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VoIP Capacity Analysis in IEEE 802.11 WLAN

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***Abstract*—Wireless voice over IP (VoIP) is an important emerging service in telecommunications due to its potential for replacing cell phone communication wherever wireless local area network (WLAN) is installed. Recent studies, however, suggest that the number of voice calls that can be supported in the widely deployed IEEE 802.11 WLAN is limited. In this paper, we utilize a so-called transmission opportunity (TXOP) parameter of a medium access control protocol as a simple solution to improve the VoIP capacity. We provide a detailed analytical model to show that the capacity can be improved significantly, and discuss the implications of the TXOP parameter in terms of the maximum number of calls the 802.11 network can support. The analytical results are validated by simulations for a wide range of parameters. Furthermore we investigate the impact of the buffer at the access point (AP) on the number of obtainable voice calls. We show that there exists an optimal buffer size where the maximum voice capacity is achieved, but further increasing the buffer beyond this value will not result in an increased voice capacity. Based on this finding a closed form expression for the maximum number of voice calls is developed as a function of TXOP value. Finally, we propose a simple yet accurate voice capacity approximation formula for voice capacity estimation in WLAN and provide some insights that an be gained from this formula.**

I. Introduction

In recent years IEEE 802.11 wireless local area network (WLAN) has become more readily available to the public. Many places such as cafes, restaurants, airport lounges now offer free wireless access. Decreasing costs for wireless equipments, and subsequently the integration of 802.11 technologies in mobile devices, such as Laptops, Pocket PCs and cell phones, drive an increased demand for wireless access. As wireless becomes more accessible, wireless voice over IP (VoIP) is an important emerging service due to low cost and its potential for replacing cell phone communication wherever WLAN is installed. However, recent studies [1], [2], [3] suggest that the number of voice calls that can be supported in a WLAN is limited. For example, only 5 to 7 calls are supported using a G.729 voice codec with a 10 ms sampling rate [2], [4].

While the IEEE 802.11 standard was originally designed to support best effort services in WLAN, a new IEEE 802.11e standard was ratified in 2005 to meet the growing volumes of real-time traffic (such as voice traffic) that requires some degree of quality of service (QoS) [5]. It extends the access mechanism in the medium access control (MAC) protocol of the IEEE 802.11 standard by allowing adjustment of a number of MAC parameters that were previously fixed.

There have been several papers in the literature that investigate the voice capacity of a WLAN based on both the IEEE 802.11 and 802.11e protocols. In particular, the limited capacity for VoIP over WLAN is discussed by Cai et al. [2], where the authors provide an analytical model to show that the access point is a bottleneck in the 802.11a/b WLAN. A slightly different method is used by Hedge et al. [3], who provide a capacity analysis based on a network delay. In [3] the authors also extend their model to include IEEE 802.11e focusing on voice throughput in the presence of background (TCP) traffic. Wang et al. [6] propose a multiplexing scheme to ease the downlink traffic from the access point to the wireless VoIP nodes. However, this requires changes to the protocol and network infrastructure. To this end, it requires multiplexing and demultiplexing entities and the wireless network to be multicast enabled, which may not be desirable. Dangerfield et al. [7] use a so-called transmission opportunity (TXOP) parameter defined in the 802.11e standard to improve the voice capacity and show that significant improvements can be made based on real measurements over a WLAN test-bed. This solution does not require any additional equipments, or changes to the 802.11 MAC protocol.

In this paper, we develop a queueing model to analytically evaluate the performance gain that can be obtained using the above TXOP parameter. We show that the TXOP solution can improve the voice capacity in WLAN significantly. We also show that the same improvement can be achieved as reported in [7], but with a smaller TXOP value. Furthermore we study the impact of the buffer size at the AP on the number of voice calls that the WLAN can support. We validate our analytical results using the NS-2 simulation [8] for different voice codecs and TXOP values. In addition, we investigate the impact of the AP buffer size on the maximum number of voice calls. We will show that there exists a minimum buffer size with which the voice capacity is maximum. Based on this finding we develop a closed form expression for the voice capacity in an 802.11 WLAN. Using this closed form expression, we propose a simple yet accurate voice capacity approximation formula and provide some insights that can be gained from this formula.

The rest of the paper is organized as follows. In Section II, we describe a simple solution using TXOP parameter to improve the VoIP capacity in WLAN. We then provide in Section III the detailed queueing model to evaluate the improvements in the proposed solution. In Section IV we validate our queueing model by comparing analytical results with simulation and study the impact of the AP buffer size on the voice capacity. In Section V we propose a simple yet accurate voice capacity approximation formula for voice capacity estimation and provide some insights that an be gained from this formula. Finally, we conclude our paper in Section VI.

