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Abstract: The authors propose a cyclic shift polling method that can be used along with silence detection on IEEE802.11 wireless LANs for increasing the number of supported voice communications. The method spreads the total voice packet rejection rate to all active stations in order to increase the number of stations that experience specific voice quality. The cyclic shift polling method is implemented only at the access point of each wireless LAN and does not require any modifications to the existing access protocol. Analytic expressions and comparative numerical results are presented for the performance of the IEEE802.11 point co-ordination function. The analysis shows how the cyclic shift polling method along with silence detection on the mobile stations can be used for increasing the number of supported voice calls.

1 Introduction

The IEEE802.11 wireless LANs standard specifies the use of distributed co-ordination function (DCF) and point co-ordination function (PCF) in its medium access control (MAC) sublayer architecture [1]. DCF was developed for asynchronous data transmission, where all stations share the medium using the carrier sense multiple access with collision avoidance (CSMA/CA) protocol and a random backoff mechanism. PCF was developed for supporting time-bounded services, where a point co-ordinator (PC), the access point (AP) of the basic service set (BSS), determines which station has the right to transmit. The performance of DCF for asynchronous data transfers was investigated in depth in [2–4], but little work has been done to model the performance of PCF in the case of real-time traffic.

PCF has been used for packet voice transmission [5–8] and the simulation results show that voice packets suffer from large delays and the supported number of voice stations is small, while [9–11] report the use of DCF for data and voice transmission requiring considerable modifications to the IEEE802.11 specifications. In this paper, communications between voice stations using silence detection and belonging to different BSSs are considered. DCF is used for data traffic and signalling messages, while PCF is used for voice traffic by utilising an efficient polling scheme at the point co-ordinator of each BSS. We estimate an upper bound to the number of voice calls a BSS can handle, while keeping the overall rate of dropped voice packets below 0.01 and guaranteeing a minimum bandwidth for data traffic. Further, we estimate the voice packet delay distribution inside each BSS, from the time instant a voice packet is generated until the PC receives the packet. In our analysis, we assume that the channel is error-free, there are no hidden terminals, the effect of propagation delay is negligible, there are no overlapping BSSs and the channel bit rate is 5.5 or 11 Mbit/s [12].

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2 System description

A BSS network is considered that supports PCF and DCF functions by using a time-sharing mechanism, as shown in Fig. 1. The period of time the DCF operates is called the contention period (CP), implying that the stations contend for access, while the period of time the PCF operates is called the contention-free period (CFP), implying that the stations do not contend for access. The start of CFP is determined at specific time instances according to the contention-free repetition interval (CFPRI). The PC determines the CFP interval length according to the available traffic and the size of its polling list. Each CFP starts with the transmission of a *Beacon* packet and terminates by a *CF–END* packet at or before a maximum duration, called *CFPmaxDuration*. If at the nominal *Beacon* transmission time, called the target beacon transmission time (TBTT), the medium is busy due to DCF traffic, then the *Beacon* packet is delayed and the CFP is reduced. In this case, the PC ends the CFP no later than TBTT plus the value of *CFPmaxDuration*. Since the actual duration of CFP and CP may vary, the selection of *CFPmaxDuration* must allow a minimum duration of CP at which at least one data frame may be sent. All the time parameters that define the coexistence of PCF and DCF are contained in the *Beacon* packet.

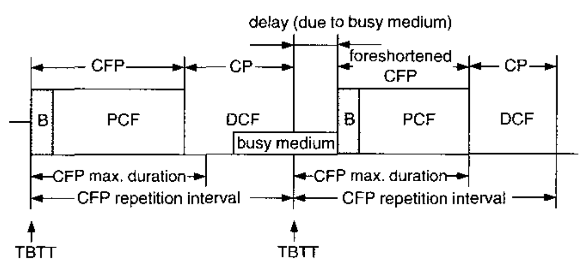

Fig. 1 *CFP/CP time-sharing mechanism*
B = Beacon

Each voice station desiring to make a voice call issues a request to the PC and if the call is accepted by the PC and the called station, the two stations start exchanging voice packets. Based on the voice call requests, the PC formats a polling list with the stations belonging to its BSS and having established a connection. The PC starts a CFP round by transmitting a *Beacon* packet when it detects the channel idle for PCF interframe space (PIFS). After a short interframe space (SIFS), the PC sends a *CF–Poll* packet to the first station on its polling list. If the PC has a packet for the station that has to be polled, it sends a *Data+CF–Poll* packet. No later than SIFS time after receiving the *CF–Poll* or the *Data+CF–Poll* packet, the station sends its voice packet, as a *Data* or *Data+CF–Ack* packet to the PC. The *Data+CF–Ack* type is the response to the *Data+CF–Poll* and includes the voice packet and an acknowledgment to the preceding *Data* packet. If the polled station has no packet to send, it transmits a *Null* packet in the case of *CF–Poll* or a *CF–Ack* in the case of *Data+CF–Poll*. The PC acknowledges the previously received voice packet, if any, and gives the right to transmit to the next station on its polling list. Fig. 2 depicts the rules under which voice packets are transmitted during CFP. The PC polls station A without transmitting any *Data* packet, while station A transmits its own *Data* packet. The PC polls station C and transmits its *Data* packet simultaneously, while station C responds by transmitting its own *Data* packet and the acknowledgment of the previously received packet.

