Design and Analysis of a Prioritized Adaptive Multiple Access Scheme for VoIP over WLANs

Bing Feng*, Zhen Wang*, Chi Zhang*, Yuguang Fang†
*Key Laboratory of Electromagnetic Space Information, Chinese Academy of Sciences
University of Science and Technology of China
†Department of Electrical and Computer Engineering
University of Florida, Gainesville, USA
Email: {fengice, wang1992}@mail.ustc.edu.cn, chizhang@ustc.edu.cn, fang@ece.ufl.edu

***Abstract*—Voice capacity over wireless local area networks (WLANs) can be increased by the statistical multiplexing among on/off voice calls. However, in previously proposed schemes, the admitted voice calls that transit from silence state to talkspurt state are mixed up with the new voice calls to contend for the channel. The increase in the traffic load of new voice users could degrade the performance of ongoing voice calls. In this paper, we propose a novel MAC scheme for VoIP over WLANs, referred to as PAMA (Prioritized Adaptive Multiple Access). The key features of the proposed scheme are that 1) the admitted voice calls have higher priority access to the channel than the new voice calls, based on the fact that maintaining the required QoS of ongoing calls is more important than admitting new calls; 2) the dedicated contention window for admitted voice calls is dynamically adjusted according to the current estimation of the number of active ongoing voice calls to guarantee the QoS requirements; 3) a two-state Markov model is established to evaluate the system performance. Analytical and simulation results demonstrate that PAMA can increase the voice capacity while still satisfying the QoS of admitted voice calls.**

I. Introduction

Voice over Internet Protocol (VoIP) is an increasingly popular service due to its low cost. Meanwhile, WLANs have been widely deployed to provide cheap wireless access capability. These two technologies have driven the important and promising application of Voice over WLAN (VoWLAN) such as smartphones running voice applications. However, how to improve the VoIP capacity over WLANs while satisfying the Quality of Service (QoS) for voice users is still an open issue and a challenging work.

The legacy 802.11 Medium Access Control (MAC) standard [1] including Point Coordination Function (PCF) and Distributed Coordinate Function (DCF) is originally designed for best-effort data traffic and lacks built-in mechanisms for providing QoS for delay-sensitive services. Both analysis [2] and experiment [3] results have shown that given the QoS requirements such as voice packet loss probability, the VoIP capacity over 802.11 is very limited. To providing QoS in 802.11, the new 802.11e standard [4] proposes Hybrid Coordination Function (HCF) that contains HCF Controlled Channel Access (HCCA) and Enhanced Distributed Channel Access (EDCA). However, the contention-based EDCA only provides statistical priority so that it can not guarantee the strict QoS [5]. On the other hand, the polling-based HCCA does provide guaranteed QoS for real-time services such as VoIP and has received more attentions. However, how to improve the VoIP capacity in HCCA still needs to be tackled.

Although there are have been several papers that improve the VoIP capacity over WLANs by introducing new schemes such as queue aggregation scheme [6] and call admission control strategy [7], but the multiplexing scheme [8]–[10] that statistically multiplex on/off voice users is the most important method to significantly improve the VoIP capacity. In normal conversation, when one is talking, the other one is silent. Thus, human speech consists of silence (off) periods and talkspurt (on) periods, which allows for silence suppression to provide multiplexing capacity gain. The conventional speech detectors used in application layer can detect the silence-to-talkspurt and talkspurt-to-silence state changes. However, in HCCA, once a voice user is added to the polling list, the Access Point (AP) will poll it in subsequent cycles until the whole call ends. Thus, when voice users are in silence periods with no voice packets to transmit, it exists null-polling overhead that decreases the channel efficiency.

Various schemes [8]–[10] have been proposed in the literature to exploit the silence periods of conversation speech to improve channel efficiency to enhance the VoIP capacity in polling-based schemes. disconnecting voice users from the channel during the silence periods is easy, but how to ensure the successful access whenever the silence voice users begin talkspurts is a challenging work due to the unpredictable state transition. Most proposed schemes adopt contention schemes. When silence to talkspurt state change occurs, the first voice packet of a talkspurt needs to contend for the channel. Once the transmission is successful, the voice user is added to the polling list for collision-free transmissions of the rest of the whole talkspurt. However, previously proposed schemes mainly focus on how to provide higher priority access to voice users than best-effort data traffic. The admitted voice

This work was supported in part by the Natural Science Foundation of China (NSFC) under Grants 61202140 and 61328208, by the Fundamental Research Funds for the Central Universities of China, by the Innovation Foundation of the Chinese Academy of Sciences under Grand CXJJ-14-S132, and by the Program for New Century Excellent Talents in University of China under Grant NCET-13-0548. The work of Y. Fang was also partially supported by the US National Science Foundation under grants CNS-1409797 and CNS-1343356.

