifted in each PCF round. Whenever the AP cannot complete its active polling list during a CFP round, the AP starts its polling sequence with the first station on its polling list at the next CFP round. The advantages of the CSSR polling scheme are the following: The cyclic shift of the polling list spreads uniformly the dropped voice packet to all active stations in the network and therefore it increases the number of voice stations handled by PCF, compared to the case of using only silence detection (Ziouva and Antonakopoulos, 2001-2). The temporary station removal procedure provides more bandwidth for actual voice transmissions, thus the network performance is further improved. Finally, the AP's polling list management scheme does not require any modification on the wireless stations MAC protocol and is rather simple to implement.

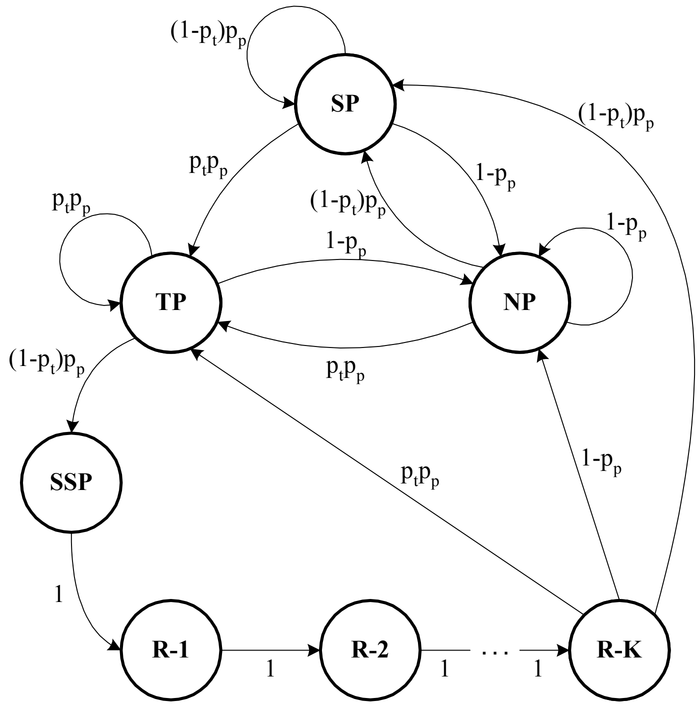

Figure 2. The CSRR polling process: Finite-state Markov chain for a voice station

For analyzing the CSSR polling scheme, a voice station is modeled as a discrete-time Markov chain that is shown in Figure 2. All state transitions occur at the end of a PCF round. During a CFP period, a voice station generates packets with probability p_t and may be polled with probability p_p. A voice station is in *Silent-Polled* (**SP**) state when it is polled and has no voice packet to transmit. The station leaves the SP state and enters the *Not-Polled* (**NP**) state when polling cannot be completed and the station is among the stations that are not polled. The station leaves the SP state and enters the *Talk-Polled* (**TP**) state when a voice packet is generated and the AP polls the station. A voice station remains in the TP state if it is in talk spurt and is polled continuously. If the station is in talk spurt and the AP does not poll it, the station moves to the NP state. If the station's talk spurt ends and the AP polls the station, the station transmits a NULL packet and moves to the *Start of Silence-Polled* (**SSP**) state and remains in this state until the next PCF round. Entering the SSP state, the station is removed from the active polling list for 1…K rounds and passes through the *Removed-i* (**R-i**) states. After K rounds, the station is repositioned on the polling list and, the station returns either to the NP, TP or SP state, depending on its position in the polling list and the transmission of a voice or a NULL packet. If Π_X is the steady state probability of a station being at state X, we can calculate the stationary distribution of the discrete Markov chain if the probabilities p_t and p_p are know.

