Supporting VoIP Services in IEEE 802.11e WLANs*

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Abstract. Voice over Internet Protocol (VoIP) over Wireless Local Area Network (WLAN) is becoming popular thanks to its cost efficiency. However, it has been a challenge to provide good quality of VoIP services in WLANs, which is due mainly to (i) the nature of contention-based channel access of WLAN Medium Access Control (MAC); (ii) the presence of coexisting non-real-time data traffic; and (iii) the time-varying WLAN capacity caused by transmission rate diversity and variation of stations over time. In this paper, we propose a simple, effective and viable solution to improve the quality of VoIP services in 802.11e contention-based WLANs, which basically utilizes the advanced features of 802.11e MAC for QoS support. The key ingredients of our solution include (i) a priority queue to serve the VoIP traffic with higher priority than the non-real-time data traffic; and (ii) a conservative history-based admission control scheme for VoIP services, which accommodates the transmission rate diversity and variation of ongoing VoIP sessions over time. Simulation results demonstrate that our solution admits as many VoIP calls as possible without compromising the quality of their services.

Keywords: IEEE 802.11e EDCA, VoIP, QoS.

1 Introduction

Voice over IP (VoIP) and IEEE 802.11 Local Area Network (WLAN) have seen tremendous growth in recent years. IEEE 802.11 WLAN [1] has become the dominant technology for indoor broadband wireless networking. VoIP has been widely adopted in the enterprise and residence environments thanks to the various advantages such as a lower-cost, easy setup, and the integration of voice and data networks. The emergence of many VoIP vendors and VoIP service providers such as Skype [3] also speeds up the usage of VoIP.

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How to provide high Quality of Service (QoS) for VoIP applications in 802.11 WLANs has received considerable research attention. Originally, the 802.11 WLAN was designed to support best-effort services which do not have stringent QoS requirements, such as Internet-based Non-Real-Time (NRT) data services like Web browsing, e-mail, and file transfer. Therefore, many efforts to support the QoS in legacy 802.11 WLAN have been made [4, 5, 6]. However, those still have inherent inefficiencies such as the lacks of admission control, QoS signaling, differentiated channel access, and so on.

The 802.11e [2], which is an amendment to the legacy 802.11 Medium Access Control (MAC), was designed with the aim to support QoS [7, 8, 9]. The 802.11e MAC is expanding the 802.11 application domain by enabling Real-Time (RT) services such as voice and video services. The 802.11e MAC protocol is called the Hybrid Coordination Function (HCF), which contains a contention-based channel access mechanism (EDCA). EDCA is an enhanced version of the legacy Distributed Coordination Function (DCF) for QoS support. Most of the off-the-shelf 802.11e-compliant products, which are certificated by the Wi-Fi alliance [10], implement EDCA.

However, even when the 802.11e EDCA is employed, there are still some challenges to provide high-quality VoIP service in WLANs as follows: (i) the difficulty in quantitatively controlling channel occupancy of stations due to the contention-based channel access of EDCA; (ii) the presence of coexisting NRT data traffic; and (iii) the time-varying WLAN capacity caused by transmission rate diversity and variation of stations over time.

In order to address the above issues, we propose an effective and standard-compliant solution for improving the quality of VoIP services in 802.11e contention-based WLANs. It utilizes the advanced features in the 802.11e such as service differentiation mechanism and admission control framework, and consists of the following components: *Priority Queuing (PQ)* and *Call Admission Control (CAC).*

(1) Priority Queueing (PQ): Different from the legacy 802.11 MAC with a single First-In-First-Out (FIFO) transmission queue, an EDCA MAC contains multiple queues with different channel access priorities. This means that, when a WLAN carries a mixed traffic of voice and NRT data packets, the 802.11e EDCA MAC is able to provide differentiated services to VoIP applications which have stringent QoS requirements. The idea of PQ is to give voice queue strictly higher priority than NRT data queue by assigning proper channel access parameters to each of the queues so that NRT data queue can access the channel only when RT queue is empty. Moreover, in order to mitigate the bottleneck issue at the AP, which limits the VoIP capacity [11], we adopt a simple contention-free access scheme for the AP (called PIFS Access) by controlling channel access parameters of its AC_VO queue.

(2) Call Admission Control (CAC): One of the key elements in improving the quality of VoIP services is effective call admission control, which determines whether to admit a new VoIP call based on the available capacity of the WLAN,

so as to maintain the QoS of admitted VoIP calls while accommodating as many new calls as possible [12]. However, it is nontrivial to obtain an accurate estimation of the available WLAN capacity. This is because the link conditions between the AP and stations fluctuate due to multipath fading and/or user mobility. In this paper, we propose an admission control scheme based on the framework provided in the 802.11e standard. The proposed scheme predicts the future transmission rates of ongoing VoIP sessions based on their transmission histories and then calculates the expected amount of VoIP service time to determine the admission of a new VoIP call. Moreover, it limits the channel occupancy times of the admitted VoIP sessions by assigning the maximum allowable channel access time to each admitted VoIP session, which is referred to as *Medium Time (MT)* and derived based on its QoS requirements and transmission rate.

The rest of this paper is organized as follows. Section 2 introduces the EDCA admission control framework and discusses the necessities of admission control for VoIP services over IEEE 802.11e WLAN. The details of the proposed solution are described in Section 3. Section 4 presents the simulation results, and the paper concludes in Section 5.

