IEEE 802.11E ENHANCEMENT FOR VOICE SERVICE

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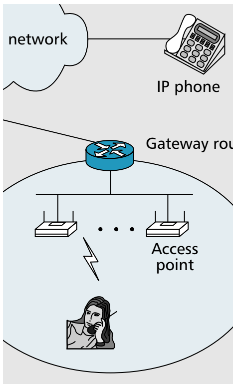

Motivated by the promising voice over IP (VoIP) technology and the wide service availability of wireless local area networks (WLANs), the application of voice over WLAN (VoWLAN) is expected to experience a dramatic growth in the near future.

ABSTRACT

Motivated by the promising voice over IP technology and the wide service availability of WLANs, the application of voice over WLAN (VoWLAN) is expected to experience dramatic growth in the near future. Originally designed for high-rate data traffic, WLANs may experience bandwidth inefficiency when supporting delay-sensitive and low-rate voice traffic. This article proposes mechanisms to enhance QoS support capability of IEEE 802.11e for voice service. Unnecessary polling of silent voice stations is avoided, and header and control overheads are suppressed significantly. Compared to IEEE 802.11e, our mechanisms can greatly improve the capacity of WLANs to support voice service.

INTRODUCTION

Although originally designed for data services, the Internet can also support real-time traffic such as voice and video. The technology of voice over Internet Protocol (VoIP), also known as Internet telephony, IP telephony, or packet voice, enables real-time voice conversations over the Internet. It has attracted much interest from academia and industry because of the following facts [1]:

- VoIP has much lower cost than traditional telephone service.
- The universal presence of IP makes it convenient to launch VoIP applications.
- There is increasing demand for networks to interact with end users having real-time data, voice, and video images, leading to the requirement for integrated voice, data, and video services.
- The emerging digital signal processing (DSP) and voice coding/decoding techniques make VoIP more and more mature and feasible.

Therefore, VoIP is anticipated to offer a viable alternative to traditional public switched telephone network (PSTN).

To provide person-to-person (instead of place-to-place) connections anywhere and anytime, the Internet is expected to penetrate the wireless domain. One very promising wireless network is the wireless local area network (WLAN), which has shown the potential to provide high-rate data services at low cost over local area coverage. Working in the license-exempt 2.4 GHz industrial, scientific, and medical (ISM) frequency band, the IEEE 802.11b WLAN offers a data rate up to 11 Mb/s, while IEEE 802.11a WLAN and European Telecommunications Standard Institute (ETSI) HIPERLAN/2 can support data rates up to 54 Mb/s at the 5 GHz frequency band. As a wireless extension to the wired Ethernet, WLANs typically cover a small geographic area, in hotspot local areas where the traffic intensity is usually much higher than in other areas.

The promising VoIP technology and wide deployment of WLANs are expected to drive the application of voice over WLAN (VoWLAN), which will experience a dramatic increase in the near future [2]. Figure 1 shows a typical VoWLAN system where voice conversation happens through the access point (AP). At the sender, the analog voice signal is compressed and encoded by a codec. After inclusion of the Real-Time Transport Protocol (RTP)/User Datagram Protocol (UDP)/IP headers during the packetization procedure at the transport and network layers, voice packets are transmitted over the networks and finally to the receiver end. At the receiver, a playout buffer is usually used to alleviate the effect of delay jitter. Then the receiver applies depacketization and decoding to recover the original voice signal.

One major challenge for VoWLAN is quality of service (QoS) provisioning. Originally designed for high-rate data traffic, WLANs may experience bandwidth inefficiency when supporting delay-sensitive and low-rate voice traffic. Hence, it is essential to enhance the QoS support capability of current WLAN standards, such as the most popular IEEE 802.11 standard.

