# Voice Communications over IEEE802.11 Wireless LANs interconnected using ATM links

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# Voice Communications over IEEE802.11 Wireless LANs Interconnected Using ATM links

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## Abstract

*This paper presents a new polling scheme, that can be used at the Access Point of an IEEE802.11 wireless network as part of the Point Coordination Function, in order to provide guaranteed QoS support of voice traffic. The so called Cyclic Shift polling scheme along with the use of silence detection at the mobile terminals increases the supported number of calls, as a function of the overall rate of dropped voice packets and the Point Coordination Function repetition interval. The advantages of utilizing this polling scheme are demonstrated by using the voice delay and voice jitter distributions. The delay and jitter distributions confirm that the voice packet delay introduced by the wireless network can be less than 25 ms even in the case of stations of different wireless networks when the IEEE802.11 LANs are interconnected via ATM infra-structure.*

## 1. Introduction

Wireless LAN technology is rapidly becoming a crucial component of computer networks connecting users with multimedia capabilities. The medium access control (MAC) and physical layer (PHY) protocol specifications are defined by IEEE802.11 standard [1]. In the MAC sublayer architecture, the Distributed Coordination Function (DCF) and the Point Coordination Function (PCF) coexist. DCF employs the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) protocol and a random backoff mechanism, while PCF uses a polling mechanism controlled by a Point Coordinator (PC) operating at the Access Point (AP) of each Basic Service Set (BSS). Such centralized control is more suitable to support time-bounded traffic than the CSMA/CA protocol.

The integration of voice and data communications in a wireless network by means of the IEEE802.11 protocol is a topic of intense research interest. Currently, IEEE802.11 Wireless LAN Working Group is working on the MAC protocol enhancement, in order to support various Quality of Service (QoS) requirements. Visser [2] simulated the combination of speech traffic and data traffic over an IEEE802.11 network using statistical multiplexing under the assumption that the voice activity occurs between stations in different BSSs. In [3] and [4], Crow's simulations suggest that an echo canceller is required for handling on/off speech traffic exchanged among different BSSs. These papers consider that the channel bit rate is 1 and 2 Mbps and conclude that polling is inefficient and doesn't meet the time requirements for real-time traffic. Romans [5] presents a hybrid protocol for wireless LANs, which combines a TDMA access mechanism to support voice and the CSMA/CA access mechanism to support data. This hybrid protocol is designed for use on a frequency hopping system and offers up to 4 reliable voice connections. At [6] a modified DCF access mechanism is proposed in order to provide real-time applications. Zahedi [7] integrates voice and data using the Transmission Control Protocol/Internet Protocol (TCP/IP) protocol operating in a CSMA environment and adopts the Improved Multiband Excitation (IMBE) low speed vocoder. In [8], Liu adapts the power-saved (PS) mode of the IEEE802.11 specifications for carrying the voice traffic produced by G.729 speech coders. Finally, in [9] we examined the characteristics of the service that constant bit rate (CBR) voice traffic experiences when it is supported by the PCF access procedure of an IEEE802.11 LAN, while the DCF access procedure supports data traffic. Results were derived for scenarios with and without echo cancellation and for channel bit rates 5.5 and 11 Mbps, the high data rate extension of the IEEE802.11 standard using Direct Sequence Spread Spectrum [10].

Papers [2]-[4] conclude that the PCF is inefficient for packet voice transmission, while papers [5]-[8] demand considerable modifications to the IEEE802.11 specifications that may potentially impact backward compatibility and future adoption of the IEEE802.11 wireless LAN. Therefore, it will be prudent to adopt QoS approaches that do not require major changes. In this paper, we extend and improve the work presented in [9] by proposing a more efficient management scheme of the Point Coordinator's polling list, assuming silence detection at the voice stations, and allowing voice packets exchange between different IEEE802.11 LANs. Specifically, PCF is used for voice traffic and DCF for data traffic and signaling messages. At every PCF round, the Access Point of each BSS implements a cyclic shift on its polling list. The use of silence detection at the mobile terminals in conjunction with the proposed management scheme at the PC's polling list results to the increase of the number of supported voice calls. We consider voice activities inside a BSS and between different BSSs interconnected via ATM networks. An upper bound to the number of voice conversations that a BSS can handle is estimated, while the overall rate of lost packets is limited below $10^{-2}$ and a predetermined minimum bandwidth for data traffic is guaranteed. Simulations of the proposed scheme were performed using 32 kbps adaptive differential pulse code modulation (ADPCM) voice coders at the mobile terminals along with silence detection. The main advantage of the proposed polling scheme is that it does not require any modifications on the mobile terminals.

Section 2 describes the integration of voice and data on IEEE802.11 networks and the interchange of voice packets via ATM infra-structure, while the APs of the BSSs employ the proposed polling scheme. In Section 3, we present the simulation model, while Section 4 discusses extensive simulation results.