2 TSPEC and EDCA Admission Control for VoIP Services

In an IEEE 802.11e WLAN, a VoIP station sets up a virtual connection, called *Traffic Stream (TS)*, with the AP before commencing any actual voice packet transfer in order to provide the prescribed QoS for its VoIP call. The admission controller located at the AP determines whether to admit a new VoIP call based on the available capacity of the WLAN and the QoS requirement of the VoIP call. If the new VoIP call is admitted, the corresponding VoIP TS is set up between the new VoIP station and the AP.

The QoS requirement and traffic characteristics of a TS, called *Traffic Specification (TSPEC)*, usually can be provided from the application layer, e.g., VoIP application, via station management entity (SME), which is a cross-layer entity and can internally communicate with multiple protocol layers. The TSPEC is submitted to the admission controller located at the AP by a station when it requests the admission of its VoIP call and wants to set up the corresponding TS. Then, the TSPEC is used by the admission controller to make the admission decision.

Fig. 1 shows the admission control and VoIP TS setup procedure for VoIP services, which is based on the 802.11e standard [2]. An ADDTS Request frame, which conveys the TSPEC element, is transmitted by a VoIP station to the AP in order to request a VoIP TS setup. The TSPEC consists of several parameters like Nominal MSDU Size, Mean Data Rate, Delay Bound, Minimum PHY Rate, Medium Time, and so on. Most of parameters except *Medium Time (MT)*, which is the amount of time allowed for the corresponding VoIP station to access the medium per one-second period, are specified by a VoIP station and delivered to the AP when it requests the admission of its VoIP call. If the AP decides to

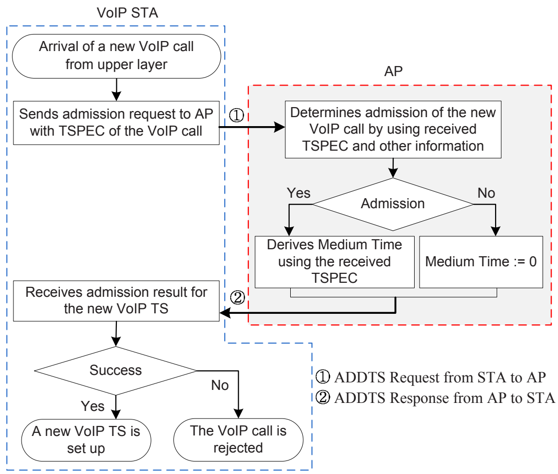


Fig. 1. Traffic Stream (TS) setup and EDCA admission control procedure for VoIP services

accept the request, the AP derives the MT from the parameters conveyed in the TSPEC element of the ADDTS request frame. Then, the AP sends the derived MT to the requesting VoIP station via an ADDTS Response frame.

After receiving the ADDTS Response frame, the admitted VoIP station i records the MT as Admitted Time $\mathcal{A}_i$ by $\mathcal{A}_i = \mathcal{M}_i \cdot t_a$, where $\mathcal{M}_i$ is the MT of VoIP TS i and t_a is an averaging period. $\mathcal{A}_i$ represents the maximum amount of time that the station can use to transmit packets belonging to the corresponding VoIP TS within every t_a-second time window, where a design parameter t_a $(1 \leq t_a \leq 100)$ is an integer [2]. For each packet transmission, a VoIP station increases the *Used Time* ($\mathcal{U}$), which is the amount of time used to attempt VoIP packet transmissions, and if $\mathcal{U}_i$ is larger than or equal to $\mathcal{A}_i$, VoIP station i cannot transmit more voice packets via AC_VO until the next t_a interval.[1]

Now, the problems left for admission control are as follows: (i) how to decide the admission of a new VoIP call (i.e., a VoIP TS); and (ii) how to derive the MT of admitted VoIP TS. The proposed solutions to these problems are presented in Section 3.

3 Proposed Solution for Improving Quality of VoIP Services in 802.11e EDCA

Our objective is to improve the quality of VoIP services in IEEE 802.11e WLANs. To achieve this goal, we propose a solution that implements the following modules at the AP.

[1] Actually, the VoIP station may transmit voice packets via other Access Categories (ACs) where no admission control is required such as Best Effort Access Category (AC_BE). However, in this paper, we assume that voice packets are transmitted via the Voice Access Category (AC_VO) only.

- *Priority Queueing via Controlling Channel Access Parameters* to provide service differentiation between voice traffic (i.e., AC_VO) and NRT data traffic (i.e., AC_BE); and
- *Call Admission Control* to control the admission of new VoIP calls into the network and to efficiently allocate MTs to admitted VoIP sessions.

Fig. 2 shows the system architecture of the AP with our proposed solution. A packet from the upper layer can be classified into one of four ACs based on various parameters such as Ethernet, TCP/IP, and IEEE 802.1D/Q parameters [2, 13].

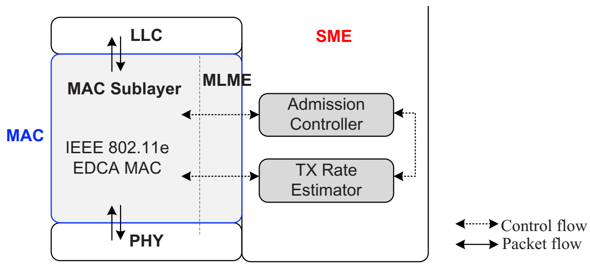


Fig. 2. Overview of the proposed solution. MLME is MAC Layer Management Entity.