In our analysis, we consider that each voice station generates a voice packet every TBTT seconds and the voice packet generation interval is equal to the CFPR interval. Receiving the *Beacon* packet, which contains all the required timing information, the stations determine the TBTT instances and update their clocks accordingly. A voice packet is transmitted each time the station is polled by the PC. Since the transmission of a *Beacon* packet may be delayed due to DCF traffic, each voice packet suffers a variable delay until the station is polled, as shown in Fig. 3. For constant bit rate voice traffic, if the number of stations on the polling list is determined by the available duration of CFP for a given CFPR interval and all stations are served once per CFP, then the voice delay of all stations is upper bounded by the CFPR interval without any packet loss rate. Using silence detection and a maximum rate of rejected voice packets, the number of supported voice calls can be increased. Depending on the number of stations in talk-spurt and silence mode and the actual CFP duration, the stations at the end of the polling list experience more rejections, since a new packet forces the station to delete the packet previously not transmitted. In order to provide voice quality comparable to the telephone network, the packet rejection rate for each station must be less than 0.01 [13, 14].

3 PCF analysis

Initially, the PCF performance is evaluated when constant bit rate voice coders are used at the mobile stations. Let T_{CFPR}, T_{CFP} and T_{CP} denote the duration of CFPR, CFP and CP intervals, respectively. Then, according to Fig. 1

$$T_{CFPR} = T_{CFP} + T_{CP} \tag{1}$$

The values of T_{CFP} and T_{CP} may vary but their sum remains constant. A minimum bandwidth is reserved for DCF, which is determined by the IEEE802.11 standard as

$$T_{\min CP} = T_{\max MPDU} + 2SIFS + 2a + 8T_{Ack} + DIFS \tag{2}$$

where a is the *SlotTime* time unit used by the mobile terminals for updating their backoff counters, while $T_{\max MPDU}$ and T_{Ack} are the transmission times of a maximum length data packet (transmitted during DCF) and of an Ack packet (acknowledgment to a received data