LIMITATIONS OF IEEE 802.11 IN SUPPORTING VOICE

As a real-time application, VoWLAN is delay-sensitive but can tolerate a certain level of packet loss. Hence, delay and delay jitter are the main QoS measures. Each voice packet should be transmitted within a delay bound. Also, the delay jitter (i.e., variation of voice packet delay) should be carefully controlled as it may degrade voice quality more severely than delay. Traditionally, an appropriately designed playout buffer is an effective way to deal with delay jitter and make the voice understandable. Therefore, a delay bound guarantee is the main QoS requirement for voice under consideration in this article.

As the most popular WLAN standard, IEEE 802.11 defines a mandatory distributed coordination function (DCF) and an optional centralized point coordination function (PCF). DCF is based on carrier sense multiple access with collision

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avoidance (CSMA/CA), where collision is resolved by binary exponential backoff. The optional request-to-send (RTS)/clear-to-send (CTS) dialog can also be applied to further deal with the hidden terminal problem. Mainly designed for data transmission, DCF does not take into account the delay-sensitive nature of real-time services. On the other hand, with PCF, a contention-free period (CFP) and a contention period (CP) alternate periodically. During CFP, when polled, a station gets permission to transmit its DATA frames.[1] The main drawbacks of PCF include bandwidth waste when two stations in the same basic service set (BSS) (which is composed of an AP and a number of stations associated with the AP) try to communicate with each other, uncontrolled transmission time of a polled station, and unpredictable CFP start time [3].

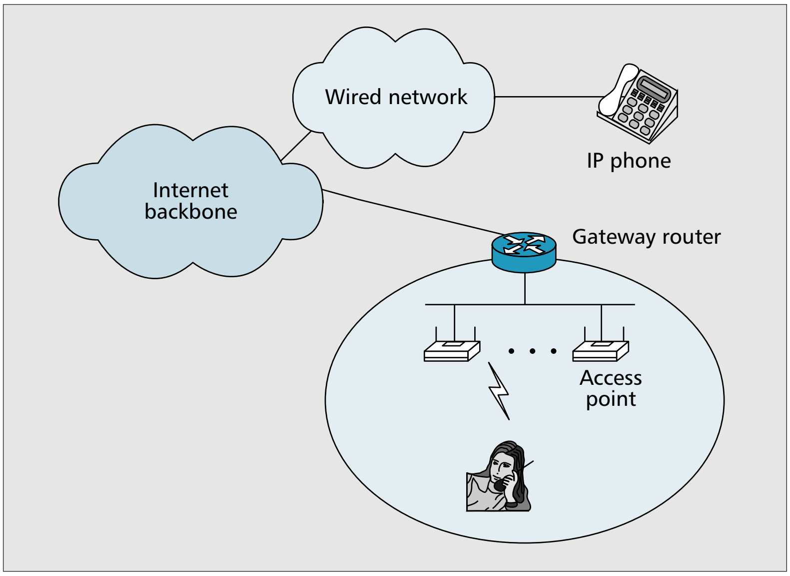
To enhance the legacy IEEE 802.11 medium access control (MAC), the IEEE 802.11e draft [4] proposes new features with QoS provisioning to real-time applications [5]. As an extension of DCF, the enhanced distributed channel access (EDCA) provides a priority scheme to distinguish different traffic categories by classifying the arbitration interframe space (AIFS), and the initial and maximum contention window sizes in the backoff procedures. In the IEEE 802.11e draft, the hybrid coordination function (HCF) can assign specific transmission durations by a polling mechanism. A station can be polled in either CFP or CP. In addition, the direct link protocol allows a station to transmit frames directly to another station.

Neither DCF nor EDCA is effective or efficient to support delay-sensitive voice traffic. Their contention-based nature and binary exponential backoff mechanism cannot guarantee that a voice packet is successfully delivered within the delay bound. In addition, the time to transmit the payload of a voice packet is only a very small portion of the total time to transmit the packet, due to overhead such as the RTP/UDP/IP headers, MAC header and physical (PHY) preamble, and IFSs. Subsequently, the capacity to accommodate voice traffic in DCF or EDCA is very limited. For example, IEEE 802.11b can support approximately 10 simultaneous two-way voice calls if a G.711 codec is used [6].