II. A Simple Solution to Improve VoIP Capacity

Consider a scenario where multiple voice calls are initiated simultaneously in an infrastructure WLAN network, as shown in Fig. 1. In this network, all the voice clients will communicate via an access point (AP). Thus voice traffic to and from any mobile node in that network must flow through the common AP acting as a base station. Since every station including the AP has the same chance to access the wireless channel, the probability of the AP winning a channel access is decreasing with an increasing number of wireless nodes that maintain a voice call. It is because the AP has to compete against all the wireless nodes to access the channel for every packet of the downlink stream. As the probability of the AP winning channel access decreases, the AP becomes a bottleneck and packets from the downlink streams start to build up at the AP. If the number of voice calls keeps increasing, at some stage, the AP will start to drop packets and the quality of the voice call starts to degrade. Here we define κ as a packet loss threshold, over which the satisfactory user-perceived quality for a call can not be maintained.

Similar to [7], in this paper we will give preference to the AP when competing for channel access by setting a larger $TXOPLimit$ at the MAC layer. The $TXOPLimit$ is the maximum duration during which the TXOP holder maintains uninterrupted control of the medium after obtaining a transmission opportunity [5]. During the period of $TXOPLimit$, the TXOP holder can transmit multiple packets. To avoid contentions from other nodes during the time duration of $TXOPLimit$, the TXOP holder is allowed to commence its transmission of a consecutive packet after a short inter-frame space (SIFS) following the completion of the immediately preceding frame exchange sequence, i.e., on receipt of an acknowledge (ACK) frame. Figure 2 shows the simplified structure of packets transmission (TXOP-frame) from the AP using TXOP.

In the next section, we develop a queueing model to evaluate the effectiveness of this solution in terms of voice capacity in WLAN.

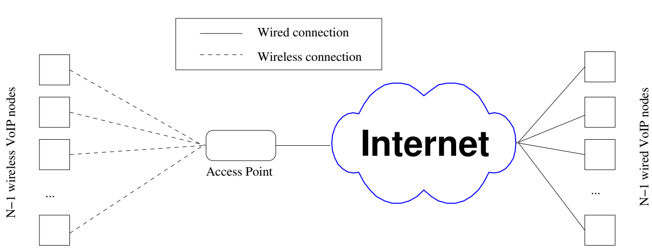

Fig. 1. VoIP network topology.

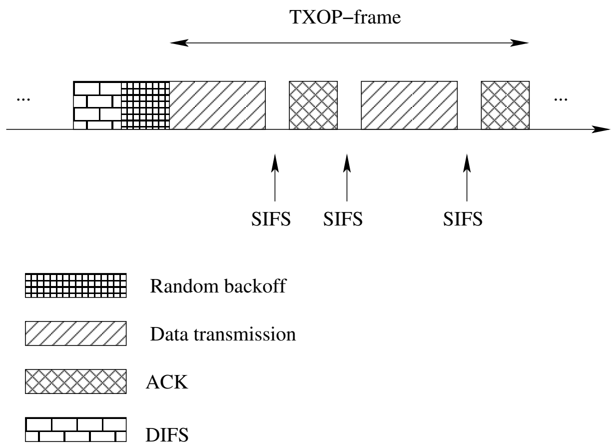

Fig. 2. TXOP-frame on the wireless channel as seen by the AP

III. Analytical Model

We consider an IEEE 802.11 infrastructure WLAN (Fig. 1) consisting of one access point (AP) and $N-1$ wireless nodes. Each wireless node maintains a full-duplex VoIP call to a node outside of the wireless network. As discussed earlier, all traffic from and to the wireless nodes have to traverse the access point. We assume basic access is used over an ideal channel without interference or hidden terminals, and channel access is controlled by the *distributed coordination function* (DCF). Note that in this paper we do not consider a subtle difference between the backoff process in DCF and the one defined in [5, Section 9.9.1.3]. Also, in this scenario we only consider voice traffic and no other traffic such as TCP traffic.

In the following indices a and n correspond to the access point and a wireless node, respectively. These indices will be omitted from the notations whenever we refer to both the access point and wireless nodes.

Let λ_n be the packet arrival rate of a wireless node in the network shown in Fig. 1. The arrival rate at the AP is a superposition of all the individual rate of voice traffic from $N-1$ wireless nodes and is given by $\lambda_a = (N-1)\lambda_n$. Denote the packet service rate of the AP and a wireless node by μ_a and μ_n, respectively. Here we use an $M/G/1/K$ queue to model the wireless node as well as the AP, where packets are served in batches of TXOP packets at the AP. Note that K is the number of packets that can be queued at a *station* (wireless node and/or AP) and can take different values at the wireless node and the AP. The queue utilization can be expressed as $\rho = \lambda/\mu$ where ρ is also the probability that a station has a packet to send. Thus a station will be idle with probability $1-\rho$.