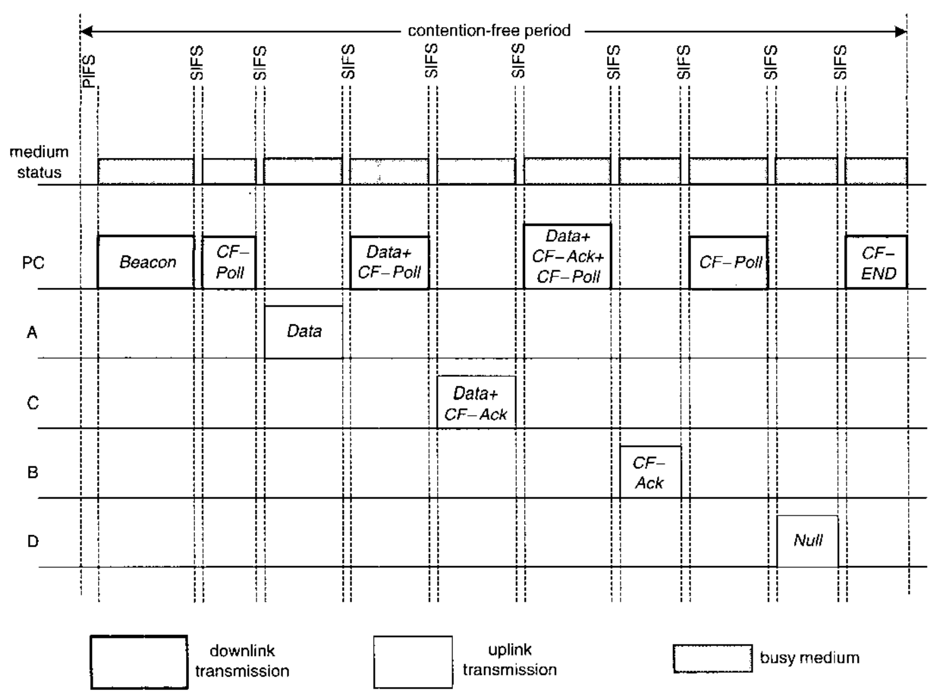

Fig. 2 *Voice packets polling during CFP*

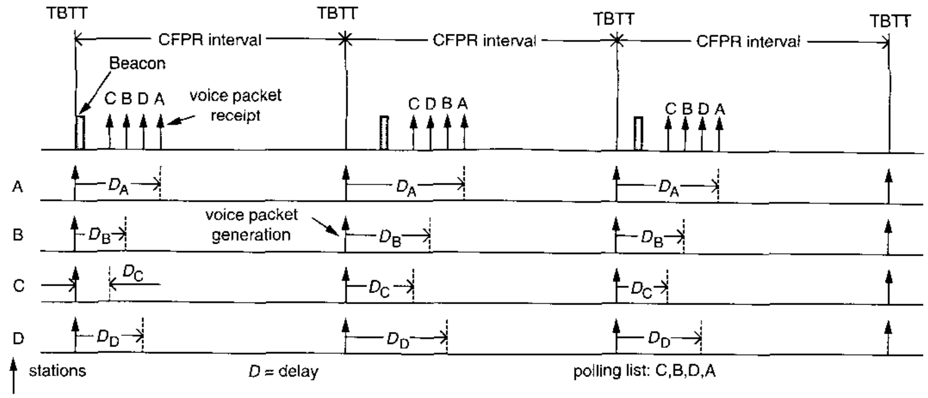

Fig. 3 *Voice traffic management*

packet), respectively. DIFS is the interframe time that the channel must be idle before a station starts its transmission during DCF.

Since CFP may be reduced temporarily due to asynchronous traffic, T_{FS} is defined as a random variable that represents the CFP start delay. According to [1], the maximum value of this random variable is

$$T_{\max FS} = T_{RTS} + T_{CTS} + T_{\max MPDU} + T_{Ack} + 3SIFS \quad (3)$$

where T_{RTS} and T_{CTS} are the transmission times of RTS and CTS packets. To avoid rejection of voice packets during the reduced CFP, the PC must be able to poll all stations on its list. The duration $T'_{CFP} = T_{\max CFP} - T_{\max FS}$ of the reduced CFP determines the maximum number of voice stations the PC can handle, and T_{CFPR} becomes

$$T_{CFPR} = T_{\max FS} + T'_{CFP} + T_{\min CP} \quad (4)$$

When stations of different BSSs exchange voice packets, there is downlink traffic (from the PC to the station) and uplink traffic (from the station to the PC) inside each BSS, and the total time required to exchange both packets T_{ex} is given by

$$T_{ex} = T_{down} + T_{up} + 2SIFS \quad (5)$$

For constant bit rate voice traffic, T_{down} is the transmission time of a *Data+CF–Poll* or a *Data+CF–Ack+CF–Poll* packet and T_{up} is the transmission time of a *Data* or a *Data+CF–Ack* packet, as shown in Fig. 2 (e.g. PC to station C). Let $N_{\max}$ denote the maximum number of voice stations that can be handled during PCF, then

$$T'_{CFP} = PIFS + T_{Beacon} + N_{\max} T_{ex} + SIFS + T_{CF-END} \quad (6)$$

where T_{Beacon} and T_{CF-END} are the transmission times of the *Beacon* and *CF–END* packets, respectively.

Using (4) and (6), we can calculate the maximum number $N_{\max}$ of voice stations an IEEE802.11 BSS network can support for various values of the CFPR interval, which is

$$N_{\max} = \left\lfloor \frac{T_{CFPR} - T_{\max FS} - PIFS - T_{Beacon} - SIFS - T_{CF-END} - T_{\min CP}}{T_{ex}} \right\rfloor \quad (7)$$

The total delay the voice packets experience depends on the station's position on the polling list. As shown in Fig. 3, the last station on the polling list suffers the highest delay. Since the CFPR interval determines the maximum CFP duration, the delay of the last station D_{LS} depends on the CFPR interval duration and is given by

$$D_{LS} = T_{\max FS} + PIFS + T_{Beacon} + N_{\max} T_{ex} \quad (8)$$

Finally, the available bandwidth BW_d for asynchronous data transfers (including the $T_{\max FS}$ time) is derived by

$$BW_d\% = \frac{T_{CFPR} - PIFS - T_{Beacon} - N_{\max} T_{ex} - SIFS - T_{CF-END}}{T_{CFPR}} \cdot 100 \quad (9)$$

4 Voice transmission using silence detection and cyclic shift polling

A voice source alternates between talk spurts and silence. By employing silence detection in a mobile terminal, the PCF can handle more voice transmissions since a station in a talk spurt generates voice packets periodically, but no voice packets are generated during a silent period. In the following analysis, we study how the silence detection can be used for increasing the number of supported voice calls and how the polling process can be modified to further increase the number of supported conversations.