TX Rate Estimator caches the time used to attempt the transmission of a voice packet belonging to each admitted VoIP session, including the wasted time for a failed transmission attempt. When a new VoIP call requests its admission, *TX Rate Estimator* estimates the future transmission rates of all the currently ongoing VoIP TSs based on the cached history. The estimation algorithm is presented in Section 3.3. Based on the rate estimation and an assumption that the new VoIP call transmits at its minimum PHY rate if admitted, *Admission Controller* calculates the total VoIP service time, determines the admission of the new VoIP call, and then calculates the MT of each admitted VoIP TS. Moreover, while calculating the total VoIP service time, it determines the channel access parameter values for AC_VOs of stations that can minimize the channel time occupied by VoIP traffic, thus increasing the channel utilization for AC_BE (i.e., NRT traffic). Then, the AP updates channel access parameters of stations and MTs of all the admitted VoIP TSs. The calculation details of the VoIP service time and admission control algorithm is presented in Section 3.3.

3.1 System Model and Assumptions

We consider G.711 [14] – the simplest voice codec. Note that our analysis and design could be applied to other voice codecs as well. The G.711 codec generates a 64 kbps data stream, based on an 8-bit Pulse Coded Modulation (PCM), with the sampling rate of 8000 samples/s. We assume that Voice Activity Detection (VAD) is not used, which means that the VoIP traffic is a CBR (Constant Bit

Table 1. IEEE 802.11b PHY and VoIP Parameters

Parameter	Values
Slot Time (σ)	20 μs
SIFS	10 μs
PIFS	30 μs
PHY Overhead ($\mathcal{O}_{\mathrm{PHY}}$)	192 μs
MAC Overhead ($\mathcal{O}_{\mathrm{MAC}}$)	30 bytes
ACK Length ($\mathcal{L}_{\mathrm{ACK}}$)	14 bytes
Voice Packet MSDU Size ($\mathcal{L}_{\mathrm{voice}}$)	Voice Data (160) + RTP Header (12) + UDP/IP Headers (28) + SNAP Header (8) = 208 bytes
Mean Data Rate (ρ_{voice})	208 bytes / 20 ms = 83.2 kbps

Rate) traffic. We assume that a voice packet is generated every 20 ms. Thus, the amount of voice data carried in a packet is 160 bytes = 8000 samples/s $\times$ 8 bits/sample $\times$ 20 ms. Real-Time Protocol (RTP) over User Datagram Protocol (UDP) is usually used for the VoIP transfer. When an IP datagram is transferred over the 802.11 WLAN, it is typically encapsulated by the IEEE 802.2 Sub-Network Access Protocol (SNAP). Accordingly, the size of a voice packet at the 802.11 MAC Service Access Point (SAP),[2] i.e., the voice packet MSDU (MAC Service Data Unit) size, is 208 bytes, as shown in Table 1.

3.2 Priority Queueing via EDCA Parameter Setting

In this paper, we consider only AC_VO and AC_BE out of four ACs for simplicity. For service differentiation between AC_VO and AC_BE, we use the strict priority queueing by properly setting the channel access parameters of the two ACs. Moreover, we adopt a simple contention-free access (called PIFS access) for the AP's AC_VO [15], which allows the AC_VO of the AP to transmit a pending voice packet after a PIFS idle time without any contention.

Parameter Setting for AC_VO. In the 802.11e standard [2], AIFSN[AC_VO], which is the arbitration interframe space number for AC_VO, is an integer greater than 1 for stations and an integer greater than 0 for the AP [2]. Moreover, the values of minimum and maximum contention window size for AC_VO, i.e., CW_{min}[AC_VO] and CW_{max}[AC_VO], can be set to zero. Therefore, for AC_VO of the AP, we can use the smallest access parameter values of AIFSN[AC_VO] = 1 and CW_{min}[AC_VO] = CW_{max}[AC_VO] = 0 for downlink voice packet transmissions. This allows the AC_VO of the AP to transmit the pending voice packets after a PIFS idle time without backoff. This scheme is referred to as *PIFS Access* for the rest of this paper.

[2] MAC SAP is the interface between the MAC and the higher layer, i.e., the IEEE 802.2 Logical Link Control (LLC) layer.

On the other hand, AC_VO of a station uses AIFSN[AC_VO] = 2, which is the smallest value for a station, and both $\mathrm{CW}_{\min}$[AC_VO] and $\mathrm{CW}_{\max}$[AC_VO] are set to a properly chosen value based on the given parameters, i.e., the number of VoIP stations and their transmission rate distribution. How to find the proper $\mathrm{CW}_{\min}$[AC_VO] value is presented in Section 3.3. $\mathrm{CW}_{\max}$[AC_VO] uses the same value as $\mathrm{CW}_{\min}$[AC_VO] so that delay and delay jitter performance of VoIP traffic can be improved without doubling the contention window size after a transmission failure.

Parameter Setting for AC_BE. In order to prevent AC_BE from accessing the channel while AC_VO has any voice packet to transmit, an AC_BE uses AIFS[AC_BE] value as follows:

$$\mathrm{AIFS[AC\_BE]} = \mathrm{AIFS[AC\_VO]} + \mathrm{CW}_{\min}\mathrm{[AC\_VO]} \cdot \sigma, \tag{1}$$

where σ is a backoff slot time and AIFS[AC_BE] and AIFS[AC_VO] are the arbitration interframe space for AC_BE and AC_VO, respectively. Therefore, after a channel busy period, an AC_BE can start its backoff only if there is no AC_VO with pending packets. For $\mathrm{CW}_{\min}$[AC_BE] and $\mathrm{CW}_{\max}$[AC_BE], the default values provided in the standard [2] are used.