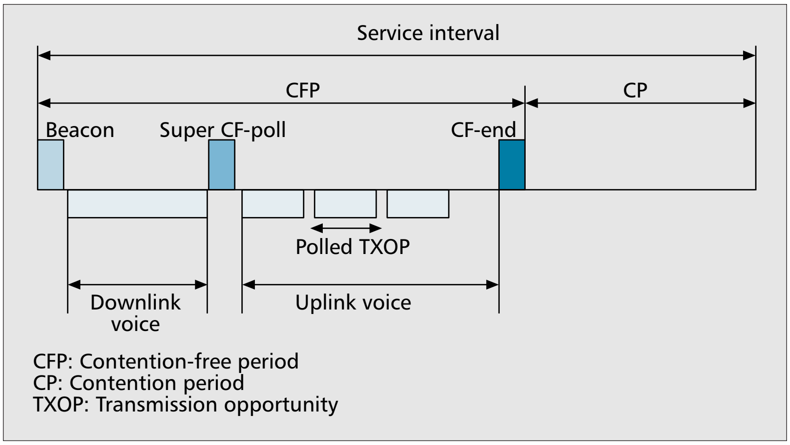
In order to guarantee the delay requirement of voice service, controlled access is preferred in WLAN, in which the AP polls each voice station periodically. To efficiently utilize the radio resources, two challenging issues need to be tackled:

- Voice multiplexing — Generally, voice traffic can be represented by an `on/off` model: active voice users (in the `on` state) transmit at a constant rate, and inactive users (in the `off` state) do not transmit, and the durations of the states are independent and exponentially distributed. It is desired to achieve statistical multiplexing based on this property in VoWLAN.
- Overhead reduction — The overhead due to RTP/UDP/IP headers and the polling procedure may significantly degrade system efficiency, and should be suppressed as much as possible.

To address these two issues, this article contributes toward enhancing the QoS support capability of IEEE 802.11e for voice services.



■ **Figure 1.** *The architecture for VoIP over WLAN.*



■ **Figure 2.** *The proposed structure of an HCF service interval.*

The Service Interval Structure

We propose a modified structure of the IEEE 802.11e HCF service interval, as shown in Fig. 2 where the beacon interval is equal to the service interval. In each service interval there are two periods: CFP and CP. The CFP is used to accommodate voice stations, and can be further divided into downlink and uplink portions. The downlink portion is used for the AP to send voice packets to mobile stations. After the transmission of all the downlink voice packets, the AP sends a super CF-Poll frame (to be further discussed), which grants a transmission opportunity (TXOP) to each station in its polling list. No acknowledgment (ACK)/retransmission is required for voice transmission in order to avoid the retransmission delay. In the CP, the AP and all the stations can contend for the channel. It is mainly used to serve data stations and transmit the first few packets of each voice station's talk spurt. The length of the service interval is fixed and depends on the delay bound of voice traffic. The lengths of the CFP and CP depend on the voice traffic load.

The QoS enhancement in the proposed service interval structure consists of four components:

[1] *The DATA frame means the information frame of voice or data traffic.*

In order to achieve a high resource utilization, the network designers should consider the on/off characteristic of voice traffic, so that resources are allocated to stations only when they are in a talk spurt. However, the IEEE 802.11e draft does not describe a polling method in HCF to achieve voice traffic multiplexing.

voice traffic multiplexing, deterministic access priority of voice, overhead reduction, and call admission control, as elaborated in the following.

VOICE TRAFFIC MULTIPLEXING

In order to achieve high resource utilization, network designers should consider the `on/off` characteristic of voice traffic so that resources are allocated to stations only when they are in a talk spurt. However, the IEEE 802.11e draft does not describe a polling method in HCF to achieve voice traffic multiplexing. Generally, it is easy for the AP to recognize the ending moment of a talk spurt, but it is difficult to know the exact starting moment of a talk spurt. The AP may still need to poll a voice station even during its silent periods in order not to miss the beginning of a talk spurt, which is not efficient. Here we propose a more efficient polling scheme to achieve voice traffic multiplexing.