## 2. Internetworking environment description

We consider a BSS network that employs PCF and DCF functions using a time-sharing mechanism. When DCF is used, the network is in a Contention Period (CP) and the stations must contend for gaining access. When PCF is used, the network is in a Contention Free Period (CFP), and the stations do not contend for transmitting their frames. Each CFP begins with a Beacon frame transmission and alternates with the CP. The PC generates each CFP at a predetermined time instance, which is defined by the Contention Free Repetition Interval (CFPRI) parameter and is called Target Beacon Transmission Time (TBTT). The length of CFP is based on the available traffic and the size of the polling list. The PC may terminate any CFP frame at or before a maximum duration, called *CFPmaxDuration*. The CFP is foreshortened, if at the nominal Beacon transmission time, the medium is busy due to DCF traffic. 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 IEEE802.11 standard defines a maximum value for the amount of time that the CFP is foreshortened and a minimum duration for CP. All timing parameters that determine the coexistence of PCF and DCF are contained in the Beacon frame.

Each voice station desiring to make a voice call issues a request to the PC and if this request is accepted by the PC and the called station, the two stations start exchanging their voice packets during CFP and under the PC control. The polling list of each PC consists of entries which either describe two connected stations belonging to the local BSS or a local station that has establish a connection with a station belonging to another BSS. So, the PC may control two types of voice traffic:

- *Traffic inside a BSS:* In this case the PC participates only on the polling sequence and voice packets are transmitted directly from one station to another, without the PC intervention, and
- *Traffic between different BSSs:* In this case the PC of each BSS participates on the polling sequence and is used as a store-and-forward device for voice packets.

Figure 1 illustrates the operation of PCF with an example of a polling list containing two stations ($S_A$ and $S_C$) inside a BSS and two stations ($S_B$ and $S_D$) communicating with stations on other BSSs. When the CFP starts, the PC sends a CF-Poll to the first station ($S_A$) of the polling list. The station responds by sending its voice packet to the other station in the BSS, no later than SIFS time (Short Interframe Space) after receiving the CF-Poll from the PC. When the destination station receives the voice packet, an ACK packet is returned to the source station and the PC waits a PCF interframe space (PIFS) following the ACK packet, before polling the next station ($S_C$) of the polling list. If the polled station has no frame to transmit, the response is a NULL packet. The next station on the polling list is $S_B$, and the PC sends a Data+CF-Poll, since it has a packet for this station. The $S_B$ station sends its voice packet and confirms the reception of the voice packet with a CF-ACK+Data. The PC has a voice packet for the $S_D$ station and must acknowledge the received packet from the $S_B$ station, therefore the PC sends a Data+CF-ACK+CF-Poll. Since the $S_D$ station received a voice packet, but has no packet