The 802.11 protocol [9] specifies that a node has to wait a random period of time measured in backoff slots before attempting to transmit its packet. The backoff is uniformly and randomly selected from $[0, CW-1]$, where CW is the current contention window with the initial value of CW_{min}. Collision occurs if more than one station transmit in the same slot. If a collision occurs, the contention window of the sender is doubled unless the maximum value ($CW_{max} = 2^m CW_{min},\ m \geq 1$) has been reached, and the packet is scheduled for retransmission. The contention window is reset to CW_{min} when the packet has been successfully transmitted or discarded when the retransmission limit R ($R \geq m$) is reached. In this paper we assume constant and independent collision probability, denoted as c for a station. From the time the packet is ready to be sent until the packet is successfully transmitted, the average backoff experienced by a station can be approximated as

$$\overline{w} = \sum_{i=0}^{m} c^i(1-c)\sum_{k=0}^{i}\frac{2^k CW_{min}}{2}$$
$$+\sum_{l=m+1}^{R-1} c^l(1-c)\frac{\sum_{k=0}^{m} 2^k CW_{min} + (l-m)CW_{max}}{2}$$
$$+c^R\frac{\sum_{k=0}^{m} 2^k CW_{min} + (R-m)CW_{max}}{2}. \quad (1)$$

During the average backoff period $\overline{w}$, the average number of transmission attempts ϕ from a station is given by

$$\phi = (1-c) + ... + c^R(R+1) = \frac{1-c^{R+1}}{1-c}. \quad (2)$$

Given that a station has a packet to send, the conditional attempt probability τ that a station is attempting to transmit the packet in any given slot is

$$\tau = \phi/\overline{w}. \quad (3)$$

The collision probability seen by the AP, and a wireless node, however, can be expressed as a function of the above conditional attempt probability as follows.

$$c_a = 1-(1-\rho_n\tau_n)^{N-1}, \quad (4)$$
$$c_n = 1-(1-\rho_n\tau_n)^{N-2}(1-\rho_a\tau_a), \quad (5)$$

where $\rho\tau$ is the probability that a station is transmitting a packet in a slot. Note that (4) and (5) are based on the fact that the AP can collide with one of the $N-1$ wireless nodes, whereas a wireless node can only collide with the AP or one of the remaining $N-2$ wireless nodes.

Moreover the packet average service time of a wireless node can be decomposed into three parts: 1) the average backoff of the station, 2) the collision and successful transmission time of the packet itself, and 3) the interruptions to the backoff due to collision and successful transmission by the other wireless nodes and the AP. Here the successful transmission time of a packet is defined as

$$T_s = T_{DIFS} + T_p + T_{SIFS} + T_{ACK}, \quad (6)$$

where T_{DIFS} is the duration of the distributed inter-frame space, T_p is the time it takes to transmit the packet itself, T_{SIFS} is the duration of the short inter-frame space and T_{ACK} is the time to transmit the acknowledgement (including headers).

The collision time is given by

$$T_c = T_p + T_{ACK_{timeout}} + T_{DIFS}, \quad (7)$$

where $T_{ACK_{timeout}}$ is the timeout period of an unsuccessful transmission. Note that T_c is the actual collision time of each collision experienced by a station, and thus using c, R, and T_c, the average collision time of a station is given by

$$\bar{t} = \sum_{i=1}^{R} c^i(1-c)iT_c = \frac{c(1-(R+1)c^R + Rc^{R+1})}{1-c}\, T_c. \quad (8)$$

Let σ be the backoff slot duration, $T_{\overline{s}} = 2T_{SIFS} + T_p + T_{ACK}$, and $T_{\hat{s}} = T_s + (TXOP-1)T_{\overline{s}}$, then the average service time of a packet transmitted by a wireless node is given by

$$\frac{1}{\mu_n} = \overline{w}_n\sigma + \left(\frac{\bar{t}_n}{2} + T_s\right) +(N-2)\rho_n\left(\frac{\bar{t}_n}{2} + T_s\right) +\frac{(N-1)}{TXOP}\rho_n\left(\frac{\bar{t}_a}{2} + T_{\hat{s}}\right) \quad (9)$$

Because the AP is allowed to send up to $TXOP \geq 1$ consecutive packets, the average service time of the TXOP-frame consists of two parts: 1) the average service time of the first packet in the TXOP-frame ($1/\mu_{a1}$), and 2) the total average service time of all the subsequent packets in the TXOP-frame ($1/\mu_{a2}$). The service time of the first packet is calculated similar to (9), and depends on 1) the average random backoff of the AP, 2) the collision and transmission time of the packet itself, and 3) the interruptions to the backoff due to collisions and successful transmission of the wireless nodes. This average service time is given by

$$\frac{1}{\mu_{a1}} = \overline{w}_a\sigma + \left(\frac{\bar{t}_a}{2} + T_s\right) +(N-1)\frac{\lambda_n}{\mu_a}\left(\frac{\bar{t}_n}{2} + T_s\right). \quad (10)$$

Note that despite the AP has a higher $TXOPLimit$, the actual collision times are the same for the access point and the wireless nodes since a wireless node can only collide with the first packet of the TXOP-frame sent by the AP after it secures the channel access.