978-1-4799-5952-5/15/$31.00 ©2015 IEEE

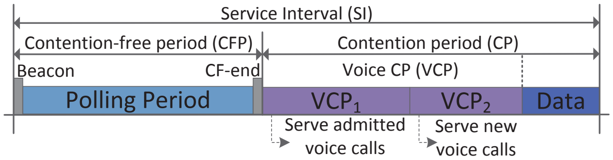

Fig. 1. The structure of a service interval in PAMA.

users are not differentiated from new voice users, and both of them have the same priority access to the channel. In practical networks, when the traffic load of new voice users increases, the successful access probability for admitted voice users decreases so that the specified requirement of access delay is not met. The delayed voice packets will be discarded, which degrades the QoS of voice users and even leads to voice users quitting the ongoing voice calls.

In this paper, based on the fact that, from the voice user's point of view, the service of a ongoing voice call is more important, as degrading QoS in the middle of ongoing voice calls is more annoying than blocking of new voice calls, we propose the prioritized adaptive multiple access (PAMA) scheme. The PAMA completely separates the admitted and new voice calls, and a dynamic contention period is dedicated for admitted voice calls. The dedicated contention period is backoff contention window whose size is dynamically adjusted according to the current estimation of the number of contending admitted voice users so as to keep the contention failure probability for ongoing voice calls below a certain threshold.

The rest of this paper is organized as follows. Section II describes the proposed MAC scheme in details along with the voice source model. Section III provides the performance analysis. Section IV presents the analysis and simulation results, and Section V concludes the paper.

II. PROTOCOL DESIGN OF PAMA

A. Basic Description of PAMA

The PAMA is a centralized channel access mechanism, and it is based on the structure of the IEEE 802.11e HCF. Time is divided into repeating service intervals by the beacon frames, and the service interval guarantees the maximal delay bound of all admitted voice calls. Each cycle begins with a beacon frame transmitted by the AP, and the remaining time includes a contention-free period (CFP), where the polling-based HCCA is used to provide collision-free transmissions, and a contention-period (CP), where contention-based schemes can be used. During the CFP, the AP polls every user in its polling list and only the polled user is allowed to transmit. The CFP ends with a CF-end frame transmitted by the AP. The proposed PAMA uses the same polling scheme as in IEEE 802.11e HCF during CFP. PAMA uses a novel CP structure while the EDCA is used in IEEE 802.11e HCF. This paper focuses on two traffic types, namely, voice and data, and the voice services have higher priority.

As shown in Fig. 1, during the CP, to guarantee strict priority, the voice contention period (VCP) is served ahead of best-effort data period. The VCP is further divided into two periods. The first period (VCP_1) is used to serve admitted voice calls while the second period (VCP_2) is used to serve new voice calls. Thus the admitted voice calls are protected from new voice calls that may degrade the performance of existing voice calls. Note that the existence of VCP_1 is due to the admissions of voice calls. In other words, VCP only contains the VCP_2 initially. As in 802.11e, the initial contention window size for VCP_2 is $CW_2[0] = 3$. It is emphasized that the voice contention periods VCP_1 and VCP_2 are backoff contention window from which voice users select backoff counters, and both are variable in every cycle. The dynamic adjustment of VCP_2 for new voice users is based on our previous work proposed in [11]. In WLANs, all nodes communicate with the AP, so the AP can know the state of transmissions during VCP_2. Based on c_c, the recorded number of *collision backoff counters* that selected by multi nodes, the AP dynamically adjusts the contention window size of VCP_2 in every cycle.