Consider the case in which a station initiates a voice call to either another station in the same BSS or another user outside the WLAN. If the call can be admitted, the AP will add the station to the end of its polling list. Since the duration of each service interval is fixed and the arrival rate of voice packets is a constant in a talk spurt, each station (in the on state) will be granted a fixed TXOP just big enough to accommodate the generated voice packets during a service interval. If a polled station has no packet to send or cannot use up all the time of a TXOP, the AP considers the station in a silent period and deletes it from its polling list, except newly added (to the polling list) stations. When a station previously in an `off` state has voice packets to send and finds itself being excluded in the super CF-Poll frame, the station contends for the channel during the next CP. Once it gets the channel, it sends out all the voice packets in the buffer (as long as the transmission time does not exceed the TXOP). The AP monitors all the packets sent in the CP. For every voice packet, the AP records the sender address (or ID) and adds it to the end of its polling list. If the station is newly added to the list during the last service interval, the AP will retain it in the list, even though it may not use up all the TXOP or has no packet to send in the current service interval, since a few voice packets at the beginning of a talk spurt were sent during the previous CP.

Once a voice station is being polled, all subsequent voice packets in the same talk spurt will be transmitted in the CFP. Hence, the voice station does not need to contend for the channel anymore.

DETERMINISTIC ACCESS PRIORITY OF VOICE IN CP

Another challenging issue is raised by voice multiplexing: To meet the strict delay requirement of voice traffic, it should be guaranteed that a voice station can access the channel successfully during the CP when needed.

In the CP of IEEE 802.11e, EDCA is used. It applies different initial and maximum contention window sizes, and different IFS values to provide differentiation to different types of traffic. However, it provides only statistically rather than deterministically prioritized access to high-priority traffic such as real-time voice. In other words, prioritized access for high-priority traffic is only guaranteed in the long term, but not for every contention. Since each station continues to count down its backoff timer once the channel becomes idle for an IFS, a low-priority packet with a large initial backoff timer will eventually count down its backoff timer to a small value, most likely smaller than the backoff timer of a new backlogged high-priority packet. Then the low-priority packet grabs the channel, resulting in the high-priority packet waiting a long time for the next competition [7]. With such statistically prioritized access it is hard to satisfy the delay requirement of each voice packet. Furthermore, when applying EDCA, with the increase of low-priority traffic loads, the collision probability seen by high-priority traffic increases. High-priority traffic can suffer performance degradation due to low-priority traffic offering heavy loads [8]. To provide QoS guarantee for voice traffic regardless of the data traffic load in WLAN, data stations should not transmit in the CP until no voice station contends for the channel. As a result, deterministically prioritized access is more appropriate. Only a few voice packets at the beginning of each talk spurt need to contend in the CP, which does not significantly degrade the QoS of data traffic.

A simple way to provide deterministically prioritized access is to modify EDCA so that the AIFS of the data access category (AC) ($AIFS[AC\_data]$) is equal to the summation of AIFS of voice AC ($AIFS[AC\_voice]$) and the maximum contention window size of the voice AC ($CW_{\max}[AC\_voice]$). However, it is not efficient in terms of channel utilization. The number of voice packets is expected to be small in a CP, and all the data packets have to wait a long time before getting the channel, resulting in a waste of resources.