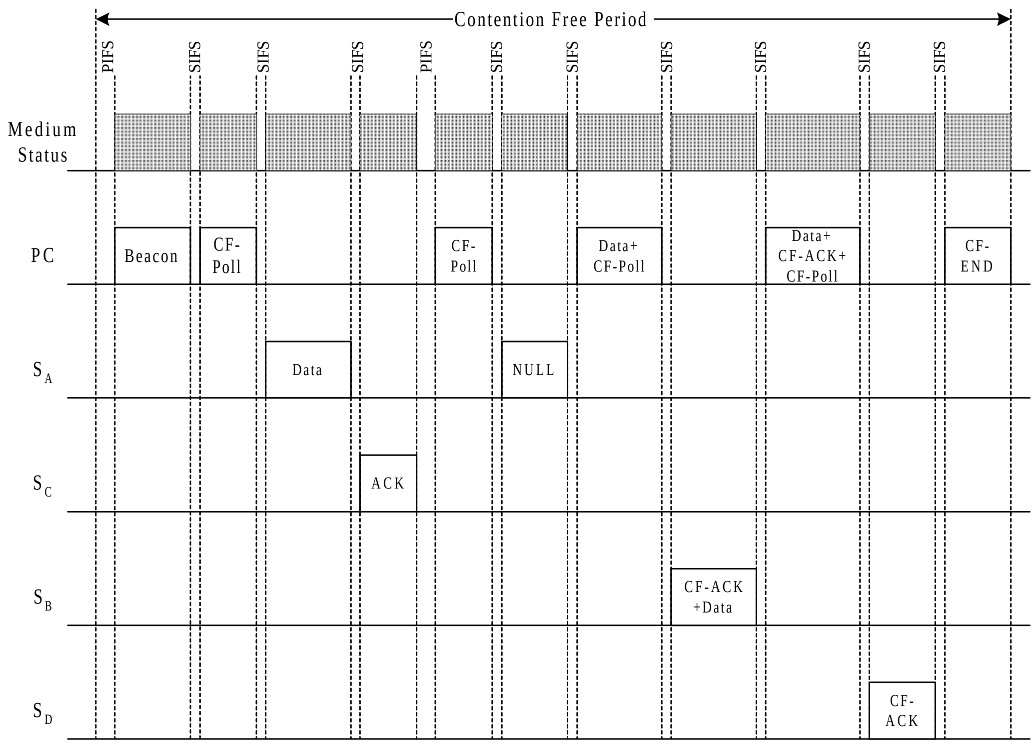


**Figure 1. Voice transmission over an IEEE802.11 BSS network**

to send, it responds with a CF-ACK. The PC terminates the CFP with a CF-END packet.

The PC has two buffers, one for the packets destined to another BSS and the other for packets received from other BSSs. Since silence detection is employed, when a station is in talk spurt a voice packet is generated at every CFPR interval and is transmitted over the network each time the station is being polled by the PC. In each voice station, if a new packet is generated before an old packet has been transmitted, the old packet is discarded. Using silence detection, the PCF can handle more voice transmissions by allowing a bounded rate of dropped packets. For keeping the packet dropping rate low, the time between two successive polling instants of the same station must be close to the voice packet generation interval. So, we consider that the PC restarts the polling sequence with the first station on its polling list during each PCF round and polls each station once during CFP. In this case, the voice packets of a station, depending on its position on the polling list, suffer a variable delay until the polling instances of that station arrive, since the DCF traffic modifies the CFP length. If the total duration of the transmitted voice and NULL packets exceed the *CFPmaxDuration*, then the last stations are not polled and their voice packets, if these stations are in talk spurt, are discarded by new packets. In this case the fairness requirement is violated, since the last stations on the polling list experience larger delays and worst packet dropping rates.

## 2.1. IEEE802.11 LANs interconnection using ATM links

In our system, an ATM infra-structure supporting AAL2 (adaptation layer type 2) is used for interconnecting different IEEE802.11 BSSs. The AP implements the functions of an interworking unit (IWU) and provides a mapping between the IEEE802.11 and ATM formats and addresses. An AAL2 packet consists of a 3 octets header and a payload of 1-45 octets that can be filled with the voice information. The ATM cell includes a header (5 octets), a start field (1 octet) and a payload (47 octets). AAL2 packets can be multiplexed and packed into ATM cells [11]. The multiplexing and packing process depends on the voice packet size defined by the CFPR interval of the BSS network. For simplicity, we consider that an ATM cell is filled with an AAL2 packet, thus the voice packet size is 44 octets, which results to 11 ms CFPR interval with a 32 Kbps ADPCM vocoder.

### 2.2. The polling scheme versus the performance objectives

The use of silence detection results to the increase of the number of voice stations supported by the network, compared to the case of not using silence detection. Whenever the PC cannot complete its polling list during a CFP period, the PC restarts its polling sequence with the first station on its polling list. According to [12], a bound of 1 % packet loss can be tolerated without adversely affecting the subjective quality of supported voice services and this bound determines the maximum number of voice stations that can be handled by PCF. As it was mentioned, voice packet delays and dropping rates increase for stations of higher positions on the polling list. For uniformly distributing the delays and packet dropping rates to all voice stations, we propose an improved scheme for the PC's polling list, the so called Cyclic Shift Polling process. According to this scheme, the PC cyclically shifts the stations on the polling list at the beginning of each PCF round. 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. To visualize the effect of Cyclic Shift polling process on the dropping rate of each station, in Table 1 we present simulations for a BSS of 14 voice stations in which we examined the dropping rate of each station in case of using silence detection without cyclic shift and in case of using silence detection with cyclic shift. The simulation parameters employed are summarized in Section 4.

The 1% bound of dropped packets must be limited to 0.5% inside each BSS, in the case of communications between different BSSs, since the voice packets of a transmitting station are dropped either

- if the station is in talk spurt and the AP of the BSS, in which this station belongs, does not poll this station and therefore a new packet discards the previously generated packet, or
- if the AP of the BSS that the receiving station belongs has a packet to deliver but the station is not polled during the current round.

So, the voice packets of a station are dropped independently in each BSS and for satisfying the 1% overall voice packet rate, if $p_{drop}$ denotes the probability of dropped voice packets, then the drop rate inside each BSS must satisfy the condition: $p_{drop} \leq 5 \cdot 10^{-3}$. In the case of WLANs interconnection using ATM infra-structure, ATM can be considered error-free, since its BER is better than $10^{-9}$ and the discard of a voice packet is due to the internal behavior of a BSS.