Any subsequent packet in the TXOP-frame has a service time of $T_{\overline{s}}$ only. The average service time of the remaining TXOP - 1 packets in the TXOP-frame is then given by

$$\frac{1}{\mu_{a2}} = (TXOP-1)\,T_{\overline{s}}, \quad (11)$$

and thus the average service time of a *packet* sent by the AP can then be calculated as follows.

$$\frac{1}{\mu_a} = \frac{1}{TXOP}\left(\frac{1}{\mu_{a1}} + \frac{1}{\mu_{a2}}\right). \quad (12)$$

Equations (3), (4), (5), (9) and (12) constitute a non-linear system of equations that can be solved iteratively to obtain the collision probabilities c and the conditional attempt probability τ, as well as ρ and μ.

Having obtained ρ and μ, we require the average packet loss κ of a voice call to be less than 2% to have an acceptable level of quality [10]. The maximum number of supported voice calls C is the number of calls such that the packet loss probability of a voice call p is maintained to be less than κ. Because the average packet loss seen by the AP is also the average packet loss of an individual call, and the buffer at the AP is the bottleneck, p can then be approximated by

$$p = \frac{(1-\rho_a)\rho_a^K}{1-\rho_a^{K+1}} \tag{13}$$
$$= \frac{\left(1-\frac{(N-1)\lambda_n}{\mu_a}\right)\left(\frac{(N-1)\lambda_n}{\mu_a}\right)^K}{1-\left(\frac{(N-1)\lambda_n}{\mu_a}\right)^{K+1}}.$$

To obtain C, we repeatedly solve the above non-linear system of equations with incremental number of voice calls. Note that (13) assumes exponential service time at the AP and thus it is only an approximation for the packet loss probability. We will investigate the accuracy of the analytical model using this approximation in the next section.

IV. Results and Discussion

In this section we validate our model by comparing the analytical results with simulation. Simulation is performed using NS-2 (version 2.28) with the EDCA extension from the TU Berlin [11]. Table I provides an overview of the parameters used in our analysis and simulation.

TABLE I
USED NETWORK PARAMETERS FOR AN IEEE 802.11 WIRELESS LAN

Channel rate	11MBit/s
Basic rate	1MBit/s
Backup Slot length σ	20 μs
SIFS	10 μs
DIFS	50 μs
CW_{min}	32
CW_{max}	1024
Max. backoff stage (m)	5
Retry limit (R)	7
Buffer size (K)	50 packets
Traffic/Data details	
PLCP & Preamble	192 μs
MAC Header + PCS	24.7 μs
RTP/UDP/IP Header	29.1 μs
Voice payload	7.27 μs (G.729, 10 ms) 58.18 μs (G.711, 10 ms)
ACK frame	112 μs

In the following for validation purposes we set the buffer size at the AP equal to 50 packets as indicated in Table I. The impact of different buffer values will be studied in more detail later on.

In Fig. 3 we show the packet loss probability p of the AP for G.729 and G.711 voice calls with a 10 ms sampling rate and selected values of TXOP parameter. It can be observed that for TXOP = 1 the network can accommodate up to 7 voice calls using G.729 codec before the packet loss threshold κ is exceeded. For TXOP = 5 the voice capacity is almost doubled with 12 voice calls using the same codec. Similar number of voice calls are reported in [7], but with a larger TXOP value where the authors set the TXOP value equal to the number of expected voice calls. In Figs. 4 and 5 we compare our analytical results with simulation using G.729 and G.711 codecs, respectively. Observe that the analytical results match those of the simulation closely. Note that in some cases due to the rounding error there is one call difference between the analytical results and results obtained from simulation.

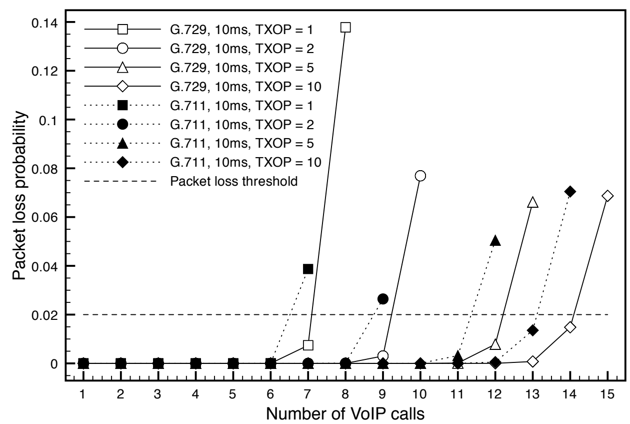

Fig. 3. Packet loss probability at the AP for G.729 and G.711 voice calls with a 10 ms sampling rate and selected values of TXOP, obtained using (13)