Inspired by the idea of black-burst contention [9], here we propose a more efficient scheme to provide deterministically prioritized access, by minor modifications to IEEE 802.11e EDCA. In our scheme the system parameters (e.g., $CW_{\min}$, $CW_{\max}$, and AIFS for voice traffic and data traffic) remain the same as those in IEEE 802.11e. In addition, the contention behaviors for data stations remain the same as in IEEE 802.11e. The contention behaviors of voice stations are modified as follows. For a contending voice station, after waiting for the channel to be idle for $AIFS[AC\_voice]$, instead of further waiting for the channel to be idle for a duration of backoff time, the voice station will send a black burst (i.e., pulses of energy) to jam the channel, and the length of the black burst (in the unit of slot time) is equal to its backoff timer. After the completion of its own black burst, the station monitors the channel. If the channel is still busy (which means at least one voice station is sending a black burst), the station will quit the current contention, choose a backoff timer randomly from its contention window, and wait for the channel to be idle for $AIFS[AC\_voice]$ again. Otherwise, the station that sends the longest black burst will send its voice packets. It is possible that two or more voice stations happen to send the same longest black burst, resulting in a collision. Collisions should be resolved as the AP cannot put the collided stations into the polling list. Since there is no ACK frame sent back to acknowledge successful transmission, it is difficult for the sender to recognize the collision. To address the problem in our scheme, for the first packet from a voice station received in the CP, the receiver should send back an ACK frame to the sender. If no ACK is received after transmission, a voice station doubles its contention window (until the maximum contention window size is reached), chooses a backoff timer, and continues to contend in the CP.

In a CP, if there is a voice contender, the data stations will sense the black burst during the *AIFS*[*AC_data*] (> *AIFS*[*AC_voice*]), and defer their transmissions. When a collision happens between voice stations, the data stations will wait for the channel to be idle for the duration of ACK timeout plus *AIFS*[*AC_data*] before they attempt to acquire the channel, which ensures that voice stations will not lose the channel access priority to the data stations even when a collision happens. Furthermore, when all the active voice stations are included in the polling list, the data stations can make full use of the CP resources.

Note that by using the above scheme, the waiting time (before getting the channel) of a voice station is larger than that in EDCA, since the voice station with the largest backoff timer instead of the smallest backoff timer (as in EDCA) gets the channel. However, as the number of voice stations contending for the channel simultaneously is very likely small, the initial and maximum window sizes for voice AC can be set to small values, so the negative effect of our scheme should be negligible.

Overhead Reduction

To support voice over WLANs, it is important to reduce the overhead to improve the transmission efficiency over the radio link. In the following, we propose two methods to reduce the header overhead and control overhead, respectively.

Header Overhead Reduction

The large packet header overhead can significantly affect the capacity of WLAN to support voice service. For example, if a GSM 6.10 codec is used, a voice packet payload is 33 bytes, while the RTP/UDP/IP overheads are 40 bytes. In addition, the PHY preamble, MAC header (36 bytes), and control packets all consume bandwidth. As a result, the overall efficiency is less than 3 percent [10]. Actions need to be taken to alleviate the effect of the overhead.

In the literature, various header compression techniques for VoIP have recently been proposed. The RTP/UDP/IP headers can be compressed to as little as 2 bytes [11]. The compression technique is adopted in our research.

In our proposed scheme the MAC layer header overhead is further reduced by aggregating the buffered voice packets of a voice station together and transmitting them in one MAC frame. The AP polls each voice station periodically after every service interval, which depends on the minimum delay bound of voice traffic. Within each service interval, several voice packets may be generated and buffered by each voice station. In order to increase the efficiency, we combine the payload of these packets together and add a common MAC layer header instead of sending them one by one. It reduces the overall MAC layer header and PHY preamble overhead.