**Table 1. The effect of Cyclic Shift polling process on the dropping rate of each station**

| Station number | Channel bit rate = 5.5 Mbps, $BER_{bad}$ = 0, CFPR itreval = 11 ms, 14 voice stations | |
|---|---|---|
| | Silence detection without cyclic shift (%) | Cyclic shift and silence detection (%) |
| 0 | 0.00 | 0.08 |
| 1 | 0.00 | 0.09 |
| 2 | 0.00 | 0.06 |
| 3 | 0.00 | 0.08 |
| 4 | 0.00 | 0.06 |
| 5 | 0.00 | 0.09 |
| 6 | 0.00 | 0.05 |
| 7 | 0.00 | 0.08 |
| 8 | 0.05 | 0.05 |
| 9 | 0.18 | 0.09 |
| 10 | 0.41 | 0.05 |
| 11 | 0.77 | 0.09 |
| 12 | 1.13 | 0.07 |
| 13 | 1.83 | 0.09 |

## 3. The simulation model

Our simulation model includes a BSS model, in order to characterize a single BSS with an AP, a voice model that generates voice packets, a burst error model that represents the fading medium and an ATM model to interconnect different BSSs for speech communication. For reducing the complexity of the model the following assumptions have been made:

- The effect of propagation delay on the BSS model is considered negligible.
- The "hidden terminal" problem is not addressed.
- No stations operate in power-saving mode.
- No overlapping BSSs exist.

This model concentrates on the PCF effectiveness for transmitting packetized voice between stations in the same or different BSSs using the Cyclic Shift polling scheme.

### 3.1. The BSS model

A BSS network supports asynchronous data users during its contention period and voice users during its contention free-period. We limit the CP duration to its minimum value, so the PC allocates most of the CFPR duration for contention free services. Since the asynchronous data traffic may enlarge this minimum duration of the CP, the CFP duration is foreshortened while the CFPR interval remains constant. Let $T_{CP\min}$ denotes the minimum duration of CP, $T_{FS\max}$ the maximum duration the CFP is delayed, $T_{CP}$ the duration of CP, $T_{CFP}$ the duration of CFP and $T_{CFPR}$ the duration of CFPR, then:

$$T_{CP\min} \le T_{CP} \le T_{CP\min} + T_{FS\max}$$

$$T_{CFPR} = T_{CP} + T_{CFP}$$

$$T_{CFPR} - \left(T_{CP\min} + T_{FS\max}\right) \le T_{CFP} \le T_{CFPR} - T_{CP\min}$$

We model the asynchronous data traffic with a time generating function that modifies the CFP duration. This function follows the uniform distribution with parameters $T_{CPmin}$ and $T_{CPmin}$ + $T_{FS\max}$. The values of $T_{CPmin}$ and $T_{FS\max}$ are defined in [1].

The voice users transmit during CFP, according to the PCF rules described in Section 2. A receiving station may belong to the same BSS with the transmitting station or in another BSS. An internal single voice packet buffer and a single packet transmit buffer are maintained in each voice station. Each newly generated voice packet fills the internal buffer and when this buffer is full, the older packet is discarded. The AP implements the Cyclic Shift polling scheme. We consider that it has an infinite input buffer for voice packets coming from the ATM and an infinite output buffer for forwarding voice packets to the ATM. This consideration annihilates the probability of lost voice packets at the AP.

### 3.2. Voice model

A voice source alternates between talk spurts and silent periods. In case of using silence detection, a station that is in a talk spurt, generates voice packets periodically, but if it is in a silent period, no voice packets are generated. We adopt the discrete-time version of a well-known model proposed in [13], according to which, each voice source is modeled by a two-state discrete-time Markov chain. The duration of a voice talk spurt fits the exponential distribution with average equal to 400 ms and the duration of silent periods also follows the exponential distribution with average equal to 600 ms [14].

In [4] mean spurt and silent length is in the order of 1 to 2 sec. The mean spurt and silent length fall into two regions, depending on hangover, a technique used by silence detectors to avoid sudden end-clipping of speech and to bridge short speech gaps such as those due to stop consonants. If the hangover is small, mean spurt is around 200 to 400 ms, and the mean silent length is around 500 to 700 ms [14], [15]. In this paper we assume a small hangover since it is commonly used in newer silence detectors. If at a TBTT instant a station enters in a talk spurt, at the next PCF round the station generates a voice packet that is transmitted when the PC polls the station. If at a TBTT instant a station becomes silent, no voice packet is generated at the next PCF round and the station transmits a NULL or a CF-ACK packet when it is polled.