In the basic structure of SI, the broadcast CF-end frame is used to announce the end of CFP and initiate the CP. The CF-end frame piggybacks contention window sizes CW_1 corresponding to VCP_1 and CW_2 corresponding to VCP_2. The total contention window size (CW_v) for the voice CP is $CW_v = CW_1 + CW_2$. When newly arriving voice users enter the network or the admitted voice users need to contend for the channel due to the silence-to-talkspurt state changes, they should wait for the arrival of the broadcast CF-end frame to get the latest CW_1 and CW_2 to contend for the channel with backoff counters randomly selected within $[0, CW_1 - 1]$ and $[CW_1, CW_1 + CW_2 - 1]$, respectively. The leftover bandwidth are used by data traffic. Note that after receiving the CF-end frame, data users need to first set their backoff counters to CW_v to defer their access to the channel. Then when the backoff counters of data users reach zero, the data users enter the data contention period left by voice users and employ the legacy DCF to contend for the channel. To protect the fixed SI as in 802.11e, a data user does not transmit if its transmission cannot be finished before the arrival of the next beacon frame.

In PAMA, during VCP, each voice user employs the standard CSMA/CA scheme and executes the backoff procedure with a random backoff counter. PAMA does not adopt the binary exponential backoff scheme used in legacy 802.11 for voice users during the VCP. After its contention transmission (success or collision), every voice user defers its next contention until the arrival of the next broadcast CF-end frame. It is worth noting that the admission control scheme is achieved by VCP_2. The size of contention window of VCP_2 can be set to zero when there are no available resources for new voice calls. Thus, all new voice calls will defer their accesses to the channel until the size of VCP_2 is no longer zero.

To distinguish the end of a entire call from the end of a current talkspurt, two indicators are included in the MAC header of frames. The 1-bit queue indicator (QI) is used to notify the AP of the talkspurt-to-silence state change, then when $QI = 1$, the AP removes the voice user from the polling list. The 1-bit call indicator (CI) is used to notify the AP of the end of the whole call, and the AP keeps a admission counter (AC) to record the number of admitted voice users in the network. When $CI = 1$, the AP decreases the value AC by one. When a new voice users is admitted, the AC is increased by one.

B. Dynamic Adjustment of Contention Window for Admitted Voice Calls

In PAMA, to statistically multiplex the on/off voice calls, when a talkspurt ends, the reservation is released. Then when a new talkspurt begins, the successful contention transmission of the first voice packet of the new talkspurt acts as a reservation. If the voice packets generated at the beginning of a talkspurt can not be transmitted successfully within bounded delay, the delayed voice packets are dropped. The voice quality degrades with increased packet dropping. Thus, the voice contention period for admitted voice calls should be dynamically adjusted to keep the packet loss probability below a threshold.

During a conversation speech, the two-state Markov model presented in [12] is used to describe the voice activity. In the voice traffic model, the silence (off) periods alternate with the talkspurt (on) periods, and both of them follow exponential distributions. The mean length of a talkspurt period and of a silence period are denoted by t_1 and t_2, respectively. When a voice user is in talkspurt state, the probability that a transition to silence state happens after duration T is $\beta = 1 - e^{-T/t_1}$. Correspondingly, the probability that a voice user in silence state makes a transition to the talkspurt state after duration T is $\alpha = 1 - e^{-T/t_2}$.

A admitted voice user is always in one of three states [13], namely, the silence state, the contention state, and the reservation state. The state transition of each admitted voice users is presented as follows. A voice user is in silence state if it is in silence periods. When the voice user generates a new talkspurt and has not successfully contend for the channel, it passes into the contention state (transition $T_{s,c}$). When it successfully accesses the channel and is added to the polling list, this voice user enters the reservation state (transition $T_{c,r}$). If the reservation is unsuccessful before a talkspurt ends, it remains in the contention state (transition $T_{c,c}$). If a talkspurt ends before a reservation is made successfully, the voice user returns to silence state (transition $T_{c,s}$). When a voice user in reservation state completes a talkspurt, it returns to silence state (transition $T_{r,s}$). Otherwise, the voice user remains in reservation state to transmit voice packets (transition $T_{r,r}$).

The voice system can be fully described by the following state variables. At the beginning of cycle k, let N_k be the number of admitted voice users in the network, and let S_k, C_k, and R_k be the number of admitted voice users in silence state, contention state, and reservation state, respectively. Clearly,

$$N_k = S_k + C_k + R_k. \tag{1}$$

Then, at the end of cycle k and just before the beginning of the following cycle $k+1$, we have

$$N_{k+1} = N_k + A_k - D_k \tag{2}$$

where A_k is the number of newly admitted voice users during the cycle k, and D_k is the number of voice users that end their whole conversations and leave the network during the cycle k. And we have

$$\begin{aligned} R_{k+1} &= R_k + N_{c,r}^k - N_{r,s}^k \\ C_{k+1} &= C_k - N_{c,r}^k + N_{s,c}^k - N_{c,s}^k \end{aligned} \tag{3}$$

where $N_{c,r}^k$, $N_{r,s}^k$, $N_{s,c}^k$, and $N_{c,s}^k$ denote the number of admitted voice users that correspond to the transitions $T_{c,r}$, $T_{r,s}$, $T_{s,c}$, and $T_{c,s}$ during the cycle k, respectively.