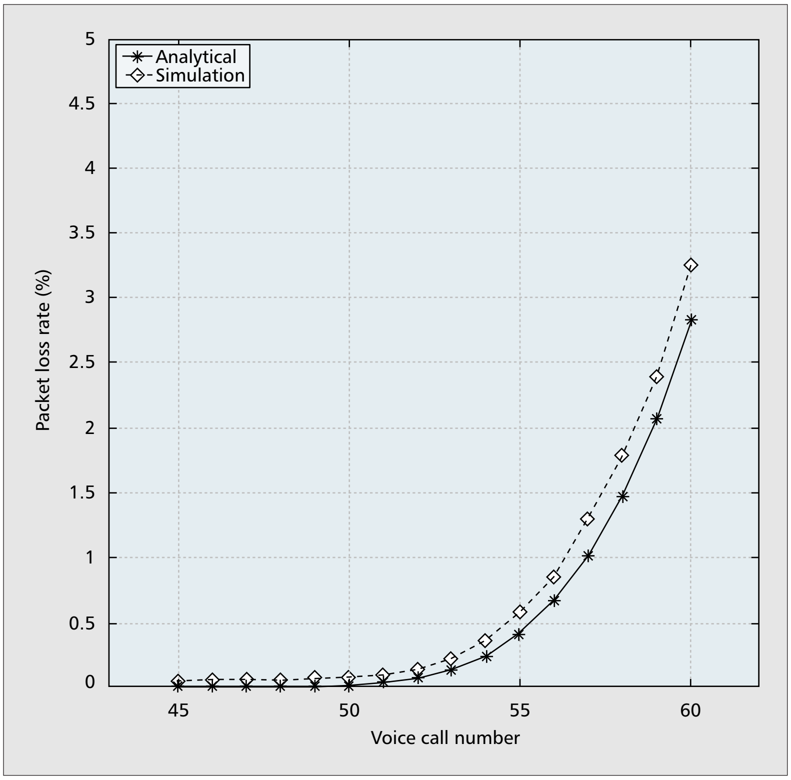
Control Overhead Reduction

In the IEEE 802.11e draft, another type of overhead in HCF is due to frequent poll frames from the AP to mobile stations. The AP sends a QoS CF-Poll frame to each station (according to its polling list) one by one to grant the TXOP. However, this polling method is inefficient due to the large overhead. According to the IEEE 802.11 standard, the CF-Poll frame is required to be transmitted at the basic rate (2 Mb/s as an example in this article) regardless of the data rate. The size of CF-Poll frame is 36 bytes in the 802.11e draft, including 10 bytes for frame/sequence/QoS control and frame check sequence (FCS), 24 bytes for station ID, and 2 bytes for duration ID. Its transmission time at 2 Mb/s is $36 * 8/2 = 144$ µs. Furthermore, considering the PHY overhead (192 µs), the total transmission time is 336 µs. The transmission time for a 69-byte voice packet (33-byte payload with a GSM 6.10 codec and 36-byte MAC header) at 11 Mb/s data rate is $69*8/11 = 50.2$ µs. Compared to the voice packet transmission time, the overhead contributed by the CF-Poll frame is quite large. For a WLAN accommodating N voice users, the total overhead contributed by CF-Poll is $336 * N$ µs, which is significant. On the other hand, if we combine the CF-Poll frames of N voice users into one super CF-Poll frame, we can reduce the overhead significantly. The super CF-Poll frame keeps an entry of 24-byte station ID and 2-byte transmission duration for each active voice station. The super CF-Poll frame is of variable size, depending on the number of voice stations being polled. The order of the station address (or station ID) contained in the super CF-Poll frame indicates the transmission order of the stations, and transmission duration field associated with each station specifies its maximum transmission time for each transmission opportunity. For N voice users, the size of a super CF-Poll frame is $10 + (24 + 2)*N$ bytes. For $N = 20$, the transmission time for a super CF-Poll frame is 2312 µs, significantly less than the total transmission time for 20 CF-Poll frames (6720 µs).

In order to further improve efficiency, we omit the super CF-Poll frame if it is the same as the one in the last service interval, or use a small frame to indicate no change. Each voice station records the content of the last super CF-Poll frame. If there is no super CF-Poll frame sent in the current service interval, the voice stations will follow the last super CF-Poll frame to determine the order of their transmissions.

The IEEE 802.11e draft allows a CF-Poll to be piggybacked with a DATA (or ACK) frame. When the AP has DATA (or ACK) to send to a station and at the same time wants to poll it, the AP would send a DATA + CF-Poll (or ACK + CF-Poll) frame. In our scheme a super CF-Poll frame is used instead of sending the DATA + CF-Poll (or ACK + CF-Poll) frame for the following reasons. First, our scheme supports voice

In the literature, various header compression techniques for VoIP have recently been proposed. The RTP/UDP/IP headers can be compressed down to as little as 2 bytes. The compression technique is adopted in our research.