$$\Pi_{TP} = \frac{p_p p_t}{1+Kp_p^2 p_t\left(1-p_t\right)}, \quad \Pi_{SP} = \frac{p_p\left(1-p_t\right)\left(1-p_p p_t\right)}{1+Kp_p^2 p_t\left(1-p_t\right)}, \quad \Pi_{NP} = \frac{1-p_p}{1+Kp_p^2 p_t\left(1-p_t\right)}$$

$$\Pi_{SSP} = \frac{p_p^2 p_t\left(1-p_t\right)}{1+Kp_p^2 p_t\left(1-p_t\right)}, \quad \Pi_R = \frac{Kp_p^2 p_t\left(1-p_t\right)}{1+Kp_p^2 p_t\left(1-p_t\right)}$$

Π_R is the steady state probability of a station being removed from polling list for K PCF rounds. If $K = 0$ (a polling scheme without station removal) the CSRR polling process represents the Cyclic Shift polling process.

4. Performance Evaluation

In this section, an analytical approach is presented to derive an upper bound to the number of voice stations that can be handled by PCF, when silence detection and the CSSR polling scheme are employed.

4.1. The voice packet dropping probability

Since the CFPR interval can be selected so that the voice delay requirements are always satisfied, the parameter that defines the PCF performance is the probability of dropped packets P_{drop}. According to (Sriram and *et al*, 1999) and (Jayant and Christensen, 1981), 1 % of dropped voice packets loss can be tolerated. In this case, we can estimate an upper bound to the number of voice stations accommodated by PCF for various values of K, while the condition $P_{drop}< 0.01$ is satisfied. In order to decrease the number of rejected packets when a station is removed from the polling list, the duration of the station removal is limited to the hangover duration. Furthermore, for two connected stations A and B, when e.g. station A is in silent station B is usually in talk spurt, thus the AP of station A has to send the packets received by the AP of station B, to station A when this station is removed from its polling list. Under these circumstances, packets from station A are dropped either when it is in talk spurt and its AP does not poll it, or when the station is in talk spurt and is polled by its AP but the AP of station B does not poll station B, therefore the packets of station A are not delivered to station B. So, the probability of dropped voice packets is given by:

$$P_{drop}^{(A)} = \Pi_{NP}^{(A)} p_t + \Pi_{TP}^{(A)} \Pi_{NP}^{(B)} \tag{1}$$

Since, the steady state probabilities Π_X are functions of probabilities p_t and p_p and these two probabilities are considered equal for the two communication stations, the probability of dropped voice packets can be calculated if p_t and p_p are known.

4.2. The basic system probabilities

According to (Fine and Tobagi, 1986) and (Friedman and Ziegler, 1989), the probability p_t that a voice station is in talk spurt is given by $p_t = d_t/(d_t + d_s)$. If p_{np} is the probability that a station is not polled, then p_p will be derived by: $p_p = 1 - p_{np}$. The probability p_{np} depends on the number N of voice stations that form the main polling list (N stations generate $2N$ packets, uplink and downlink), the number N_r of stations that do not participate in the active polling list during a PCF round (remaining $2N - N_r$ packets to be exchanged) and the number N_p of stations that are polled during a PCF round. Let $N_{t\max}$ denotes the maximum number of talk packets that can be handled by PCF. If $2N - N_r \leq N_{t\max}$, every station is polled during a PCF round, but if $2N - N_r > N_{t\max}$, only N_p stations can be polled, the stations holding the first N_p positions on the active polling list, while the rest ($N - N_r - N_p$) stations are not polled. So the conditional probability $p_{np|Nr,Np}$ that a station is not polled during a PCF round when there are N_r and N_p stations is given by:

$$p_{np|N_r,N_p} = \left(N - N_r - N_p\right) \big/ \left(N - N_r\right) \qquad \text{if} \quad 0 \leq N_r \leq 2N - N_{t\max} - 1$$