Let $\hat{C}_{k+1}$ be the estimated number of contending voice users that will contend for the channel during VCP$_1$ in cycle $k+1$. Based on the estimated values C_{k+1} and S_{k+1}, the AP needs to estimate the the values $N_{s,c}^{k+1}$ and $N_{c,s}^{k+1}$. Then $\hat{C}_{k+1} = C_{k+1} + N_{s,c}^{k+1} - N_{c,s}^{k+1}$. Based on the voice model, when S_{k+1} voice users are in silence state and C_{k+1} voice users are in contention state at beginning of cycle $k+1$, the estimated $N_{s,c}^{k+1}$ and $N_{c,s}^{k+1}$ are $S_{k+1} \cdot \beta$ and $C_{k+1} \cdot \alpha$, respectively. Thus, the AP can get the estimated value of $\hat{C}_{k+1}$.

During the cycle k, we define the contention failure probability P_f as the ratio of the number of failed transmissions to the total number of contending admitted voice users, i.e.,

$$P_f = \frac{\hat{C}_k - CW_1[k] \cdot P_{suc}}{\hat{C}_k} \tag{4}$$

where $\hat{C}_k$ is the number of contending voice users estimated by the AP, and the $CW_1[k]$ is the corresponding target contention window size for VCP$_1$ during cycle k. The P_{suc} is the probability that a backoff counter is selected by exactly one contending admitted voice user, which is distributed as binomial, i.e.,

$$P_{suc} = \binom{\hat{C}_k}{1} \left(\frac{1}{CW_1[k]}\right) \left(1 - \frac{1}{CW_1[k]}\right)^{\hat{C}_k - 1}. \tag{5}$$

To guarantee the quality of admitted voice calls, the P_f should satisfy $P_f = P_{tra}$, where P_{tra} is the target value. Then we get the approximate value of $CW_1[k]$ for VCP$_1$

$$CW_1[k] = \left\lceil \frac{1}{1 - e^{\frac{\ln(1-P_{tra})}{\hat{C}_k - 1}}} \right\rceil. \tag{6}$$

The latest contention window size for VCP$_1$ is broadcasted by the CF-end frame in every cycle.

III. PERFORMANCE ANALYSIS

A. Analytical Model

For admitted voice system with N_v admitted voice calls, it is described by a discrete Markov chain by observing the system states at the beginning of every cycle. We develop the analytic model proposed in [13], [14] to evaluate the performance of the proposed scheme. Let $X_i = (R_i, C_i)$ be the system state in cycle i. Note that $S_i = N_v - R_i - C_i$. The two-dimensional Markov process $\{X_i\}$ has a finite number of states and the stationary state distribution of system exists. When in steady-state X_i, there are R_i voice users in the polling list, C_i

contending voice users, and S_i silent voice users. Let π_i be the probability that the system is in steady-state X_i. Then let $\boldsymbol{\pi} = [\pi_0, \pi_1, ..., \pi_m]$ be the stationary probability vector. The state space of this Markov process is

$$\Omega = \{R_i, C_i | 0 \le R_i, C_i \le N_v, and \quad R_i + C_i \le N_v\}, \quad (7)$$

and the value of m, the dimension of Ω, can be get by $m = \sum_{l=0}^{N_v} N_v - l + 1 = \frac{1}{2}(N_v+2)(N_v+1)$.

Let $p_{i,j}$ be the transition probability from state X_i to state X_j, and we get the transition probability matrix $\boldsymbol{P} = [p_{i,j}]_{m \times m}$. Based on the theory of Markov chain, we have the matrix equation $\boldsymbol{\pi} = \boldsymbol{\pi P}$ and the normalization equation $\sum_{l=0}^{m-1} \pi_l = 1$.