According to [6] and [13], the duration of a voice talk spurt follows the exponential distribution with average value $1/\lambda$, while the duration of silent periods also follows the exponential distribution with average value $1/\mu$. Therefore, the probability p that a voice station is in a talk spurt is given by $p = \mu/(\lambda + \mu)$. If at a TBTT instant a station is in a talk spurt, at the next PCF round the station transmits a voice packet when it is polled. If at a TBTT instant a station is in silence mode, no voice packet is generated at the next PCF round and a *Null* or a *CF–Ack* packet is transmitted when the station is polled. Therefore, if the station is in talk-spurt mode, either a *Data* or a *Data+CF–Ack* packet is transmitted during an uplink transmission. If the station is in a silent state, a *Null* or a *CF–Ack* packet is transmitted. During a downlink transmission, either a *Data+CF–Poll* or a *Data+CF–Ack+CF–Poll* packet is transmitted if the station is in talk-spurt mode, otherwise a *CF–Poll* or a *CF–Ack+CF–Poll* packet is transmitted. Since the *Null* (*CF–Ack*) and the *CF–Poll* (*CF–Ack+CF–Poll*) packets have the same size, the transmission time of a packet indicating a station in a silent state is equal to the transmission time T_{Null} of a *Null* packet, independent of the direction of transmission.

In our model, we assume that all BSSs use the same CFPRI value. Therefore, when the mobile stations are in talk-spurt mode they generate voice packets of equal length. The *Data* (*Data+CF–Ack*) and *Data+CF–Poll* (*Data+CF–Ack+CF–Poll*) packets have the same size for both directions of transmission. When a *Null* packet T_{Null} is transmitted, the total required time is $T_s = SIFS + T_{Null}$, while $T_t = SIFS + T_{Data}$ is the total required time for the transmission of a voice packet (*Data* packet) T_{Data}. We also consider that a station makes, at most, one state transition during a CFPR interval, since the duration of the CFPR interval is always much lower than the time between two consecutive state transitions.

During each CFP period we consider that there are N voice stations having established connections in a BSS and the number of packets exchanged during CFP is $2N$ (downlink and uplink). Let N_t of the $2N$ packets be due to stations in a talk state. According to [14], each station can be in a talk state with probability p, independently of all other stations. Although this assumption does not take into account the speaker interaction during a conversation, it is used in most cases [5, 6, 8] and [14] since it does not add serious complexity to the system model and does not affect the model's accuracy significantly. In this case, N_t is a random variable ($N_t \leq 2N$) having a binomial probability mass function

$$P[N_t = n] = \binom{2N}{n} p^n (1-p)^{2N-n} \qquad (10)$$

So, the duration of each CFP period is determined by N_t and $2N$.

4.1 *Effect of silence detection on the number of supported conversations*

In the following analysis, it is shown how silence detection increases the number of supported conversations N ($N > N_{\max}$), when the maximum available bandwidth is used for PCF. Let k denote the increase in the number of conversations, so that the number of packet exchanges during CFP becomes

$$2N(k) = 2N_{\max} + 2k, \quad k = 1, 2, 3, \ldots \qquad (11)$$

During a CFP period, if N_t stations are in talk-spurt mode, $2N$–N_t stations are in silence. If $N_tT_t + (2N - N_t)T_s \leq 2N_{\max}T_t$, then the PC can complete its polling list during a CFP round, otherwise some stations are not polled during the current round. This inequality remains valid as long as the number of talk stations remains less or equal to a maximum value $N_{t\max}$. If $N_t > N_{t\max}$, the PC cannot complete its polling list during a CFP round, rejects the stations that cannot be served and restarts the polling sequence with the first station on its polling list at the next CFP round. The rejected stations experience variable packet loss rate, with the last station experiencing the highest loss rate. A bound of 0.01 packet loss can be tolerated and this bound determines the increase in the number of voice stations handled by PCF when silence detection is used. Since the voice packets pass through two BSSs for end-to-end transmission, the packet drop rate inside each BSS must be limited to 0.005.

Let k_m be the value of k that constrains the increase in the number of conversations to an upper limit so that the maximum duration of CFP is not exceeded for $N_{t\max}(k)$ stations in talk-spurt mode and $2N(k)$–$N_{t\max}(k)$ silent stations. Since

$$\begin{aligned} T_{CFPR} = {} & T_{\max FS} + PIFS + T_{Beacon} + N_{t\max}(k)T_t \\ & + (2N(k) - N_{t\max}(k))T_s \\ & + SIFS + T_{CF-END} + T_{\min CP} \end{aligned} \qquad (12)$$

$N_{t\max}$ is determined by

$$N_{t\max}(k) = \left\lfloor \frac{T_{CFPR} - T_{\max FS} - T_{\min CP} - PIFS - T_{Beacon} - SIFS - T_{CF-END} - 2N(k)T_s}{T_t - T_s} \right\rfloor; \quad k \leq k_m \qquad (13)$$