### 3.3. Burst error model

For simulating the BSS behavior, a burst error model is introduced representing fading conditions in the communication medium [3], [4]. This model utilizes a two-state discrete time Markov chain. The two states are called G (good) and B (bad) and indicate that the medium is operating with a very low bit error rate (denoted by $\mathrm{BER}_{\mathrm{good}}$) and in a fading condition with a higher error rate (denoted by $\mathrm{BER}_{\mathrm{bad}}$) respectively. State G changes to state B with a transition rate $\alpha$, while state B changes to state G with a transition rate $\beta$. A packet is considered to be lost if it contains one or more erroneous bits. When a packet of $\left(n_1 + n_2\right)$ bits is transmitted, a portion of the packet (let be $n_1$ bits) can be transmitted in the state G and a portion (let be $n_2$ bits) in the state B. Therefore, a packet is transmitted successfully with probability:

$$\Pr_{success} = \left(1 - \mathrm{BER}_{\mathrm{good}}\right)^{n_1} \left(1 - \mathrm{BER}_{\mathrm{bad}}\right)^{n_2}$$

**Table 2. Attribute values for the IEEE802.11 BSS network**

| Attribute | Value | Value |
|---|---|---|
| | **Channel bit rate 5.5 Mbps** | **Channel bit rate 11 Mbps** |
| MAC header | 34 octets | 34 octets |
| Short Physical header | 15 octets | 15 octets |
| ACK, CTS | 14 octets | 14 octets |
| RTS, 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 us | 10 us |
| DIFS | 20 us | 20 us |
| PIFS | 50 us | 50 us |
| *SlotTime* | 20 us | 20 us |
| $T_{CFPR}$ | 11 ms | 11 ms |
| $T_{CPmin}$ | 3.895 ms | 2.002 ms |
| $T_{FSmax}$ | 3.605 ms | 1.818 ms |
| Voice sampling duration | 11 ms | 11 ms |
| Voice sampling rate | 32 Kbps | 32 Kbps |
| Average duration of talk | 400 ms | 400 ms |
| Average duration of silence | 600 ms | 600 ms |
| $\mathrm{BER}_{\mathrm{good}}$ | $10^{-10}$ | $10^{-10}$ |
| $\alpha$ | 30 $s^{-1}$ | 30 $s^{-1}$ |
| $\beta$ | 10 $s^{-1}$ | 10 $s^{-1}$ |

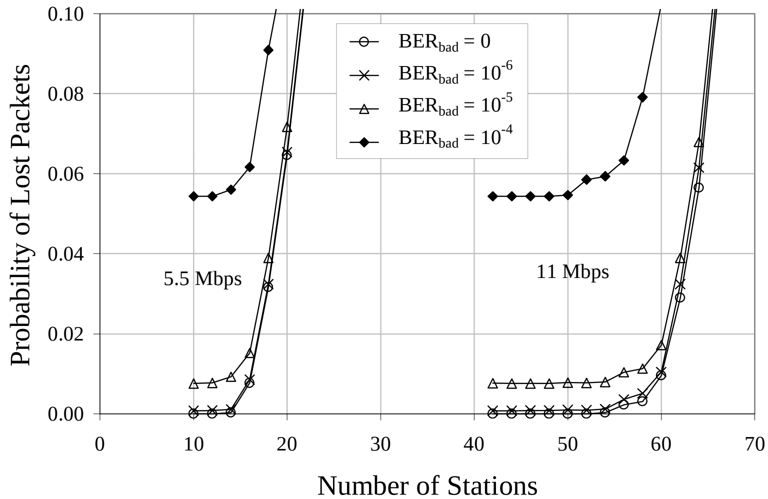


**Figure 2. The probability of dropped packets versus the number of voice stations**

### 3.4. ATM model

Since, we are interested in the investigation of the end-to-end network delay of the voice packets, the ATM network that is used for voice packet communications between different BSSs is modeled as a delay distribution, which inserts a random delay to each voice packet transmitted through the ATM network. The delay introduced by the ATM fits the exponential distribution with mean value equal to 1 ms. This delay mainly characterizes the queuing delays at a single ATM switch.

## 4. Simulation results

In this section, we present and discuss the results of our simulation demonstrating the performance of the Cyclic Shift polling scheme implemented on the AP of an IEEE802.11 BSS network for voice packet exchanges between users in the same and different BSSs, while a minimum duration is devoted to data traffic. Our system simulation was developed with VisSim LE included in Mathcad 2001. Initially, we examined the effect of the polling scheme on the probability of lost packets for each user. The probability of dropped packets $p_{drop}$, except the dropped packets due to the Cyclic Shift polling scheme and the use of silence detection, includes and the packets corrupted from the fading communication medium. According to the $p_{drop}<10^{-2}$ design objective, which results to a voice packet loss rate of less than $5\cdot 10^{-3}$ inside each BSS, we determine the maximum number of voice users, which can be supported by PCF when the Cyclic Shift polling process is employed, as the maximum packet loss probability of all different users. For finding the probability of dropped packets and the maximum number of voice users, our simulations are based on individual BSSs, since the voice packets are discarded independently in each BSS, as mentioned previously. Finally for the maximum number of voice users, the delay and jitter delay distributions have been found. The results of our analysis were derived considering the attribute values listed in Table 2. At this point, we must refer that all packets transmitted during the PCF use the optional short PHY header defined in [10] for reaching maximum throughput.