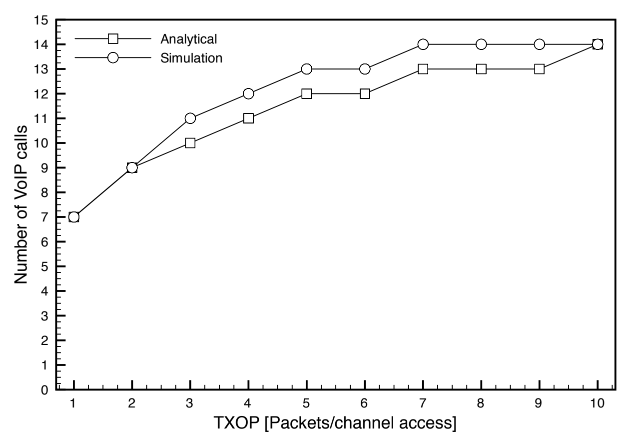

Fig. 4. Analytical and simulation results for G.729 voice calls with a 10 ms sampling rate

Even though setting larger TXOP parameter at the AP can improve significantly the voice capacity, the maximum number

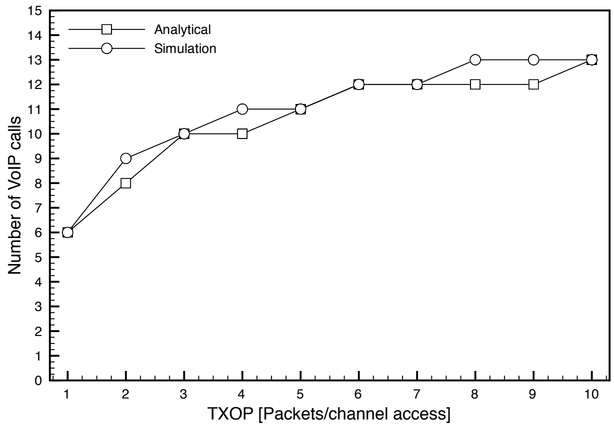

Fig. 5. Analytical and simulation results for G.711 voice calls with a 10 ms sampling rate

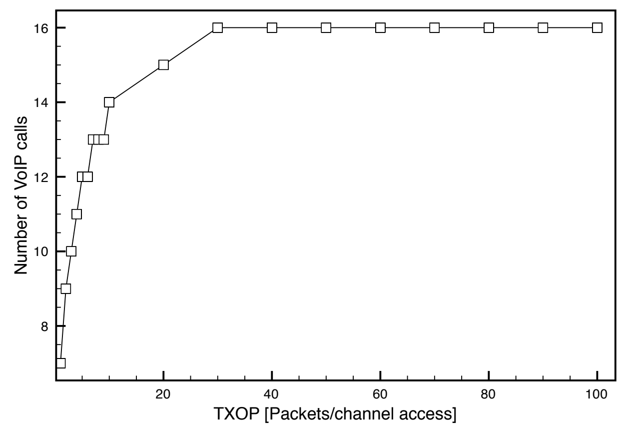

Fig. 6. Maximum number of voice calls for large values of TXOP

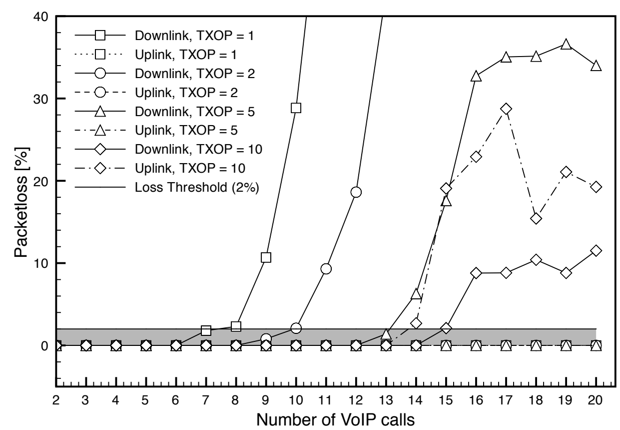

Fig. 7. Packet loss experienced by the wireless nodes for selected parameter of TXOP. The shaded area is the packet loss threshold region (0-2%).

of voice calls is limited. Figure 6 shows the asymptotic value for the number of voice calls when TXOP $\gg$ 1. For example, using G.729 voice codec with a 10 ms sampling rate, this asymptotic value is 16. The actual achievable voice capacity, however, is less than this asymptotic value. It is because increasing TXOP value will cause the bottleneck to shift from the AP to the wireless nodes. Once this happens, the wireless nodes have to wait an extended time period before gaining channel access which then results in long delay and excessive packet loss. In particular, we have found that the bottleneck shift occurs when $TXOP > C_1$ where C_1 is the number of calls which can be accommodated with a default TXOP value (TXOP = 1). The average packet loss of the AP and a wireless node obtained using simulation is shown in Fig. 7 using different TXOP values. Here it can be observed that for small TXOP values only the AP experiences some packet loss. In contrast, when TXOP = 10, it is shown that the wireless node has excessive packet loss before the AP starts to experience any loss. This indicates that the AP is no longer the bottleneck.