3.3 Conservative Admission Control for VoIP Services (CAVS)

We propose a history-based admission control scheme, called CAVS (Conservative Admission control for VoIP Services), to accommodate the transmission rate diversity and variation of VoIP stations over time. The key ideas of CAVS are (i) caching the recent transmission results of the admitted VoIP sessions; (ii) determining the admission of a new VoIP call based on a conservative history-based estimation of future transmission rates of the admitted VoIP sessions; and (iii) deriving/updating MTs of the admitted VoIP stations.

Transmission History Cache. In CAVS, whenever the AP finishes a packet transmission attempt, it caches the result as a quadruplet

$$\Theta : (t_{\mathrm{event}}, session\_id, result, time\_usage), \tag{2}$$

where t_{event} is the time instance when the AP started to transmit the packet, *session_id* is the ID of the corresponding VoIP session, *result* is 1 if the transmission was successful, 0 otherwise, and *time_usage* is the time used to complete the transmission attempt, i.e.,

$$time\_usage = \begin{cases} T_{\mathrm{voice}} + \mathrm{SIFS} + T_{\mathrm{ack}}, & \text{if TX success,} \\ T_{\mathrm{voice}} + ACKTimeout, & \text{if TX failure,} \end{cases} \tag{3}$$

where T_{voice} and T_{ack} are the transmission durations of a voice packet and a ACK frame, respectively. The AP caches the recent transmission results of the admitted VoIP TSs, and removes stale data from its cache. More specifically, the AP only caches results with $t_{\mathrm{event}} > t_0 - t_{\mathrm{win}}$, where t_0 is the current time and a design parameter t_{win} is the estimation window size. Based on this cached information, when a new VoIP call requests its admission, the AP estimates the future transmission rates of the admitted VoIP TSs.

Table 2. Time used to complete a successful transmission attempt of a voice packet at each rate of the 802.11b PHY

TX Rate r^* (Mbps)	1	2	5.5	11
$T_{\text{succ}}(r^*)$ (μs)	2394	1394	793	622

Transmission Rate Estimation. In CAVS, when a new VoIP call requests its admission, the AP first calculates the average time used to attempt a successful voice packet transmission for each admitted VoIP session, based on the cached information. For the admitted VoIP session i, it is $T_{\text{avg},i} = \frac{T_{\text{total},i}}{N_{\text{succ},i}}$, where

$$\begin{cases} T_{\text{total},i} = \sum\limits_{\substack{t_0 - t_{\text{win}} < \Theta.t_{\text{event}} \leq t_0 \\ \Theta.session\_id \,=\, i \\ \Theta.result \,=\, 1}} \Theta.time\_usage, \\ N_{\text{succ},i} = \sum\limits_{\substack{t_0 - t_{\text{win}} < \Theta.t_{\text{event}} \leq t_0 \\ \Theta.session\_id \,=\, i}} \Theta.result. \end{cases} \tag{4}$$

Then using $T_{\text{avg},i}$, the AP estimates the future transmission rate of the admitted VoIP session i as follows:

$$r_{\text{next},i} = \begin{cases} \min\left\{r_m^*,\ r_{\text{curr},i}\right\}, \text{ if } T_{\text{avg},i} \leq T_{\text{succ}}\left(r_{m-1}^*\right), \\ \min\left\{r_j^*,\ r_{\text{curr},i}\right\}, \text{ if } T_{\text{succ}}\left(r_j^*\right) < T_{\text{avg},i} \leq T_{\text{succ}}\left(r_{j-1}^*\right), \\ \qquad\qquad\qquad\quad \text{where } j = 2, \cdots, m-1, \\ \min\left\{r_1^*,\ r_{\text{curr},i}\right\}, \text{ if } T_{\text{avg},i} > T_{\text{succ}}\left(r_1^*\right), \end{cases} \tag{5}$$

where m is the number of available transmission rates, $r_{\text{curr},i}$ is the current transmission rate of VoIP session i, and $T_{\text{succ}}\left(r^*\right)$ is the time used to complete a successful transmission attempt of a voice packet at rate r^*:

$$T_{\text{succ}}\left(r^*\right) = T_{\text{voice}}(r^*) + \text{SIFS} + T_{\text{ack}}. \tag{6}$$

For example, for the 802.11b PHY, $m = 4$ and $r_1^*, \cdots, r_4^*$ are 1 Mbps, 2 Mbps, 5.5 Mbps, and 11 Mbps, respectively, and the corresponding T_{succ} values are listed in Table 2.

In general, Eq. (5) works fine in predicting the future transmission rate with random station movement. However, under certain circumstances, it may not perform well. For example, if the VoIP station of the admitted session i keeps moving away from the AP, $r_{\text{est},i}$ is then not a good estimation of session i's future transmission rate. Therefore, we consider the current transmission rate of each admitted VoIP session in the final estimation of its future rate as follows:

$$r_{\text{next},i} = \min\left\{r_{\text{est},i},\ r_{\text{curr},i}\right\}, \tag{7}$$

where $r_{\text{curr},i}$ is the current transmission rate of VoIP session i.

Worst-Case Analysis of VoIP Service Time for Admission Control. Our admission control scheme requires the quantified calculation of the VoIP service time. In this subsection, we analyze the VoIP service time during $\mathcal{T}_v$, which is the voice packet generation interval, under the worst-case scenario for uplink VoIP packet transmissions when the AP uses PIFS Access for its downlink VoIP packet transmissions.