Let NP_{LS} denote the probability that the last station on the list is not polled. If $N_t \leq N_{t\max}$, then $NP_{LS} = 0$. If $N_t > N_{t\max}$, the last station on the polling list is rejected and $NP_{LS} > 0$. If the last station is not polled, then the data from the associated station are also rejected and the $N-1$ previously polled stations on the list result in the exchange of $2N$–2 packets ($N_{t\max} - 1$ or more correspond to stations in talk-spurt mode). The probability NP_{LS} is given by

$$NP_{LS}(k) = \sum_{n=N_{t\max}(k)-1}^{2N(k)-2} \binom{2N(k)-2}{n} p^n (1-p)^{2N(k)-2-n}, \quad k = 1, 2, \ldots, k_m \qquad (14)$$

If the last station is in talk-spurt mode and is not polled, then

$$\Phi_{LS}(k) = pNP_{LS}(k), \quad k = 1, 2, \ldots k_m \qquad (15)$$

where Φ_{LS} denotes the probability that a voice packet is rejected by the last station on the polling list. The quality of the reconstructed voice remains high for these values of k that satisfy the condition $\Phi_{LS}(k) < 0.005$. From these values we determine the value k_n that corresponds to the maximum number of voice stations $N(k_n)$ the PCF can handle when silence detection is used.

If NP_{LS-i} is the probability that the station, which is i-positions before the last station on the polling list, is not polled, where

$$1 \leq i \leq \left\lfloor \frac{1}{2} \frac{2N(k) - N_{t\max}(k) - 2}{2} \right\rfloor$$

then

$$NP_{LS-i}(k) = \sum_{n=N_{t\max}(k)+2i-1}^{2N(k)-2-2i} \times \binom{2N(k)-2-2i}{n} p^n (1-p)^{2N(k)-2-2i-n} \qquad (16)$$

The probability Φ_{LS-i} that the station which is i positions before the last station on the polling list rejects a voice packet is given by

$$\Phi_{LS-i}(k) = pNP_{LS-i}(k) \qquad (17)$$

In the rest of this Section, a study is made of the delay that the voice packets experience when silence detection is used. In this analysis we do not take into account the delay introduced due to the voice packet formulation, which is constant and equal to CFPRI and does not affect the delay distribution. The delay is considered from the voice packet generation instant until the PC receives this voice packet. We use the superscript *sd* on the following quantities to indicate the use of silence detection for increasing the number of voice stations. All the quantities with this superscript are derived for $k=k_n$. For example $N^{sd}_{\max}=N(k_n)$ is the maximum number of voice stations that the BSS can support and $N^{sd}_{t\,max}=N_{t\max}(k_n)$ is the maximum number of voice stations in talk-spurt mode, corresponding to $2N^{sd}_{\max}$. The delay D^{sd}_{LS} of the last station depends on the number N_t of stations that are in talk-spurt mode and is given by

$$D^{sd}_{LS}(N_t) = T_{\max FS} + PIFS + T_{Beacon} + N_tT_t + (2N^{sd}_{\max} - N_t)T_s, \quad 1 \le N_t \le N^{sd}_{t\max} \tag{18}$$

If $P[\text{LS polled and talk}]=(1-NP^{sd}_{LS})p$ is the probability the last station is polled and is in talk-spurt mode, the conditional probability $P[D^{sd}_{LS}(N_t)|\text{LS polled and talk}]$ that the delay of the last station is $D^{sd}_{LS}(N_t)$ is calculated by

$$P\left[D^{sd}_{LS}(N_t)|\text{LS polled and talk}\right] = \begin{cases} \dfrac{p\binom{2N^{sd}_{\max}-1}{N_t-1}p^{N_t-1}(1-p)^{2N^{sd}_{\max}-N_t}}{(1-NP^{sd}_{LS})p} & \text{if } 1 \le N_t \le N^{sd}_{t\max}-2 \\ \dfrac{p(1-p)\binom{2N^{sd}_{\max}-2}{N_t-1}p^{N_t-1}(1-p)^{2N^{sd}_{\max}-N^{sd}_{t\max}-1}}{(1-NP^{sd}_{LS})p} & \text{if } N_t = N^{sd}_{t\max}-1 \\ \dfrac{p^2\binom{2N^{sd}_{\max}-2}{N_t-2}p^{N_t-2}(1-p)^{2N^{sd}_{\max}-N_t}}{(1-NP^{sd}_{LS})p} & \text{if } N_t = N^{sd}_{t\max} \end{cases} \tag{19}$$

Using (19), we can calculate the delay complementary probability distribution $P[D^{sd}_{LS}(N_t)>t]$ of the last station

$$P\left[D^{sd}_{LS}(N_t)>t\right] = \sum_{\forall N_t:\, D^{sd}_{LS}(N_t)>t} P\left[D^{sd}_{LS}(N_t)|\text{LS polled and talk}\right] \tag{20}$$

As shown in the above analysis, the voice packet rejection rate and the introduced delay are not equally distributed to all stations of the polling list and, thus, the need for a different polling scheme becomes meaningful.