We now investigate the impact of the buffer size at the AP on the number of voice calls that the WLAN can support. Studies in [7] indicated that the buffer size at the AP should be proportional to the number of supported voice calls. In the following we show that there exist a minimum buffer where the maximum capacity is achieved and further increasing K will not increase the voice capacity. Using our analytical model and simulation we obtain the maximum voice capacity for different buffer sizes and TXOP values, and show results for G.729 and G.711 voice codec and 10 ms sampling rate in Table II and Table III, respectively. Observe that the minimum buffer size (K_{min}) to achieve the maximum capacity is 30 for all the TXOP values presented in Table II. For buffer sizes smaller than K_{min} the number of voice calls which can be accommodated is reduced due to excessive packet loss at the AP.

Table II and III also show that as long as the bottleneck is at the AP (TXOP $\leq C_1$), increasing the buffer size beyond K_{min} value has no impact on the maximum number of calls that the WLAN can accommodate. Because the maximum capacity is independent of the queue size given that it is greater than K_{min}, we show below that the $M/G/1/\infty$ queue can also be used to model the AP in determining the maximum number of voice calls. In this model, there is no packet loss and the maximum capacity is calculated based on the stability condition of the AP queue. In particular, the number of voice calls is calculated based on the inequality $\lambda_a < \mu_a$, which guarantees the queue stability. Solving the above inequality for $\lambda_a = \mu_a$, the closed form expression for voice capacity as a function of TXOP parameter, denoted as $f(TXOP)$, is given by

TABLE II

COMPARISON OF ANALYTICAL AND SIMULATION RESULTS FOR THE MAXIMUM NUMBER OF G.729 VOICE CALLS WITH A 10 MS SAMPLING RATE FOR DIFFERENT BUFFER SIZE K AND SELECTED VALUES OF TXOP PARAMETER (M = ANALYTICAL MODEL, S = NS-2 SIMULATION).

	TXOP [Packets/channel access]							
	1		2		5		7	
K	M	S	M	S	M	S	M	S
10	5	6	7	9	10	11	10	11
20	6	6	8	9	11	12	12	13
30	7	6	9	9	12	12	13	13
40	7	7	9	9	12	13	13	13
50	7	7	9	9	12	13	13	14
100	7	7	9	9	12	13	13	13

TABLE III

COMPARISON OF ANALYTICAL AND SIMULATION RESULTS FOR THE MAXIMUM NUMBER OF G.711 VOICE CALLS WITH A 10 MS SAMPLING RATE FOR DIFFERENT BUFFER SIZE K AND SELECTED VALUES OF TXOP PARAMETER (M = ANALYTICAL MODEL, S = NS-2 SIMULATION).

	TXOP [Packets/channel access]							
	1		2		5		7	
K	M	S	M	S	M	S	M	S
10	5	6	7	8	9	10	10	11
20	6	6	8	8	10	11	11	12
30	6	6	8	9	11	12	12	12
40	6	6	8	9	11	11	12	12
50	6	6	8	8	11	11	12	12
100	6	6	9	9	11	11	12	12

$$f(TXOP) = \frac{1}{2}\sqrt{\left(\frac{\alpha + (TXOP-1)\,\beta}{\gamma}\right)^2 + \frac{4\,TXOP}{\gamma}} - \frac{\alpha + (TXOP-1)\,\beta}{2\gamma} \quad (14)$$

where

$$\alpha = \left(T_s + T_{DIFS} + \overline{w}_a\sigma + \frac{\bar{t}_a}{2}\right)\lambda_n,$$
$$\beta = T_{\overline{s}}\lambda_n,$$
$$\gamma = \left(\frac{\lambda_n}{\mu_a}T_s + \frac{1}{2}\frac{\lambda_n}{\mu_a}\bar{t}_n\right)\lambda_n.$$

We study the accuracy of the $M/G/1/\infty$ model by comparing the maximum number of calls calculated using (14) with results obtained from (13) which is based on the $M/G/1/K$ model. Results for G.729 and G.711 codecs with 10 ms sampling rate based on both models are depicted in Figs. 8 and 9, respectively. It can be seen that the results have good agreement over a range of TXOP values. Thus the voice capacity in a WLAN can be calculated using either $M/G/1/K$ or $M/G/1/\infty$ models. However, note that for the later we do have a closed form expression for the achievable voice capacity.

V. VOIP CAPACITY APPROXIMATION

In this section we propose a simple yet accurate approximation formula to estimate the number of voice calls in an IEEE 802.11 WLAN. To this end we provide a heuristic recursive formula which provides the maximum number of voice calls for given TXOP parameter based on the previous obtained capacity. The approximation formula is a simple alternative to obtain the voice capacity, as it does not require the repeated calculation of the fixed-point formulation developed in Sec. III. This formula also allows us to gain further insight into the voice capacity and the correlation of a variety of network parameters such as TXOP. Using this approximation formula, we will show that the voice capacity is bounded. Additionally we will show that the capacity increase only depends on C_1 capacity and the TXOP parameter. Furthermore we obtain the optimal value of TXOP parameter that maximizes the voice capacity. This optimal value is also the threshold value when the AP is no longer the bottleneck of the WLAN.