The maximum number of voice stations is defined using the probability of dropped packets. When silence detection is used, the number of supported voice users is increased, but some packets are discarded. Further, voice packets are corrupted if they contain one or more erroneous bits. With the proposed polling scheme the probability of dropped packets is equally distributed to all stations due to the cyclic shift. Figure 2 illustrates how the number of polled voice stations affects the probability of dropped packets for various values of $BER_{bad}$. The probability of dropped packets increases rapidly beyond a specific number of stations, since the *CFPmaxDuration* time parameter is exceeded many times. This Figure refers to a BSS network with only internal voice activities.

In Table 3, we present the maximum number of voice stations supported by a BSS for four different scenarios in accordance to our simulations. Scenarios 1 and 2 refer to voice connections inside a BSS for channel bit rates 5.5 and 11 Mbps respectively, while scenarios 3 and 4

**Table 3. Maximum number of voice stations achieved with the Cyclic Shift polling scheme**

| Scenario | | Maximum number of voice stations | |
|---|---|---|---|
| | | $BER_{bad}$ | |
| | | 0 | $10^{-6}$ |
| **1** | Internal voice activities<br>Channel bit rate = 5.5 Mbps | 14 | 14 |
| **2** | Internal voice activities<br>Channel bit rate = 11 Mbps | 58 | 56 |
| **3** | 5 external voice activities<br>Channel bit rate = 5.5 Mbps | 15 | 15 |
| **4** | 16 external voice activities<br>Channel bit rate = 11 Mbps | 62 | 60 |

include also external voice activities. In scenario 3, BSS operates at 5.5 Mbps and 5 stations forward their packets to others BSSs, while in scenario 4, we have a BSS at 11 Mbps with 16 external voice activities. The maximum number of voice stations is derived for voice packet dropping rate less than $5 \cdot 10^{-3}$.

The complementary probability distributions of voice delay Pr($D$ > $t$ ms) for scenarios 1 and 2 are shown in Figure 3. In this scheme, the voice packet delay remains lower than the chosen CFPR interval for any value of CFPRI and guarantees good speech quality for calls inside the same BSS. For external calls, the delay that a voice packet experiences in both BSSs remains lower than the sum of the CFPR intervals used by these BSSs. It is also obvious from Figure 4 that the voice jitter remains at very low values (most values are in the area of $\pm 2.5$ msec). As voice jitter we define the difference between consecutive values of the voice packet delay and Pr($D_{i+1}$-$D_i$ > $t$ ms) is the probability the jitter is greater than $t$ ms [15].

Furthermore, we studied the voice delay distributions for two stations from different BSSs interconnected via ATM networks. BSS-1 follows scenario 4 and BSS-2 follows scenario 3. The complementary probability distributions of voice delay for different portions of the interconnected BSSs are shown in Figure 5. The delay distribution between stations of different BSSs mainly depends on the delay introduced by each BSS. We notice that the selection of suitable CFPR interval along with the proposed polling scheme yield voice communications, which satisfy the QoS requirements, since the delay between different BSSs can be bounded to 25 ms. This is also true for the voice jitter distribution between two connected stations belonging to BSS-1 and BSS-2 respectively, as shown in Figure 6.

Figure 7 depicts the increase on the supported number of voice stations accomplished by implementing the Cyclic Shift polling process along with silence detection compared to the supported number of voice stations in case of CBR voice traffic and in case of implementing silence detection without the Cyclic Shift polling process for various values of the CFPR interval. The use of silence detection results to support of more voice stations, although the Cyclic Shift polling process further improves the network performance and spreads the voice packet dropping rate to all the voice stations uniformly. Additionally, when the CFPR interval increases, the PCF accommodates more calls at the expense of larger delays.