We make the following assumptions to simplify the problem. First, we assume that all the uplink and downlink voice packets arrive synchronously at the WLAN. Therefore, at the time instance when voice packets arrive at the WLAN, the network is temporarily congested, which is the worst-case scenario in terms of channel contention. Second, we assume the p-persistent model for the EDCA [16], instead of using the binary exponential backoff. Third, we assume that different stations transmit at different rates and the finishing order of VoIP stations' uplink packet transmissions is in the descending order of the packet transmission rate. This leads to longer VoIP service time because the wasted time in collision is determined by the longest transmission duration among packets that are involved in the collision. Fourth, we assume that there is no hidden station in the network. Finally, to simplify the analysis, we assume that the channel condition between the AP and a station is symmetric and hence the AP uses the same transmission rate as the station for the VoIP session between them.

When the number of active VoIP sessions is equal to N, the average VoIP service time $\mathcal{S}_{\text{voice}}(N)$ can be expressed as

$$\begin{aligned}\mathcal{S}_{\text{voice}}(N) &= T_N^{\text{PIFS}} + \sum_{i=1}^{N} T_s\left(i\right) \\ &\quad + \sum_{k=1}^{N} \left(\left(E\left[N_k^{\text{col}}\right] + 1\right) E\left[I_k\right] \sigma + E\left[N_k^{\text{col}}\right] E\left[T_{c,k}\right]\right),\end{aligned} \tag{8}$$

where T_N^{PIFS} is the total transmission time to transmit N downlink voice packets via PIFS access, which is given by

$$T_N^{\text{PIFS}} = \sum_{i=1}^{N} (\text{PIFS} + T_{\text{voice}}\left(i\right) + \text{SIFS} + T_{\text{ack}}), \tag{9}$$

and $T_s(i)$ is the successful transmission time of an uplink VoIP packet belonging to VoIP session i, and it is given by

$$T_s(i) = T_{\text{voice}}(i) + \text{SIFS} + T_{\text{ack}} + \text{AIFS[AC\_VO]}. \tag{10}$$

$E[I_k]$ is the average number of idle backoff slots preceding a collision or the successful transmission, $E[N_k^{\text{col}}]$ is the average number of collisions preceding the successful transmission, and $E[T_{c,k}]$ is the average collision time when the number of contending stations is k. $E[I_k]$, $E[N_k^{\text{col}}]$, and $E[T_{c,k}]$ can be derived as follows.

$E[I_k]$ is given by

$$E\left[I_k\right] = \sum_{i=0}^{\infty} iP\left(I_k = i\right) = \frac{\left(1-p\right)^{N-k+1}}{1-\left(1-p\right)^{N-k+1}}, \tag{11}$$

where $P(I_k = i)$ is the probability that there are i idle backoff slots preceding a busy period (i.e., a collision or a successful transmission), and it is given by

$$P(I_k = i) = \left((1-p)^{N-k+1}\right)^i \left(1-(1-p)^{N-k+1}\right), \tag{12}$$

where p is the channel access probability of a station. In this paper, we use a fixed p value of $\frac{2}{\mathrm{CW}_{\min}[\mathrm{AC\_VO}]+1}$. This makes a station more aggressive because the contention window size is not doubled even after a transmission failure. Accordingly, this results in a conservative estimation of the VoIP service time.

Moreover, $E[N_k^{\mathrm{col}}]$ can be calculated by

$$E\left[N_k^{\mathrm{col}}\right] = \sum_{i=1}^{\infty} iP(N_k^{\mathrm{col}} = i) = \frac{1-(1-p)^{N-k+1}}{(N-k+1)p(1-p)^{N-k}} - 1, \tag{13}$$

where $P(N_k^{col} = i)$ is the probability that there are i collisions preceding the successful transmission, and it is given by

$$P\left(N_k^{\mathrm{col}} = i\right) = \left(\frac{1-(1-p)^{N-k+1} - (N-k+1)p(1-p)^{N-k}}{1-(1-p)^{N-k+1}}\right)^i \times \frac{(N-k+1)p(1-p)^{N-k}}{1-(1-p)^{N-k+1}}. \tag{14}$$

Finally, the average collision time $E[T_{c,k}]$ is:

$$E\left[T_{c,k}\right] = \mathrm{AIFS[AC\_VO]} + \sum_{i=1}^{N-k} \sum_{j=i+1}^{N-k+1} \max\left(T_{\mathrm{voice}}(i), T_{\mathrm{voice}}(j)\right) P_{c,k}(i,j), \tag{15}$$

where $P_{c,k}(i,j)$ is the probability that the packets of stations i and j collide with each other, given that a collision occurs. It can be calculated by

$$P_{c,k}(i,j) = 1 \Big/ \binom{N-k+1}{2}, \; i = 1, ..., N-k, \; j = i+1, ..., N-k+1. \tag{16}$$

In Eq. (16), we assume that a collision is only caused by two stations' simultaneous transmissions because the probability that three or more stations transmit at the same time is very low [17], thus being negligible.