4.2 Cyclic shift polling scheme

In order to spread the rejected voice packets to all active stations uniformly, a more efficient management of the polling list is proposed, which is implemented at the access point and does not require any modification on the mobile terminals. According to the so called cyclic shift polling scheme, at the beginning of each CFP round, the PC cyclically shifts the stations in the polling list, so that the first station in the previous round becomes the last station in the current round and all other stations advance one position towards the start of the list.

The probability p_s that a station of the cyclic shift polling process is at a given position in the polling list is given by $p_s(k)=1/N(k)$, where $N(k)$ is the number of voice stations of a BSS having established connections and is given by (11). The probability NP_s that a station is not polled is the same for all stations and depends on p_s and NP_{LS-i}. This is also true for the probability of dropped voice packets Φ_s. According to (13), which gives $N_{t\max}(k)$ and for

$$1 \le i \le \left\lfloor \frac{1}{2}\cdot\frac{2N(k)-N_{t\max}(k)-2}{2} \right\rfloor :$$

$$NP_s(k) = p_s(k)\left(NP_{LS}(k) + \sum_{\forall i} NP_{LS-i}(k)\right) \tag{21}$$

$$\Phi_s(k) = p_s(k)\left(\Phi_{LS}(k) + \sum_{\forall i} \Phi_{LS-i}(k)\right) \tag{22}$$

Substituting (15) and (17) in (22), Φ_s can be calculated for various values of k. We determine the values of k that satisfy the voice quality condition $\Phi_s(k)<0.005$. If k_l is the highest value of k for which the voice quality condition is satisfied, then the number $N^{sh}_{\max}=N(k_l)$ is the maximum number of voice stations the PCF can handle when silence detection is used with cyclic shift polling. The superscript '*sh*' indicates the use of silence detection with cyclic shift polling.

The average number of stations talking during CFP that corresponds to $N^{sh}_{\max}$ stations, is given by

$$E\left[N^{sh}_t\right] = \sum_{n=0}^{N^{sh}_{t\max}} n\binom{2N^{sh}_{\max}}{n}p^n(1-p)^{2N^{sh}_{\max}-n} + \sum_{n=N^{sh}_{t\max}+1}^{2N^{sh}_{\max}} N^{sh}_{t\max}\binom{2N^{sh}_{\max}}{n}p^n(1-p)^{2N^{sh}_{\max}-n} \tag{23}$$

where $N^{sh}_{t\max} = N_{t\max}(k_l)$. In this case, the average duration of the CFP period is

$$E\left[T^{sh}_{CFP}\right] = PIFS + T_{Beacon} + SIFS + T_{CF-END} + E\left[N^{sh}_t\right]T_t + \left(2N^{sh}_{\max} - E\left[N^{sh}_t\right]\right)T_s \tag{24}$$

Therefore, the bandwidth remaining for asynchronous traffic is given by

$$BW^{sh}_d\% = \frac{T_{CFPR} - E\left[T^{sh}_{CFP}\right]}{T_{CFPR}}\cdot 100 \tag{25}$$

Although the mean delay is equal for all stations, the delay introduced in each CFP round depends on the current position j of the station (where $1\le j\le N^{sh}_{\max}$) and the number N_t of stations in talk-spurt mode (where $1\le N_t\le 2j$). Therefore,

$$D^{sh}(j,N_t) = T_{\max FS} + PIFS + T_{Beacon} + N_tT_t + (2j-N_t)T_s \tag{26}$$

The probability that the station is in talk-spurt mode and is polled is given by $P[\text{polled and talk}]=(1-NP^{sh}_s)p$ using (21) for $k=k_l$. The conditional probability of delay $P[D^{sh}(j,N_t)|\text{polled and talk}]$ for all possible combinations of (j,N_t) is given by