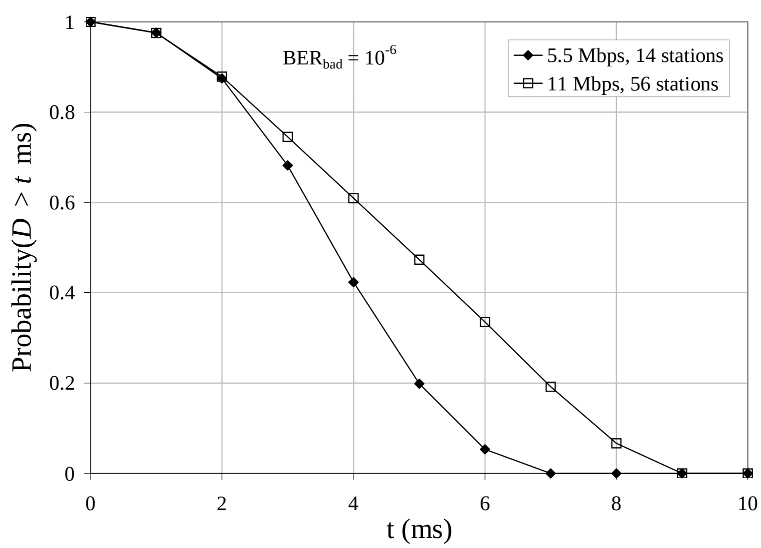


**Figure 3. Scenarios 1 and 2: The complementary probability distribution of voice delay**

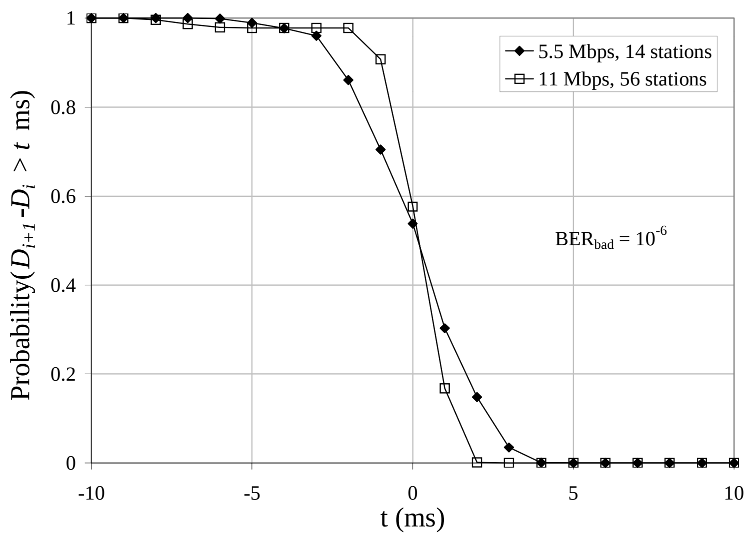


**Figure 4. Scenarios 1 and 2: The voice jitter probability distribution**

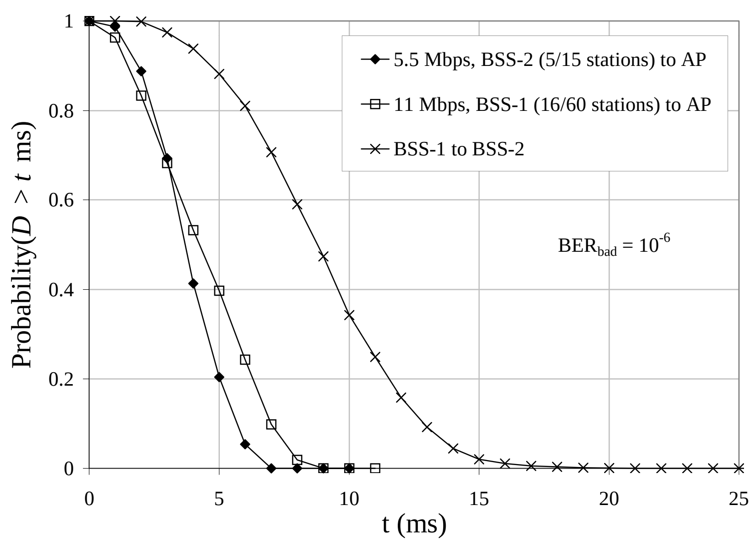


**Figure 5. Complementary probability distributions of voice delay for external voice activities**

Finally, in Table 4 we present the percentage of the CFPR interval (denoted by $CU_v\%$ ) used for voice transmissions for the cases illustrated in Figure 7, which is calculated by the following equation:

$$CU_v\% = \frac{N\frac{T_{CFPR}R_s}{R_c}}{T_{CFPR}}100 = \frac{NR_s}{R_c}100$$

where $N$ is the number of supported voice stations, $R_s$ the voice sampling rate and $R_c$ the channel bit rate. The network throughput is proportional to the number of voice stations handled by PCF and serious improvement is achieved using the Cyclic Shift polling process for higher values of CFPR intervals, because the portion of each packet devoted to header and polling overheads becomes smaller. Since the upper bound to the voice packet delays inside a BSS employing the proposed scheme is determined by the CFPR interval duration and delays are acceptable up to 150 ms when echo cancellers are used, higher throughput can be achieved by selecting larger CFPR intervals.