From now on, we analyze the parameters required to derive the medium time (MT) in Section 3.3 based on the above analysis. From Eq. (8), the average idle backoff time can be calculated by

$$\psi_N = \sum_{k=1}^{N} \left(E\left[N_k^{\mathrm{col}}\right] + 1\right) E\left[I_k\right] \sigma. \tag{17}$$

The average time wasted by VoIP station i due to its collisions until it successfully transmits its voice packet can be estimated as follows:

$$\mathcal{O}_{\text{sur},i} = \sum_{k=1}^{l_i} \sum_{j=0}^{\mathcal{C}_k} j \left(P_{c,k}\left(i\right)\right)^j T_{\text{fail}}(i), \tag{18}$$

where $P_{c,k}(i)$, the probability that VoIP station i's packet collides with another packet, given that a collision occurs when the number of contending stations is k, is given by

$$P_{c,k}\left(i\right) = \sum_{j=1, j\neq i}^{N-k+1} P_{c,k}\left(i,j\right), \tag{19}$$

and $T_{\text{fail}}(i) = T_{\text{voice}}(i) + \text{AIFS[AC\_VO]}$, which is the wasted transmission time by VoIP station i when its transmission collides. $\mathcal{C}_k = \left\lceil E\left[N_k^{\text{col}}\right]\right\rceil$ is the ceiled average number of collisions when the number of contending stations is k. VoIP station i finishes its voice packet transmission when the number of contending stations is l_i, where l_i is determined by station i's transmission rate and the worst case scenario, i.e, a higher-rate station finishes its transmission earlier. If the number of VoIP stations with the same transmission rates is more than one, each of them has a value averaged over their $\mathcal{O}_{\text{sur}}$'s.

The total transmission time overlapped by the colliding VoIP stations assuming that the collision is only caused by two stations can be estimated as follows:

$$\delta_N = \sum_{k=1}^{N} E\left[N_k^{\text{col}}\right] E\left[T_{c,k}^{\text{overlap}}\right], \tag{20}$$

where $E[T_{c,k}^{\text{overlap}}]$, the average overlapped transmission time when a collision occurs, is given by:

$$E\left[T_{c,k}^{\text{overlap}}\right] = \sum_{i=1}^{N-k} \sum_{j=i+1}^{N-k+1} \min\left(T_{\text{fail}}(i), T_{\text{fail}}(j)\right) P_{c,k}\left(i,j\right). \tag{21}$$

Admission Decision and Optimal $\mathbf{CW_{min}}$[AC_VO]. After estimating the future transmission rate of each admitted VoIP session using Eq. (5), the AP calculates the average service time $\mathcal{S}_{\text{voice}}(N+1, \text{CW}_{\min}[\text{AC\_VO}])$ using Eq. (8), where $(N+1)$ corresponds to N admitted VoIP TSs and one new VoIP call that requests the admission. Note that $\mathcal{S}_{\text{voice}}$ is the function of the number of admitted VoIP stations, their rate distribution, and $\text{CW}_{\min}$[AC_VO]. Here, when we calculate $\mathcal{S}_{\text{voice}}(N+1, \text{CW}_{\min}[\text{AC\_VO}])$, we assume that the new VoIP TS will transmit at its minimum PHY rate if it is admitted. Before the AP decides the admission, it finds the optimal value of $\text{CW}_{\min}$[AC_VO] (cw^*) as follows:

$$cw^* = \arg \min_{\text{CW}_{\min}[\text{AC\_VO}]} \mathcal{S}_{\text{voice}}\left(N+1, \text{CW}_{\min}[\text{AC\_VO}]\right), \tag{22}$$

where $\text{CW}_{\min}[\text{AC\_VO}] \in [0, 1023]$ is an integer. When an ongoing VoIP session ends, the AP also needs to find cw^* that minimizes $\mathcal{S}_{\text{voice}}$ for $(N-1)$ existing

VoIP sessions in order to increase the channel utilization of AC_BEs. cw^* is distributed to the stations via the upcoming beacon transmissions. Finally, the AP determines the admission of the new VoIP call as follows:

$$\begin{cases} \text{Admit, if } \mathcal{S}_{\text{voice}}(N+1, cw^*) < \phi_{\text{voice}} \mathcal{T}_v, \\ \text{Reject, otherwise,} \end{cases} \tag{23}$$

where ϕ_{voice} ($0 \leq \phi_{\text{voice}} \leq 1$) is the fraction of $\mathcal{T}_v$ reserved for VoIP traffic, which is a design parameter.

Calculation and Update of MT. If the AP decides to admit a new VoIP call, it derives the MTs of the ongoing VoIP TSs as well as the MT of the new VoIP TS as follows.

$$\mathcal{M}_i^{\text{new}} = \left\lceil \frac{\rho_{\text{voice}}}{8\mathcal{L}_{\text{voice}}} \right\rceil \cdot \text{SurplusMPDUTime}_i, \tag{24}$$

where SurplusMPDUTime_i, the amount of time needed to transport a voice packet belonging to VoIP TS i including the overhead due to transmission failures, is given by

$$\text{SurplusMPDUTime}_i = \mathcal{O}_{\text{sur},i} + T_{\text{voice}}(i) + \text{SIFS} + T_{\text{ack}}, \tag{25}$$

where $\mathcal{O}_{\text{sur},i}$, calculated by Eq. (18), is the expected amount of inevitable collision time wasted by VoIP station i until it successfully transmits its voice packet.