■ **Figure 3.** *Packet loss rate in IEEE 802.11e when statistical multiplexing is considered in the admission algorithm.*

Parameter	Value
DATA transmission rate	11 Mb/s
Basic rate	2 Mb/s
Average talk spurt period	352 ms
Average silent period	650 ms
Service interval *SI*	100 ms
Beacon interval *T*	100 ms
T_{CP}/T	20%
P_L	1%

Table 1. *Simulation parameters.*

and data traffic, and only voice stations are polled. For a voice conversation, when one side is speaking, the other side is very likely to be silent. Hence, the AP and voice station are not likely to have voice packets to send to each other simultaneously. Second, as required by the IEEE 802.11e draft, if a CF-Poll is piggybacked with a DATA frame, the DATA + CF-Poll frame should be transmitted at the basic rate (regardless of the data rate) in order to set the network allocation vector (NAV) of all stations that are not being polled. Therefore, the DATA may be transmitted at a rate that is below the negotiated minimum PHY rate. Third, since voice traffic is delay-sensitive and can tolerate a certain level of packet loss, an ACK/retransmission mechanism is not suitable for voice traffic.

Call Admission Control for the Controlled HCF

In order to guarantee QoS of voice traffic, it is critical to have an appropriate call admission control mechanism. The AP is responsible for admitting or rejecting a new voice call based on the available resources to ensure that all admitted voice calls are satisfied with their QoS requirements such as delay and packet loss rate. The IEEE 802.11e draft has given a reference design for admission control [4]. When there are n existing voice stations in a BSS, a new voice call indexed by $n + 1$ can be admitted if the following inequality holds:

$$\frac{TXOP_{n+1}}{SI} + \sum_{i=1}^{n} \frac{TXOP_i}{SI} \le 1 - \frac{T_{CP}}{T}, \tag{1}$$

where T is the beacon interval, T_{CP} the minimum time used for EDCA during each beacon interval, SI the service interval, and $TXOP_i$ the minimum time that needs to be allocated for call i to ensure its QoS requirements.

Such admission control algorithm is only suitable for constant-rate voice traffic without statistical multiplexing. Based on the algorithm, variable-rate voice traffic (represented by the `on/off` model) requires much more resources than what is actually needed. Here, we propose another admission control algorithm which takes into account statistical multiplexing and, at the same time, guarantees the delay and packet loss rate requirements of voice traffic. By choosing a proper service interval, voice traffic delay can be guaranteed by the controlled polling mechanism. Given the size of T, T_{CP} and SI, we can calculate the maximum number of voice packets that can be accommodated in each service interval, denoted by N_p. Given N_p, the objective of our admission control algorithm is to find the maximum number of voice calls (n) that can be admitted for a pre-set packet loss rate bound P_L. Let X_i denote the number of voice packets generated by the ith user during a service interval, and $Y = \Sigma_{i=1}^{n} X_i$. Then the following inequality should hold:

$$\frac{\sum_{Y>N_p} (Y - N_p) P(Y)}{E[Y]} \le P_L. \tag{2}$$

According to the central limit theory, the random variable $Y = \Sigma_{i=1}^{n} X_i$ can be approximated as a Gaussian random variable with mean $n \cdot \mathrm{E}[X_i]$ and variance $n \cdot \mathrm{Var}[X_i]$ when n is large. $\mathrm{E}[X_i]$ and $\mathrm{Var}[X_i]$ can be derived based on the on/off voice model. The detailed procedure to solve the above inequality is omitted here due to mathematical complexity.