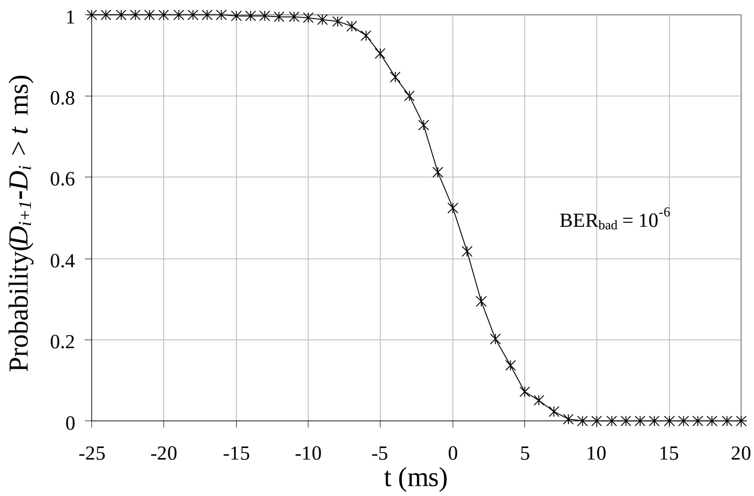


**Figure 6. The voice jitter probability distribution between two stations in different BSSs**

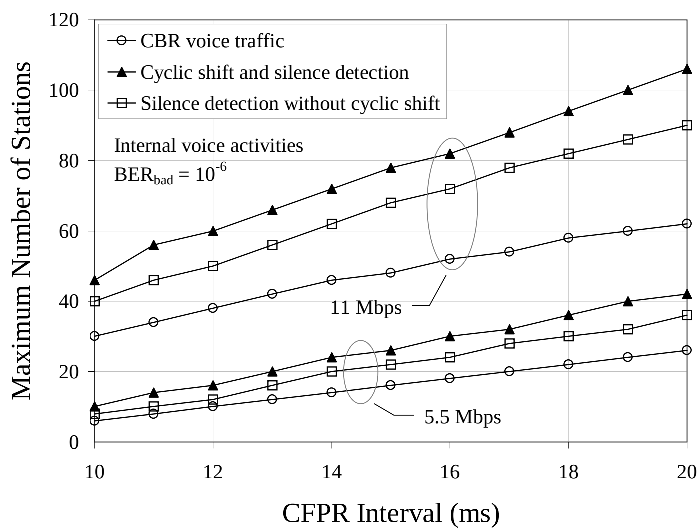


**Figure 7. The number of supported voice stations versus the CFPR interval**

## 5. Conclusions

In this paper, we presented and simulated the Cyclic Shift polling scheme applicable at the Access Point of each BSS for accomplishing QoS voice communications over IEEE802.11 LANs. The proposed polling scheme in conjunction with the use of silence detection supports a good number of calls inside and outside a BSS, which is defined by the chosen CFPR interval and the bound to the overall rejection rate of voice packets. The simple practical implementation of the proposed polling process and its fair features by equally distributing the delay and packet dropping rate to all the stations justify its use. The voice delay and jitter distributions derived by our simulations verify that the voice quality requirements are maintained for calls inside a BSS and for calls outside a BSS, when an ATM network interconnects the wireless LANs. Finally, higher throughput can be achieved and more voice users can be supported when larger values of CFPRI are used, although in this case the use of echo cancellers is unavoidable.

**Table 4. The % utilization of CFPR interval for voice transmissions**

| $T_{CFPR}$ (ms) | 5.5 Mbps | | | 11 Mbps | | |
|---|---|---|---|---|---|---|
| | CBR voice traffic | Silence detection without cyclic shift | Cyclic shift and silence detection | CBR voice traffic | Silence detection without cyclic shift | Cyclic shift and silence detection |
| 10 | 3.49 | 4.65 | 5.82 | 8.73 | 11.64 | 13.38 |
| 11 | 4.65 | 5.82 | 8.15 | 9.89 | 13.38 | 16.29 |
| 12 | 5.82 | 6.98 | 9.31 | 11.05 | 14.55 | 17.45 |
| 13 | 6.98 | 9.31 | 11.64 | 12.22 | 16.29 | 19.20 |
| 14 | 8.15 | 11.64 | 13.96 | 13.38 | 18.04 | 20.95 |
| 15 | 9.31 | 12.80 | 15.13 | 13.96 | 19.78 | 22.69 |
| 16 | 10.47 | 13.96 | 17.45 | 15.13 | 20.95 | 23.85 |
| 17 | 11.64 | 16.29 | 18.62 | 15.71 | 22.69 | 25.60 |
| 18 | 12.80 | 17.45 | 20.95 | 16.87 | 23.85 | 27.35 |
| 19 | 13.96 | 18.62 | 23.27 | 17.45 | 25.02 | 29.09 |
| 20 | 15.13 | 20.95 | 24.44 | 18.04 | 26.18 | 30.84 |

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