For a given number N_r, the number of stations that can be polled has a maximum value $N_{p\max}$ depending on the CFP length and the combination of the number of talk packets and the number of silent packets. The AP polls either $N_{p\max}$ stations or $(N–N_r)$ stations, whatever of the two events happens first. Finding the probability $P_{Np|Nr}$ that the AP polls N_p stations given that it has removed N_r stations and the probability P_{Nr} that the AP has removed N_r stations, and using the total probability theorem and the above equation, we have that

$$p_{np} = \sum_{N_r=0}^{2N-N_{t\max}-1} \sum_{N_p=\lfloor (N_{t\max}-N_r)/2 \rfloor}^{\min(N_{p\max}, N-N_r)} \frac{N-N_r-N_p}{N-N_r} P_{N_p|N_r} P_{N_r}$$

Let assume that a station is removed from the polling list with probability p_r. Then the N_r stations during a PCF round have a binomial mass function and so

$$P_{N_r} = \binom{N}{N_r} p_r^{N_r} (1-p_r)^{N-N_r}$$

The probability p_r can be calculated by Π_R, where p_p is equal to $N_p/(N-N_r)$, since during a PCF round a station can be at one of the N_p first positions of $(N-N_r)$ stations on the active polling list. In order to find the probability $P_{Np|Nr}$, we must calculate the parameters $N_{t\max}$ and $N_{p\max}$. The maximum number $N_{t\max}$ of talk packets that the AP can handle depends on the CFP maximum duration and the talk packet transmission time T_t, which is equal to T_{vp} + *SIFS*, where T_{vp} is the transmission time of a voice packet including headers. Therefore:

$$N_{t\max} = \left\lfloor \left(T_{CFPR} - T_{\max FS} - PIFS - T_{Beacon} - SIFS - T_{CF-END} - T_{CP}\right) / T_t \right\rfloor$$

where T_{CFPR}, $T_{\max FS}$, T_{BEACON}, $T_{CF\text{-}END}$ and T_{CP} are the CFPR interval duration, the maximum time the CFP can be stretched, the Beacon and CF-END frame transmission time and the CP duration respectively. The different BSSs use the same CFPR interval duration, therefore the same transmission time is used for both uplink and downlink talk packets.

The maximum number $N_{p\max}$ of stations that can be polled during a PCF round can be found when all exchanged packets are silent packets. Depending on T_t/T_s (T_s is the silent packet transmission time which equals the transmission time of a NULL packet plus the *SIFS* time), a talk packet can be replaced by one or more silent packets. The AP defines the remaining CFP duration using time information from the previously transmitted packets, since it does not know if the next station in the list is in talk or silent state. The AP considers that the downlink and uplink packets of the last station that can be polled are talk packets and thus it guarantees that the maximum duration of the CFP is not exceeded. This explains why we subtract two talk packets from $N_{t\max}$. Finally, we have to subtract the N_r downlink packets that have to be sent to the removed silent stations. So $N_{p\max}$ is given by:

$$N_{p\max} = \left\lfloor \left((N_{t\max} - N_r - 2) T_t / T_s\right) / 2 + 1 \right\rfloor$$

Finally, the probability $P_{Np|Nr}$ is determined by the number N_t of talk packets and the number $(2N_p + N_r - N_t)$ of silent packets during a CFP. Thus:

$$P_{N_p|N_r} = \sum_{N_t=0}^{\min(N_{t\max}, 2N_p+N_r)} \left[\binom{2N_p+N_r}{N_t} - g\left(N_p, N_r, N_t, N_{t\max}\right)\binom{2N_p+N_r-2}{N_t}\right] \cdot p_t^{N_t}(1-p_t)^{2N_p+N_r-N_t} f\left(\left\lceil (2N_p+N_r-N_t) T_s / T_t \right\rceil + N_t - N_{t\max}\right)$$

where $f(x)$ and $g(x)$ are two functions that define the permitted combinations of N_t talk packets and $(2N_p + N_r - N_t)$ silent packets during CFP and are given by the following two equations:

$$f(x) = \begin{cases} 1 & \text{if} \quad x = 0 \\ 0 & \text{if} \quad x \neq 0 \end{cases}$$