$$P\left[D^{sh}(j,N_t)|\text{polled and talk}\right] =$$

$$= \begin{cases} \dfrac{p_s p \binom{2j-1}{N_t-1} p^{N_t-1}(1-p)^{2j-N_t}}{(1-NP_s^{sh})p} & \text{if } 1 \le N_t \le \min(2j, N_{t\max}^{sh}-2) \\[2ex] \dfrac{p_s p(1-p) \binom{2j-2}{N_t-1} p^{N_t-1}(1-p)^{2j-N_t-1}}{(1-NP_s^{sh})p} & \text{if } j = N_{\max}^{sh} \wedge N_t = N_{t\max}^{sh}-1 \\[2ex] \dfrac{p_s p^2 \binom{2j-2}{N_t-2} p^{N_t-2}(1-p)^{2j-N_t}}{(1-NP_s^{sh})p} & \text{if } j = N_{\max}^{sh} \wedge N_t = N_{t\max}^{sh} \\[2ex] \left(p_s p \binom{2j-1}{N_t-1} p^{N_t-1}(1-p)^{2j-N_t} \sum_{n=0}^{\left\lfloor \frac{2N_{\max}^{sh}-2-N_{t\max}^{sh}}{2} \right\rfloor} \left(\binom{2N_{\max}^{sh}-2n-2-2j}{N_{t\max}^{sh}+2n-1-N_t} p^{N_{t\max}^{sh}+2n-1-N_t} \cdot (1-p)^{(2N_{\max}^{sh}-2n-2-2j)-(N_{t\max}^{sh}+2n-1-N_t)} \right)\right) / (1-NP_s^{sh})p & \text{if } N_{t\max}^{sh}-1 \le N_t \le \min(2j, N_{t\max}^{sh}+2n-1) \wedge 2N_{\max}^{sh}-2n-2-2j \ge N_{t\max}^{sh}+2n-1-N_t \end{cases} \quad (27)$$

Using (27), the delay complementary probability distribution $P[D^{sh}(j,N_t)>t]$ of each station is given by

$$P\left[D^{sh}(j,N_t)>t\right] = \sum_{\forall (j,N_t):D^{sh}(j,N_t)>t} P\left[D^{sh}(j,N_t)|\text{polled and talk}\right] \quad (28)$$

5 Numerical results

In this Section, some numerical results are presented that demonstrate the performance of IEEE802.11 networks and the improved performance of the cyclic shift polling scheme. For a better presentation of the results, three different cases are used: case 1 is related to the transmission of constant bit rate voice, case 2 uses silence detection for increasing the number of supported voice stations and case 3 refers to the proposed polling method used along with silence detection. The transmission time of a voice payload is determined by the voice packet generation interval and the type of coder used at the mobile stations. To demonstrate the results adaptive differential pulse code modulation (ADPCM) at 32 kbit/s is used, since this type of coding supports voice with high quality, introduces low delay and adds low computational complexity. In the IEEE802.11 networks, the PHY header is transmitted at 1 Mbit/s, while the MAC header and the payload are transmitted at 5.5 or 11 Mbit/s. This significantly increases the introduced overhead and the network throughput is decreased. In the IEEE802.11b specification [12], an optional short PHY (SPH) header has been defined with much shorter transmission time and is used in most of the numerical results. The mandatory long PHY (LPH) header is used at the last figure in order to study how the type of header affects the network throughput. The results of the analysis are derived using the parameter values shown in Table 1.

Table 1: IEEE802.11 wireless LAN attribute values

Attribute	Channel bit rate 5.5 Mbit/s	Channel bit rate 11 Mbit/s
Long physical header (all octets at 1 Mbit/s)	24 octets	24 octets
Short physical header (9 octets at 1 Mbit/s and 6 octets at 2 Mbit/s)	15 octets	15 octets
MAC header	34 octets	34 octets
Ack, CTS	14 octets	14 octets
RTS, CF–END, CF–Ack+CF–END	20 octets	20 octets
CF–Poll, CF–Ack+CF–Poll, CF–Ack, Null	34 octets	34 octets
Beacon	106 octets	106 octets
SIFS	10 μs	10 μs
DIFS	20 μs	20 μs
PIFS	50 μs	50 μs
SlotTime	20 μs	20 μs
Voice coding rate	32 kbit/s	32 kbit/s
Probability p a station is in talk state	0.4	0.4

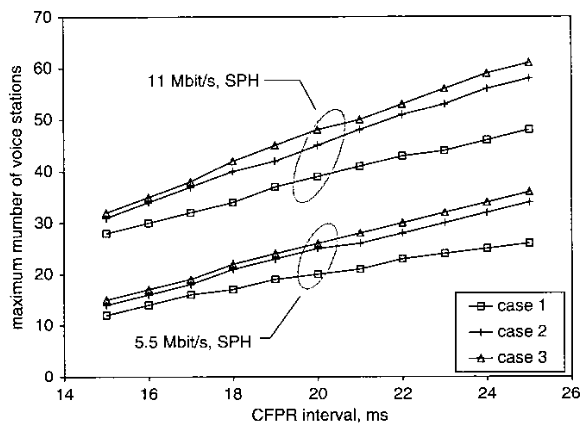

Fig. 4 *Number of supported voice stations against CFPR interval*

Fig. 4 shows the maximum number of voice stations supported by PCF in all cases, while the packet loss rate remains less than 0.01. The number of supported voice stations increases when silence detection is used, and further improvement is achieved by implementing cyclic shift polling. The number of supported conversations increases as the CFPR interval and the channel bit rate increase, since the bandwidth available for voice transmissions increases, but at the expense of larger delays.