Remind that limiting the uplink channel access time of an admitted VoIP station via its MT is for the reduction of the impact of the station's transmissions on the QoS of other VoIP TSs when the station tries to overuse the channel time due to its lower transmission rate than its estimated rate in Eq. (5) or many retransmissions due to the channel errors. Since the admission decision of the latest VoIP TS was based on the estimated rates of other admitted VoIP TSs, if some admitted VoIP stations happen to use lower rates than their estimated rates after the admission decision of the latest VoIP TS, the network might be saturated, and hence the QoS of the admitted VoIP TSs might be severely degraded. For this goal of MT, the following condition always needs to be satisfied:

$$T_{N+1}^{\text{AP}} + T_{N+1}^{\text{idle}} + \sum_{i=1}^{N+1} \mathcal{M}_i \leq 1 + \Delta_{N+1}, \tag{26}$$

where $\mathcal{M}_i$ is the current MT of VoIP TS i, $T_{N+1}^{\text{AP}} = \lceil \frac{\rho_{\text{voice}}}{8\mathcal{L}_{\text{voice}}} \rceil T_{N+1}^{\text{PIFS}}$ is the expected amount of time allowed to the AP for its transmissions of downlink voice packets per one second, $T_{N+1}^{\text{idle}} = \lceil \frac{\rho_{\text{voice}}}{8\mathcal{L}_{\text{voice}}} \rceil \psi_{N+1}$ is the expected total amount of idle backoff time of VoIP stations per one second, and $\Delta_{N+1} = \lceil \frac{\rho_{\text{voice}}}{8\mathcal{L}_{\text{voice}}} \rceil \delta_{N+1}$ is the amount of the overlapped portion due to collisions among MTs of the admitted VoIP TSs per one second. Note that T_{N+1}^{AP}, T_{N+1}^{idle}, and Δ_{N+1} are derived based on the estimated rates of the admtted VoIP TSs by using Eqs. (9), (17), and (20), respectively.

The AP updates MTs of the ongoing VoIP TSs with the newly derived MTs from Eq. (24) in order to satisfy Eq. (26) by sending ADDTS Response frames to ongoing VoIP stations without receiving the corresponding ADDTS Request frame. Note that the MT of the new VoIP TS is delivered to the corresponding station via an ADDTS Response frame.

4 Performance Evaluation

In this section, we evaluate the effectiveness of the proposed solution by using the ns-2 simulator [18]. The simulated network topology is shown in Fig. 3, where multiple mobile VoIP stations are placed inside a square region with 160 meters on the diagonal. VoIP stations communicate with the remote voice gateway via the AP which sits at the center of the square region. Our proposed solution is implemented at the AP. VoIP stations transmit and receive voice packets only and each station carries a single traffic flow. VoIP traffic is modeled by a two-way CBR session with 208-byte MSDU size and 20 ms packetization interval (i.e., $\mathcal{T}_v$ = 20 ms) according to the G.711 voice codec. We use the ITU E-model [22, 23] to assess the quality of mouth-to-ear (m2e) voice communication. It gives an overall rating R to the quality of a phone call where $0 \leq R \leq 100$. A VoIP session with $R \geq 80$ is called a *satisfactory VoIP session*.

The IEEE 802.11b PHY is used in our simulation. Table 1 lists the 802.11b PHY parameters. We assume an AWGN (Additive White Gaussian Noise) wireless channel and the background noise level is set to -96 dBm. Moreover, we use the log-distance path loss model with path loss exponent of 4, and the empirical BER (Bit Error Rate) vs. SNR (Signal-to-Noise Ratio) curves provided by Intersil [19]. We use the random waypoint model [20] to simulate the mobility of VoIP stations. The random waypoint model assumes that a user's movement follows a walk-and-pause pattern; the user chooses a random destination and moves towards it at a randomly-chosen speed less than or equal to the maximum speed in a flat restricted region. In our simulation, the maximum speed for VoIP stations is 2.5 m/s and the movement of a VoIP station is restricted within the square region shown in Fig. 3. Moreover, all stations use Automatic Rate Fallback (ARF) [21] – a widely-implemented rate adaptation scheme in WLAN devices, unless specified otherwise.

We simulate 40 VoIP sessions in the network. VoIP sessions start successively every 2 seconds from the beginning of the simulation. The call duration of each VoIP session is 30 seconds. The total simulation time is 100 seconds for each mobility scenario. All the simulation results are averaged over 50 mobility scenarios. The system parameter t_a is set to 5 – the default value provided by 802.11 specification [2]. t_{win} is set to 5 unless specified otherwise.

Fig. 4 shows the R values of a VoIP session, which is the first admitted session in a simulation, and the number of admitted VoIP sessions over simulation time with or without CAVS. In both cases, PIFS Access is used for AC_VO of the AP and the wireline delay is 150 ms. We observe that, when CAVS is used, the R values are always higher than 80. On the other hand, when CAVS is not used,