Then we can obtain $\boldsymbol{\pi}$, and we calculate the expected values of system variables R and C as follows:

$$\begin{aligned} E[R] &= \sum_{k=0}^{N_v} k \cdot p_R(k) \\ E[C] &= \sum_{k=0}^{N_v} k \cdot p_C(k) \end{aligned} \quad (8)$$

by the stationary distributions:

$$\begin{aligned} p_R(k) &= Pr\{R = k\} = \sum_{(r_l, c_l) \in \Omega, r_l = k} \pi_l \\ p_C(k) &= Pr\{C = k\} = \sum_{(r_l, c_l) \in \Omega, c_l = k} \pi_l \end{aligned} \quad (9)$$

However, before the above calculation, we must construct the matrix $\boldsymbol{P}$. According to (3), we can get the $p_{i,j}$ from the distributions of $N_{r,s}$, $N_{c,s}$, $N_{s,c}$, and $N_{c,r}$. We rewrite $p_{i,j}$ as

$$\begin{aligned} p_{i,j} =& Pr\{R_j = r_j, C_j = c_j | R_i = r_i, C_i = c_i\} \\ =& \sum_{u=0}^{c_i} \sum_{v=0}^{N_v - r_i - c_i} Pr\{N_{c,r} = u\} \cdot Pr\{N_{s,c} = v\} \\ & \cdot Pr\{N_{c,s} = c_i - u + v - c_j\} \\ & \cdot Pr\{N_{r,s} = r_i + u - r_j\}. \end{aligned} \quad (10)$$

To simplify the notation, we define the binomial distribution probability as

$$B(X, x, p) = \binom{X}{x} (p)^x (1-p)^{X-x} \quad (11)$$

and the probability that exactly r voice users among N randomly select a backoff counter that no other voice users select for, is

$$\Gamma_M^N(r) = \sum_{z=r}^{N} (-1)^{z-r} \binom{N}{z} \binom{z}{r} \cdot \frac{(M)!(M-z)^{N-z}}{(M-z)!(M)^N}. \quad (12)$$

Based on the voice traffic model, we obtain the expressions for the different terms in (10). When $N_v - r_i - c_i$ voice users are in silence state, the probability that v silence voice users begin talkspurts is

$$Pr\{N_{s,c} = v\} = B(N_v - r_i - c_i, v, \alpha) \quad (13)$$

and the probability that $c_i - u + v - c_j$ contending voice users among $c_i - u$ return to silence state is

$$\begin{aligned} & Pr\{N_{c,s} = c_i - u + v - c_j\} \\ & \qquad = B(c_i - u, c_i - u + v - c_j, \beta), \end{aligned} \quad (14)$$

as well as the probability that $r_i + u - r_j$ polled voice users among r_i return to silence state is

$$Pr\{N_{r,s} = r_i + u - r_j\} = B(r_i, r_i + u - r_j, \alpha). \quad (15)$$

For $Pr\{N_{c,r} = u\}$, it means c_i voice users randomly select backoff counters from $CW_1[i]$ which satisfies the equation (6) backoff counters, and yield u *success backoff counters* during VCP_1. Then

$$Pr\{N_{c,r} = u\} = \Gamma_{CW_1[i]}^{c_i}(u). \quad (16)$$

B. System Performance

Based on the above results and given the length of service interval T_{SI}, we will derive the maximum number of voice calls that can be supported in proposed PAMA scheme.

Let $\overline{T}_{\text{ad-voice}}$ be the average length of service time for admitted voice calls. Based on the $E[R]$ and $E[C]$, we have the following result:

$$\begin{aligned} \overline{T}_{\text{ad-voice}} = & PIFS + T_B + E[R] \cdot \overline{T}_t + PIFS \\ & + T_{CF-end} + \overline{T}_{\text{VCP}_1} \end{aligned} \quad (17)$$

where T_B and T_{CF-end} are the average duration of a beacon frame and a CF-end frame, respectively. The average duration of a polling transmission is

$$\overline{T}_t = PIFS + T_{polling} + SIFS + H + E[P] + SIFS + ACK \quad (18)$$

where $T_{polling}$ is average length of a polling frame, H is the length of MAC header and PHY header, and $E[P]$ is the average length of a voice packet payload. The average duration for VCP_1 is

$$\overline{T}_{\text{VCP}_1} = CW_1[i] \cdot (P_s \cdot T_{succ} + P_c \cdot T_{coll} + P_i \cdot \sigma) \quad (19)$$

where σ denotes the duration of an idle time slot. T_{succ} and T_{coll} are the average duration of a successful transmission and a packet collision during VCP_1, respectively. Let δ denote the air propagation delay that can be set to 1 us, then we have

$$\begin{aligned} T_{succ} &= DIFS + H + E[P] + \delta + SIFS + ACK + \delta \\ T_{coll} &= H + E[P] + DIFS + \delta. \end{aligned} \quad (20)$$

According to the equation (11), the idle probability, success probability and collision probability are given by

$$\begin{aligned} P_i &= B(E[C], 0, \tau) \\ P_s &= B(E[C], 1, \tau) \\ P_c &= 1 - P_s - P_i \end{aligned} \quad (21)$$

The $\tau = \frac{1}{CW_1[i]}$ is the probability that a contending voice user randomly select a backoff counter during VCP_1.