Performance Evaluation

Computer simulations are carried out to evaluate the performance of our proposed polling scheme and to validate the analysis of our call admission control algorithm. We choose the GSM 6.10 codec as the voice source as an example. The voice payload size is 33 bytes, and the packet interarrival period is 20 ms. We use compressed RTP/UDP/IP headers of 4 bytes in all the simulations. Other simulation parameter values are listed in Table 1. The simulation for each case runs for 3000 service intervals, and the statistics are collected in the last 2900 service intervals.

The performance improvement from applying statistical multiplexing is evaluated first. Through the referenced admission algorithm in IEEE 802.11e, the maximum admitted call number is found to be 27. However, if our admission algorithm is applied where statistical multiplexing is considered, many more voice calls (56) can be admitted with bounded (< 1 percent) packet loss rate. This can be validated by the simulated results of voice packet loss rate vs. number of voice calls in service, as shown in Fig. 3. Note that the maximum number of voice calls derived in [10] is around 11, much smaller than that using our algorithm. The reason is that contention-based EDCA is considered in [10]. The extra contention-associated overhead due to collision and idle time slots degrades system capacity. In HCF, the polled voice packets do not need to contend for the channel (except the first few packets in each talk spurt), so the contention-associated overhead is negligible. Therefore, the capacity is larger in the controlled polling access, indicating that controlled access is more suitable for voice traffic than contention-based access.

When all the mechanisms in our scheme are applied, by analysis we find that the maximum number of admitted voice calls can be increased significantly from 56 to 245. In our simulation we trace how packet loss rate changes with number of voice calls, as shown in Fig. 4 where the analytical result is included for comparison. It can be seen that the simulation results match the analytical results well.

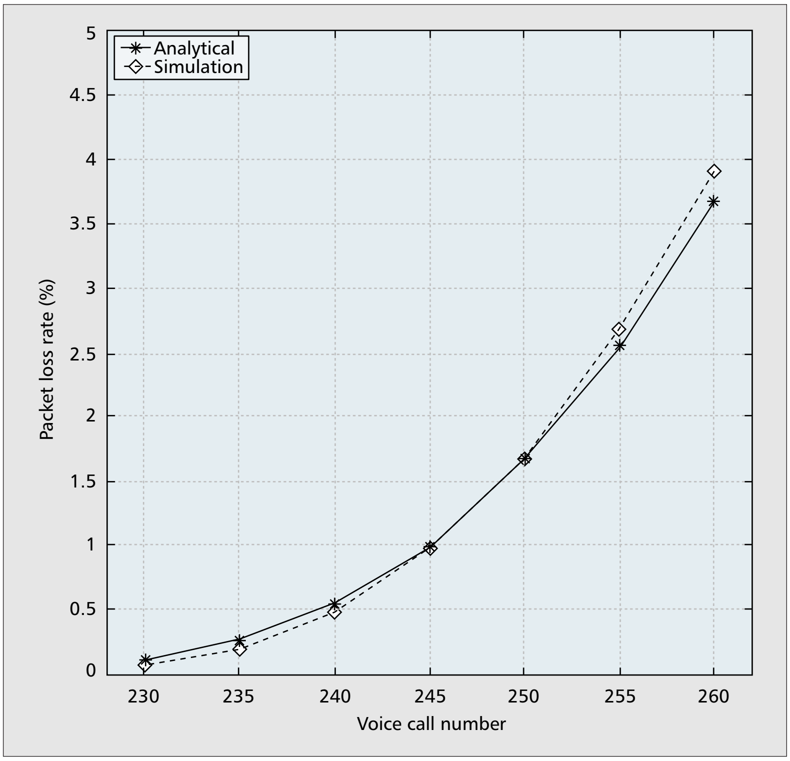


Figure 4. *Packet loss rate in our proposed scheme.*

Conclusion

Polling in IEEE 802.11e is effective to meet the delay requirements of VoWLAN applications. In this article we address capacity enhancement for WLANs supporting voice services. Our proposed solution avoids unnecessary polling of silent voice stations, and suppresses header and control overheads significantly. This research can be smoothly incorporated in the implementation of IEEE 802.11e as only minor modifications are needed.

Acknowledgments

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