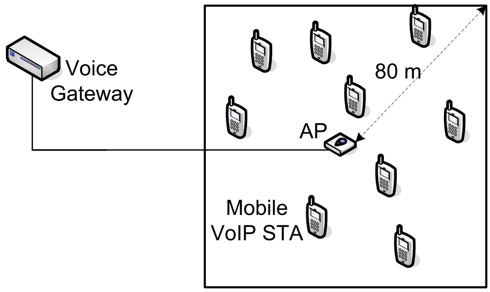
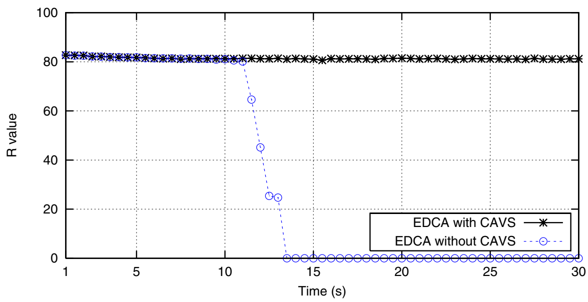
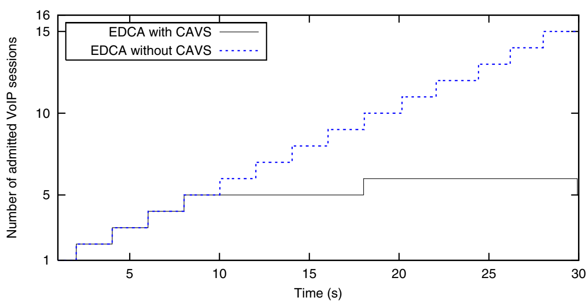


Fig. 3. The simulated network topology



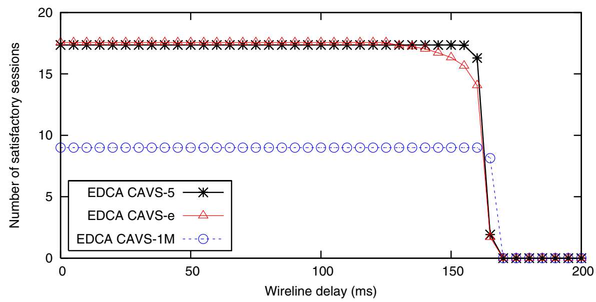
(a) R value of a VoIP session.



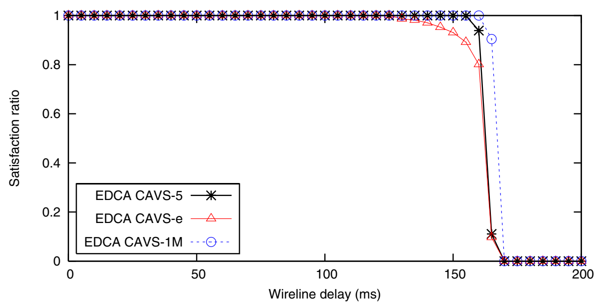
(b) The number of admitted VoIP sessions.

Fig. 4. Comparison of the cases with CAVS scheme and without CAVS scheme. The wireline delay is 150 ms.

the R values degrade severely since the network is overloaded due to the lack of admission control. As shown in Fig. 4(b), in the case of EDCA with CAVS, the number of admitted VoIP sessions is limited to 5 or 6 while, in the case of EDCA without CAVS, the number of admitted VoIP sessions increases continuously to overload the network.



(a) Total number of satisfactory VoIP sessions.



(b) Satisfactory ratio (= number of satisfactory VoIP sessions / number of admitted VoIP sessions).

Fig. 5. Comparison of CAVS schemes with different t_{win} values

To study the effect of the estimation window size (t_{win}), we simulate the following variants of CAVS:

- CAVS-t_{win}: the CAVS scheme which uses the estimation window size of t_{win} (in seconds) to estimate the future transmission rates of the admitted VoIP sessions.
- CAVS-ε: a special case of CAVS-twin; it makes an aggressive assumption that each admitted VoIP session always uses the current rate for future transmissions.
- CAVS-1M: a special case of CAVS-t_{win}; it makes a conservative assumption that each admitted VoIP session always transmits at the lowest 1 Mbps in the future.

Fig. 5(a) shows the total number of satisfactory VoIP sessions during the 100-second simulation with different CAVS schemes. Fig. 5(b) shows the *satisfactory ratio* which is the ratio of the number of satisfactory VoIP sessions to the total number of admitted sessions. As shown in Fig. 5(b), CAVS-1M results in the perfect satisfactory ratio (i.e., 1.0) before the wireline delay gets too large to prevent any satisfactory VoIP services. However, the actual number of admitted sessions for CAVS-1M is small (i.e., 9), as shown in Fig. 5(a). This is because CAVS-1M admits VoIP sessions very conservatively by assuming that all the admitted VoIP sessions will use the lowest transmission rate in the future. On the other hand, CAVS-ε admits VoIP sessions most aggressively among the considered schemes in our simulation. Note that, according to Eq. (7), future transmission rates of the admitted VoIP sessions estimated by CAVS-t_{win} (i.e., $r_{\mathrm{next},i}$) are always lower than or equal to those by CAVS-ε (i.e., $r_{\mathrm{curr},i}$). Such aggressive nature of CAVS-ε may result in incorrect admission decisions under certain circumstances, which could render a WLAN overloaded. Thus the service quality of the admitted VoIP sessions could be compromised and appears more sensitive to the increase in the wireline delay, as shown in the figures. In comparison, CAVS-5 perform better than CAVS-1M and CAVS-ε, which means that $t_{\mathrm{win}} = 5$ is a reasonable choice for our simulated network.

5 Conclusion

In this paper, we proposed a simple, effective and viable solution to support VoIP services in 802.11e contention-based WLANs, which basically utilizes the advanced features of 802.11e MAC for QoS support. Our solution includes a priority queueing scheme to serve VoIP traffic with higher priority than non-real-time data traffic, and a conservative history-based admission control scheme for VoIP services, which accommodates the transmission rate diversity and variation of ongoing VoIP sessions over time. We evaluate our proposed solution using the ns-2 based simulation. Simulation results demonstrate that our solution admits as many VoIP calls as possible without compromising the quality of their services.

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