As the duration of $\overline{T}_{\text{ad-voice}}$ is bounded by T_{SI}, the maximum number of admitted voice users can be get while the equality $\overline{T}_{\text{ad-voice}} = T_{SI}$ is satisfied.

TABLE I
SYSTEM PARAMETERS

Parameters	Value	Parameters	Value
SIFS	10 μs	Idle slot	20 μs
DIFS	50 μs	PIFS	30 μs
PHY header	192 μs	MAC header	34 bytes
Polling frame	36 bytes	ACK	14 bytes
CF-end frame	20 bytes	Beacon frame	40 bytes
Data frame rate	11 Mbps	Service interval	80 ms

IV. PERFORMANCE EVALUATION

In this section, we first give a numerical example to testify the above analysis based on a set of system parameters. Then simulation results obtained by OPNET (version 14.5) are used to evaluate the performance of PAMA and verify the accuracy of our analytical results. Table I summarizes the system parameters. The voice is coded with G.711 that is the international standard for encoding telephone audio [7]. During talkspurt periods, G.711 has fixed bit rate of 64 kb/s. For different sample periods T_{sp}, the voice payload is different. The $T_{\text{sp}} = 10$ ms corresponds to a rate of 100 packet/s, and the voice payload size is $64000/(100*8) = 80$ bytes. When T_{sp} is 20 ms, the voice payload size is 160 bytes. For the two-state voice traffic model used in our performance evaluation, the mean duration of talkspurt and silence both follow exponential distribution with values of 1 s and 1.35 s, respectively.

A. Numerical Results

Under the provided system parameters, Fig. 2 shows the analytical result of the maximum number of VoIP that can be supported in PAMA. We also show the results for G.711 voice codec with different sample periods T_{sp}. For a comparison, the results of HCCA with multiplexing scheme (i.e., HCCA+EDCF) and HCCA scheme are presented. From Fig. 2, we can see that PAMA outperforms the other schemes by significantly increasing the number of admitted uplink voice users. Based on the comparison between HCCA+EDCF and HCCA, we can see the effect of statistical multiplexing on the performance improvement. Based on the comparison between PAMA and HCCA+EDCF, we can see the effect of the proposed adaptive voice contention window on the performance improvement. In addition, from Fig. 2, we can also observe that different sample periods T_{sp} have different values of VoIP capacity, and when the T_{sp} is enlarged, more voice users can be admitted.

To see how the proposed dynamic contention window for VCP_1 works under different numbers of contending voice users in the network, the G.711 with 10 ms and 20 ms sample periods are used to illustrate. From Fig. 3, It is clear that to keep the contention failure probability P_f satisfy the target value, the contention window size for VCP_1 should be adaptively adjusted when the number of contending voice users increases. We also find that given large P_f, the contention window size increases more sharply than that of small P_f.

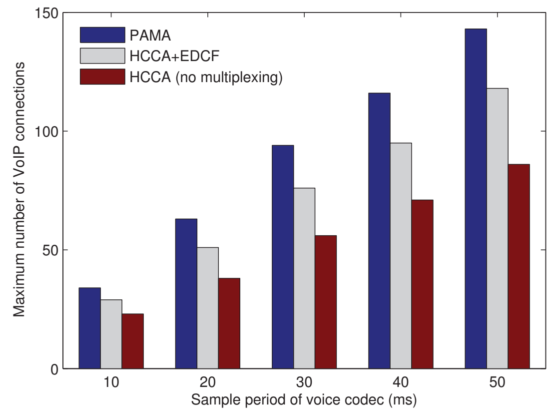

Fig. 2. The maximum number of supported VoIP in the network.