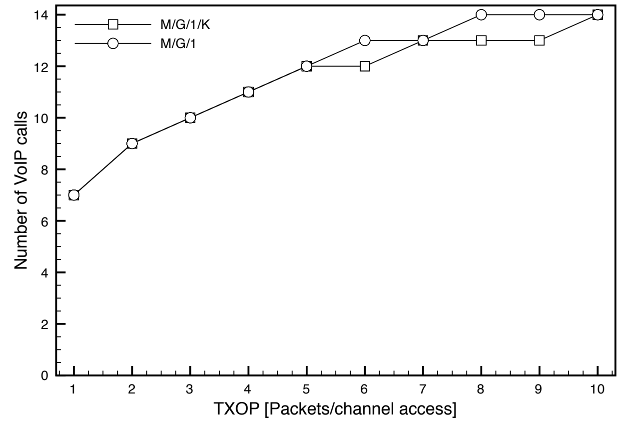

Fig. 8. Comparison of the maximum number of G.729 VoIP calls with a 10 ms sampling rate for the two different queueing models

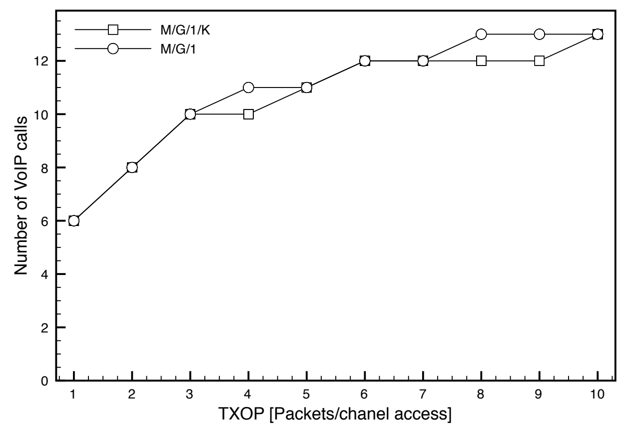

Fig. 9. Comparison of the maximum number of G.711 VoIP calls with a 10 ms sampling rate for the two different queueing models

In Sec. III we have shown that an increased TXOP parameter at the AP improves the voice capacity significantly. We observed, however, that the increase in voice capacity is not linear with TXOP. In particular, we observed that the number of additional calls gained by the increased TXOP parameter is decreasing. An investigation on the correlation of TXOP parameter and the improved voice capacity revealed that there exists a mutual dependency between TXOP and the additional number of voice calls gained per TXOP.

Recall that in this scenario we consider an IEEE 802.11 infrastructure WLAN consisting of one AP and $N-1$ wireless nodes. Each wireless node maintains a full-duplex VoIP call to a node outside of the WLAN. We have shown that for TXOP = 1 the WLAN can accommodate C_1 calls and that the AP is the bottleneck of the network. Increasing TXOP from 1 to 2 allows the AP to send 2 packets per each channel access. Thus in addition to the C_1 voice calls that have already been supported previously, the AP can now further send $C_1/2$ voice packets on the downlink. As long as the AP is the bottleneck in the WLAN, the number of additional calls in this scenario (i.e. TXOP = 2) can be approximated as $\left(\frac{C_1}{2}\right)/2$. This is because each additional call adds two packets, one on the downlink from the AP and another on the uplink from the wireless node. As a result, the total number of calls using TXOP = 2 is $C_1 + \left(\frac{C_1}{2}\right)/2$. Based on similar arguments the voice capacity with increasing TXOP can be approximated as follows:

$$\begin{aligned}
TXOP &= 1 \rightarrow C_1, \\
TXOP &= 2 \rightarrow C_1 + \left(\frac{C_1}{2}\right)/2, \\
TXOP &= 3 \rightarrow C_1 + \left(\frac{C_1}{2}\right)/2 + \left(\frac{C_1}{3}\right)/2, \\
&\vdots \\
TXOP &= n \rightarrow C_1 + \left(\frac{C_1}{2}\right)/2 + \ldots + \left(\frac{C_1}{n}\right)/2, \\
&= C_1 + \sum_{TXOP=2}^{n} \left(\frac{C_1}{TXOP}\right)/2. \qquad (15)
\end{aligned}$$

Based on (15) and the closed form expression for the voice capacity for TXOP = 1 in (14), the number of additional voice calls for $TXOP > 1$, denoted as Γ_{TXOP}, can be approximated by

$$\Gamma_{TXOP} = \left(\frac{f(TXOP = 1)}{TXOP}\right)/2. \qquad (16)$$

We can now define a recursive formula for the voice capacity. Let $f_{approx}(TXOP)$ denote the estimated number of VoIP calls for given TXOP value. Then the VoIP capacity approximation for $TXOP > 1$ is given by

$$f_{approx}(TXOP) = f_{approx}(TXOP - 1) + \Gamma_{TXOP} \qquad (17)$$

In Figs. 10 and 11 we compare approximation results with results obtained by the analytical model for the G.729 and G.711 codecs for a variety of parameter. Observe that the approximation results match those of the analytical model closely. This confirms that our proposed approximation formula is a simple alternative to an analytical model, and can be used for a variety of applications. Note that the analytical results have been validated in Sec. IV, and thus simulation results are omitted in Figs. 10 and 11.