$$g\left(N_p, N_r, N_t, N_{t\max}\right) = \\ = \begin{cases} 0 \quad \text{if} \quad 2N_p + N_r = N_{t\max} \vee N_t = 0 \\ \qquad \vee N_t + 2N_p + N_r - N_{t\max} < N_{t\max} \\ \qquad \vee \left(\left(N_t + 2N_p + N_r - N_{t\max} = N_{t\max}\right) \wedge \left(\left(N_t + 2\right)T_t + \left(2N_p + N_r - N_t - 2\right)T_s \leq N_{t\max}T_t\right)\right) \\ 1 \quad \text{if} \quad N_t + 2N_p + N_r - N_{t\max} > N_{t\max} \\ \qquad \vee \left(\left(N_t + 2N_p + N_r - N_{t\max} = N_{t\max}\right) \wedge \left(\left(N_t + 2\right)T_t + \left(2N_p + N_r - N_t - 2\right)T_s > N_{t\max}T_t\right)\right) \end{cases}$$

5. Numerical Results

According to our analytical approach, the effect of CSSR polling scheme on the PCF performance is depicted in Figure 3. The supported number of voice stations increases when the proposed scheme is used along with silence detection in contrast to the case of CBR voice traffic. Furthermore, the throughput improves when the CFPR interval and the parameter K increase, but K must not exceed the hangover duration. The results of our analysis are derived considering the attribute values used by the high data rate extension of the IEEE802.11 standard at 5.5 and 11 Mbps. We also use the optional short physical header defined in the standard for reaching maximum throughput.

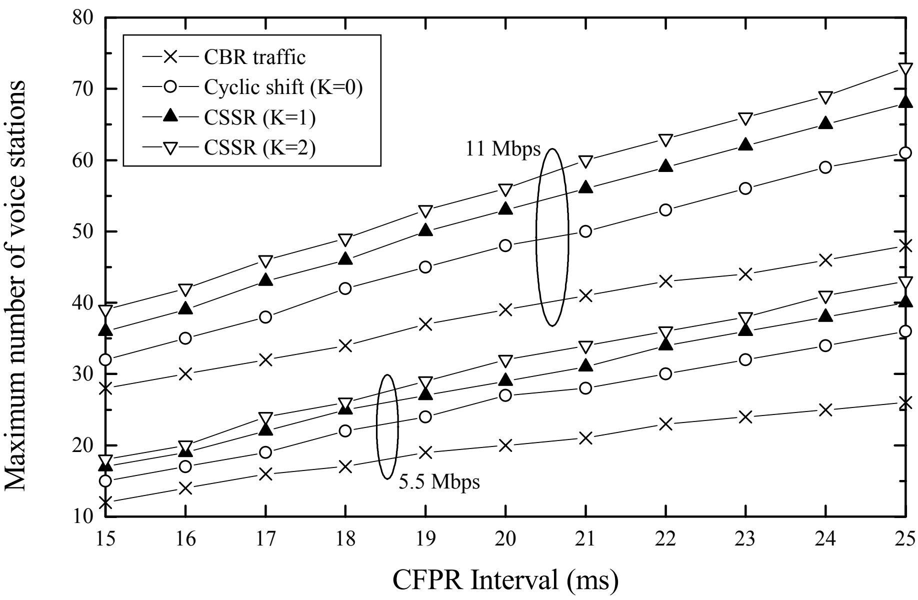

Figure 3. The effect of CSSR on the maximum number of supported voice stations

6. Conclusions

In this paper, we presented the CSSR polling scheme that can be used along with silence detection for increasing the number of supported voice communications over different IEEE802.11 WLANs. The proposed scheme improves the PCF performance by cyclically shifting the stations on the AP's polling list in each PCF round and dynamically removing stations that enter silence mode. The efficiency of the CSSR polling scheme is a function of the CFPR interval used by the network and the hangover duration of the silence detectors, which also determines the maximum duration of temporary station removal. Finally, the CSSR scheme can be easily implemented on the AP of an IEEE802.11 network, without requiring any modification on the wireless terminals access mechanism.

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