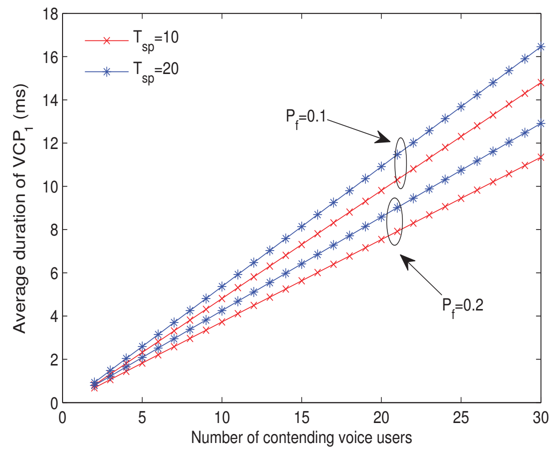

Fig. 3. Average duration required to serve contending voice users during VCP_1.

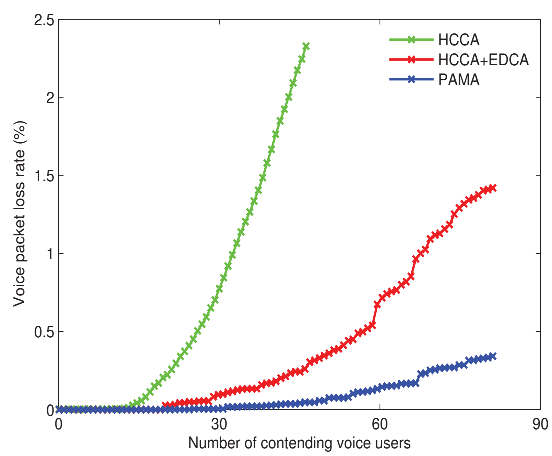

Fig. 4. Voice packet lose rate versus the number of contending voice users.

B. Simulation Results

The simulation result shown in Fig. 4 focus on the voice packet loss rate. As the result imply, the voice packet lose rate increases sharply when the number of contending voice user exceeds the maximum number of voice users that can be supported in the network. Based on the simulation results, Fig. 4 verifies the accuracy of our analytical results. Due to

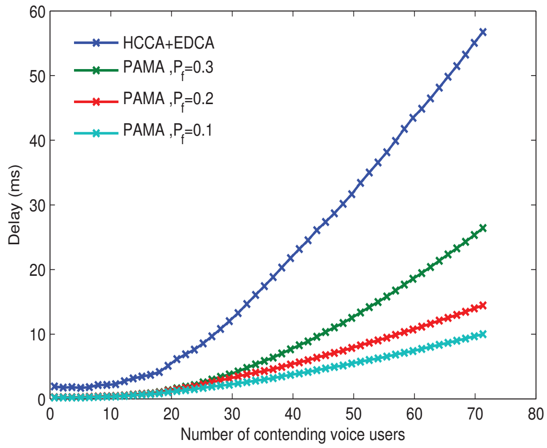

Fig. 5. Delay of voice packetd versus the number of contending voice users.

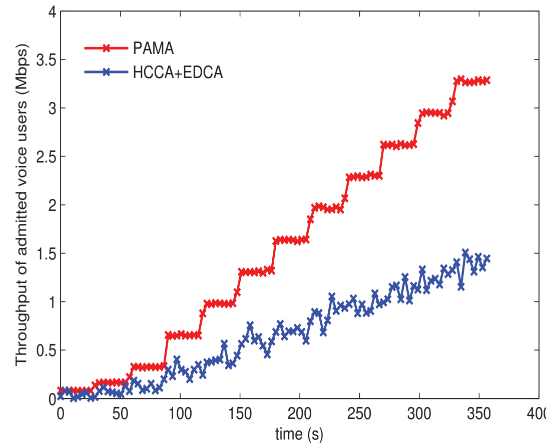

Fig. 6. Throughput of admitted voice users versus the simulation time.

both the multiplexing scheme and adaptive contention window scheme, the proposed scheme PAMA is capable of admitting more voice calls while satisfying the QoS requirements. In addition, Fig. 5 also shows the simulation result of the access delay of the first voice packet of a talkspurt during VCP$_1$. The HCCA+EDCF scheme cannot guarantee the determinist priority access to voice users, the contending voice users have to encounter access delay. On the other hand, the PAMA scheme dynamically adjusts the contention window size to keep the contention failure probability P_f that is related to the QoS requirements below a threshold. Note that the larger P_f means that the VoIP can tolerate larger access delay. If the value of P_f is too large, the QoS of voice users cannot be guaranteed. Thus, it is necessary to set a appropriate value of P_f.