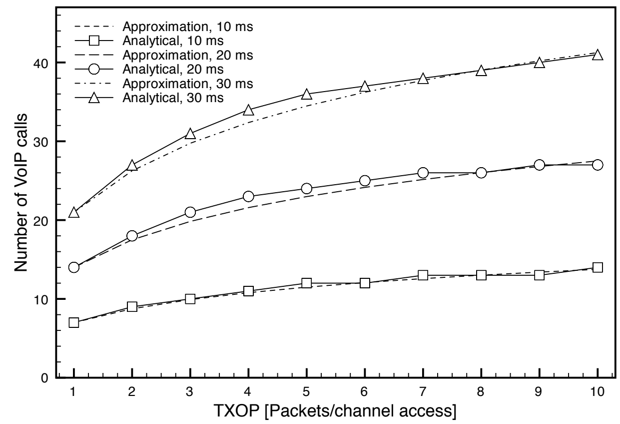

Fig. 10. Comparison of the number of G.729 voice calls obtained by (14) and (17) for different sampling rates and increasing TXOP parameter.

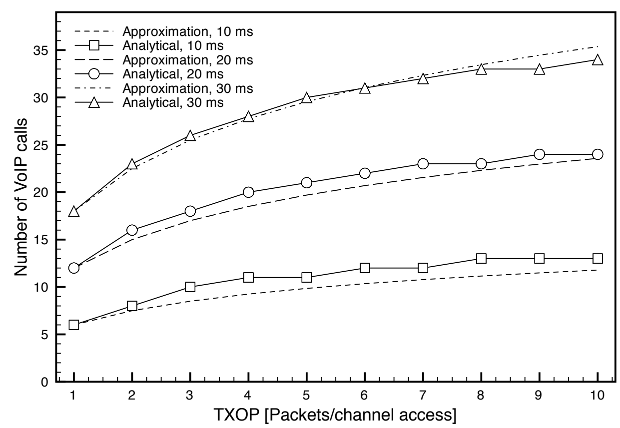

Fig. 11. Comparison of the number of G.711 voice calls obtained by (14) and (17) for different sampling rates and increasing TXOP parameter.

The approximation formula allows us to obtain further insight into the voice capacity in WLAN, and to study the relationship of various network parameters.

In particular, from (16) and (17) it can be seen that the voice capacity for increasing TXOP only depends on the TXOP value and the initial voice capacity obtained using default parameter settings.

Furthermore with an increasing value of TXOP parameter, it can be shown that Γ_{TXOP} tends to zero, thus $f_{approx}(TXOP) \approx f_{approx}(TXOP-1)$. This shows that the voice capacity is bounded. In other words using TXOP significantly improves the voice capacity in WLANs, but even then the capacity is limited. As shown, once the capacity limit is reached, further increasing TXOP will not increase the VoIP capacity. We have confirmed the voice capacity limit by simulation, and showed that the maximum number of voice calls can be achieved when $TXOP = f(TXOP = 1)$. Increasing TXOP beyond this optimal value results in excessive packets loss and long delay at a wireless node. Thus when $TXOP = f(TXOP = 1)$, the number of voice calls that can be accommodated is maximum or close to maximum.

The approximation results also show that assumptions made in [12] that TXOP should be equal to the number of wireless nodes will not hold in general, specifically when $N > f(TXOP = 1)$.

VI. Conclusion

In this paper we have developed a detailed analytical model to evaluate the performance gain that can be achieved using the configurable TXOP parameter of the IEEE 802.11e standard. We have shown that an increased TXOP for the AP can improve the voice capacity in WLAN significantly. Note that this approach does not require any changes to hardware or the protocol. We showed that there exists a optimal TXOP parameter, beyond which the voice capacity cannot be increased any further. Using simulation we confirmed this optimal TXOP parameter, and showed that any TXOP value greater than this optimal value causes the bottleneck to shift from the AP to the wireless nodes. Furthermore we have investigated the impact of the AP buffer size on the maximum number of voice calls that the WLAN can accommodate. For a given TXOP value, we have shown that there exists a minimum buffer size where this capacity is reached. Based on our observation that the number of voice calls is independent of the buffer size provided it is $\geq K_{min}$, we have induced a closed form expression for the maximum number of voice calls that a WLAN can support. To this end, this closed form expression was derived from an $M/G/1/\infty$ queueing model, and we showed that the voice capacity of a WLAN can either be calculated using an $M/G/1/K$ or an $M/G/1/\infty$ model. We used the closed form expression to propose a novel approach for the voice capacity estimation. We developed a simple yet accurate recursive approximation formula which provides the maximum number of voice calls for given TXOP parameter based on the previous obtained capacity. We showed that this approximation does not require solving the set of non-linear equations, and also allowed us to confirmed that the voice capacity is bounded and that there exists a mutual dependency between TXOP and the additional number of voice calls.

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