The efficiency of the cyclic shift polling scheme is also verified in Fig. 5, which shows the probability of voice packet rejection as a function of the number of stations in the polling list (higher than N_{max}) when silence detection is used. This probability determines the number of voice stations that can be supported by each polling method for providing acceptable voice service. Cyclic shift polling increases the number of supported stations, since, for the same number of stations, the probability of rejecting voice packets on each individual station is reduced.

Fig. 6 depicts the delay the voice packets experience for cases 2 and 3. The improved performance of cyclic shift polling is evident since it presents a better delay complementary distribution. The delay complementary distribution

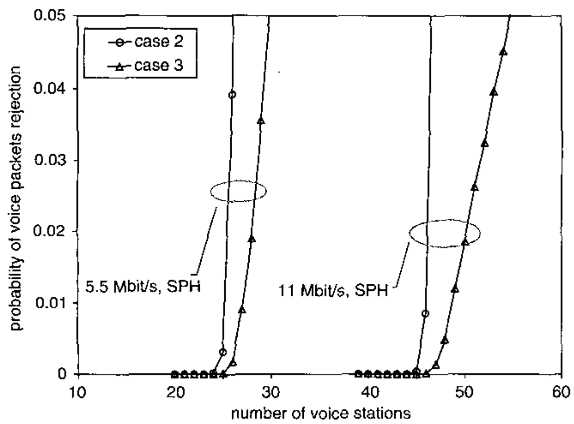

Fig. 5 *Probability of dropped voice packets against number of stations in the polling list*

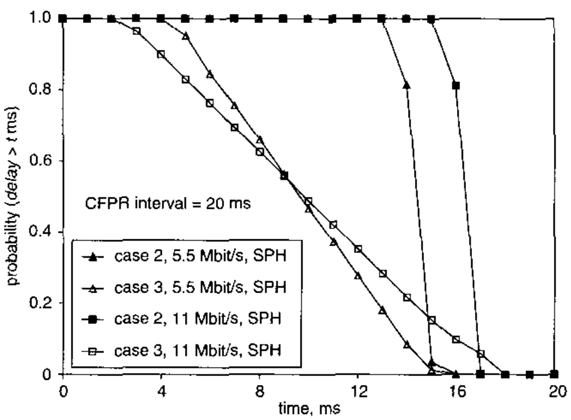

Fig. 6 *Effect of polling on the delay of voice packets*

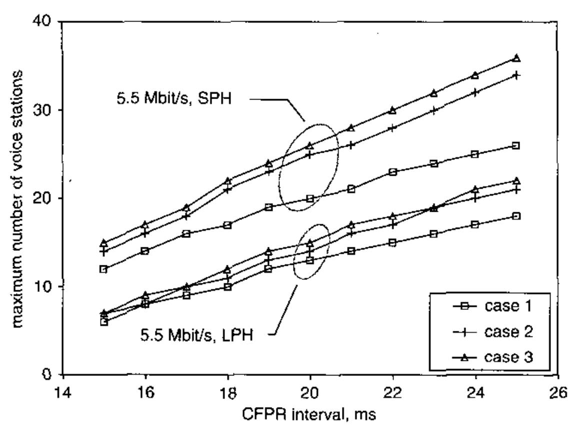

Fig. 7 *Effect of physical header on PCF performance*

of case 3 results in a lower mean delay compared to the mean delay experienced by the last station of case 2. Using cyclic shift polling the delay is uniformly distributed to all voice stations which is contrary to case 2, where the position of a station affects its delay. Finally, the delay is limited by the value of CFPRI and thus the CFPR interval determines the upper bound of the delay distribution.

More voice conversations can be accommodated by PCF when the SPH header is used due to its shorter size compared to the LPH header. As is shown in Fig. 7, the increase in the number of voice conversations achieved with the SPH header is significant (i.e. about 60% for 25 ms CFPRI). The use of cyclic shift polling with the SPH header increases even further the number of supported voice connections.

6 Conclusions

The cyclic shift polling scheme proposed in this paper can be used along with silence detection on IEEE802.11 wireless LANs for increasing the number of supported voice communications. The performance analysis of PCF shows that static polling and constant bit rate coding results in a small number of supported voice stations. The use of silence detection can increase the number of supported voice calls, but lower quality of service to the last stations in the polling list is provided, due to variable packet rejection rates. The introduction of the cyclic shift polling scheme spreads uniformly the total packet rejection rate, better quality of service is achieved and the number of supported conversations is further increased.

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