We have a new simulation where there are 10 best-effort data users still in the network, and for every 30 seconds, we add 3 new voice users into the network. The simulation time is 400 seconds. As shown in Fig. 6, in PAMA, the admitted voice users are completely separated from other services, so the throughput of admitted voice users increases steadily and all new voice users can be admitted. However, in HCCA+EDCF scheme, the admitted voice users are not been protected from other services, the throughput of admitted voice users increases slowly and the fluctuation is large.

V. Conclusion

The proposed MAC scheme PAMA provides substantial multiplexing capacity gain by exploiting the on/off characteristics of conversational speech. Based on the fact that, from the voice users perspective, degrading the QoS of ongoing voice calls is generally not considered acceptable or user-friendly, the proposed scheme differentiates ongoing voice calls from newly arriving voice calls, and gives higher priority to the ongoing voice users. In PAMA, the hardest part of exploiting silence suppression in wireless LANs is the detection of the silence-to-talkspurt state change. When silent voice users begin talkspurts, the first voice packet of a talkspurt needs to contend for the channel during the dedicated contention period that is adaptively adjusted on the basis of an estimation of the number of contending voice users. It is noted that the dedicated contention period is contention backoff window from which ongoing voice users select backoff counters. We have developed a two-state Markov model to analyze the performance of the PAMA voice system.

References

[1] Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specification, IEEE Std. 802.11, 1997.

[2] W. Wang, S. C. Liew, and V. O. Li, Solutions to performance problems in voip over a 802.11 wireless lan, *IEEE Trans. Veh. Technol.*, vol. 54, no. 1, pp. 366-384, 2005.

[3] S. Shin and H. Schulzrinne, Experimental measurement of the capacity for voip traffic in ieee 802.11 wlans, in *Proc. IEEE International Conference on Communications (INFOCOM07)*, 2007, pp. 2018-2026.

[4] Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications: Amendment 8: Medium Access Control (MAC) Quality of Service Enhancements, IEEE Std. 802.11e, 2005.

[5] K. O. Stoeckigt and H. L. Vu, Voip capacity analysis, improvements, and limits in ieee 802.11 wireless lan, *IEEE Trans. Veh. Technol.*, vol. 59, no. 9, pp. 4553-4563, 2010.

[6] L. Yang, Z. Chao, L. Jianhua, and L. Xiaokang, A mac queue aggregation scheme for voip transmission in wlan, in *Proc. IEEE Wireless Communications and Networking Conference (WCNC07)*, pp. 2121-2125.

[7] P. McGovern, P. Perry, S. Murphy, and L. Murphy, Endpoint-based call admission control and resource management for vowlan, *IEEE Trans. Mobile Comput.*, vol. 10, no. 5, pp. 684-699, 2011.

[8] R. Y. Lam, V. C. Leung, and H. C. Chan, Polling-based protocols for packet voice transport over ieee 802.11 wireless local area networks, *Wireless Communications, IEEE*, vol. 13, no. 1, pp. 22-29, 2006.

[9] P. Wang, H. Jiang, and W. Zhuang, Capacity improvement and analysis for voice/data traffic over wlans, *IEEE Trans. Wireless Commun.*, vol. 6, no. 4, pp. 1530-1541, 2007.

[10] I. A. Qaimkhani and E. Hossain, Efficient silence suppression and call admission control through contention-free medium access for voip in wifi networks, *Communications Magazine, IEEE*, vol. 46, no. 1, pp. 90-99, 2008.

[11] B. Feng, C. Zhang, B. Liu, and Y. Fang, Sequentially ordered backoff: towards implicit resource reservation for wireless lans, in *Proc. IEEE International Conference on Communications (ICC15)*, 2015.

[12] D. J. Goodman and S. X. Wei, Efficiency of packet reservation multiple access, *IEEE Trans. Veh. Technol.*, vol. 40, no. 1, pp. 170-176, 1991.

[13] G. Wu, K. Mukumoto, and A. Fukuda, Analysis of an integrated voice and data transmission system using packet reservation multiple access, *IEEE Trans. Veh. Technol.*, vol. 43, no. 2, pp. 289-297, 1994.

[14] S. Jiang, J. Rao, D. He, X. Ling, and C. C. Ko, A simple distributed prma for manets, *IEEE Trans. Veh. Technol.*, vol. 51, no. 2, pp. 293-305, 2002.