VoIP Capacity - Analysis, Improvements and Limits in IEEE 802.11 Wireless LAN

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***Abstract*—Wireless voice over IP (VoIP) is an important emerging service in telecommunication 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 the TXOP value. Finally, we propose a simple yet accurate approximation for voice capacity estimation and provide some insights gained from the approximation.**

***Index Terms*—VoIP capacity, Transmission Opportunity, Buffer size, 802.11 Wireless LAN, EDCA**

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], [4], [5] 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 10 ms sampling rate [2], [6] .

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) [7]. It extends the access mechanism in the medium access control (MAC) protocol of the IEEE 802.11 standard (referred to as enhanced distributed channel access, EDCA) by allowing adjustment of a number of MAC parameters that were previously fixed. The 802.11e standard has become part of the new specification of the IEEE 802.11 standard in 2007 [8].

There have been several papers in the literature that investigate the voice capacity of a WLAN based on both the 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 network which limits the number of voice calls it can support. 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 traffic. The maximum number of voice calls that a WLAN can accommodate based on real measurements has also been reported in [4], [5], [6]. To tackle the voice capacity limit, various solutions have been proposed. Wang et al. [9] for example, suggest 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. [10] use a so-called transmission opportunity (TXOP) parameter of EDCA 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 an analytical model to evaluate the performance gain that can be obtained using the above TXOP parameter. We confirm analytically 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 [10], but with a smaller TXOP value. Our analytical results are validated using the NS-2 simulation [11] for a range of different voice codecs and TXOP values. Also, we have conducted WLAN testbed measurements to further validate the analytical and simulations results. In addition, we investigate the impact of the buffer size at the access point (AP) on the maximum number of voice calls. We will show that there exists a minimum buffer size with which the voice capacity

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is maximized. 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 then propose a simple yet accurate voice capacity approximation and provide some insights gained from the approximation.

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. This is followed by the detailed analytical model in Section III to assess the improvements of the proposed solution. We validate our queueing model by comparing analytical results with results obtained from simulation in Section IV. In Section V we investigate the impact of the buffer size on the voice capacity, and develop a closed form expression for the voice capacity. In Section VI we propose a simple approximation to estimate the number of voice calls that a WLAN can support. We conclude the paper in Section VII.

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.

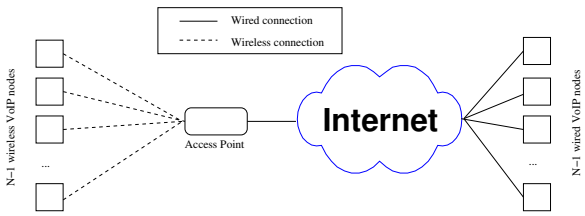

Fig. 1. VoIP network topology

In this network voice traffic to and from any mobile node 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 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 buffer. If the number of voice calls keeps increasing, at some stage, the AP will start to drop packets due to buffer overflows 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.

To address the bottleneck and improve voice capacity, similar to [10], 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 [7]. 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. In the rest of the paper, we denote the number of consecutive packets that can be transmitted during the $TXOPLimit$ period by η. We will use η and the term *TXOP value* interchangeably. Figure 2 shows the simplified structure of packets transmission (TXOP-frame) from the AP using TXOP.

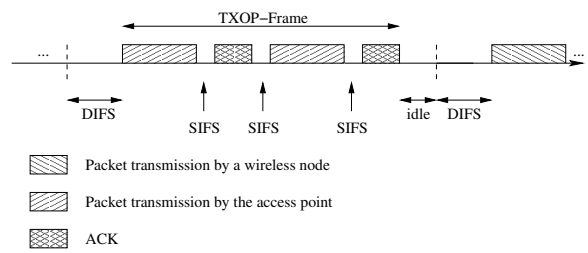

Fig. 2. TXOP-frame on the wireless channel as seen by the access point

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

III. Analytical Model

Let us consider an IEEE 802.11 infrastructure WLAN 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 using the same voice codec. Here we only consider voice traffic and no other cross traffic, such as TCP traffic. We assume EDCA basic access is used over an ideal channel without interference or hidden terminals. Unless stated otherwise, the indices a and n correspond to the access point and a wireless node, respectively.

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. Assuming packets arrive at a node according to a Poisson process, an $M/G/1/K$ queueing model can then be used to model the wireless node and the AP, where the latter serves packets in batches of η packets. 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. Also note that even though the assumption of packets arrival as Poisson process is coarse, it can in part model the superposition of multiple periodic streams at the AP which in turn is the bottleneck determining the voice capacity in this network.

The queue utilization can be expressed as $\rho_i = \lambda_i/\mu_i,\ i \in \{n, a\}$ where ρ_i is also the probability that a station has a packet to send. Thus a station will be idle with probability $1-\rho_i$. Our analysis is evolved around a fixed-point formulation between the collision probability seen by a packet transmitted on the channel and the attempt

rate per slot of a station. The latter is conditioned on that a station has packets to send, i.e. the queue is nonempty. In the following we establish the fixed-point equations and derive the average service times associated with the AP and wireless node, respectively.

The 802.11 protocol [12] specifies that a station 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 $CW_{min} = W$. 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. Let $\tau_i,\ i \in \{n, a\}$ be a conditional attempt rate per slot of a station (i.e. the ratio of the number of attempts to the time spent in backoff measured in slots provided that the station has packets to send). Knowing that ρ_i is the probability that a station has packets to send, the probability that a station is attempting to transmit in a slot is given by $\rho_i \tau_i,\ i \in \{n, a\}$ [13]. Assuming each packet collides with constant and independent probability $c_i,\ i \in \{n, a\}$, a fixed point formulation can be established for a wireless node and the AP, respectively as

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

$$c_a = 1 - (1 - \rho_n \tau_n)^{N-1}, \quad (2)$$

where $\rho_n \tau_n$ and $\rho_a \tau_a$ is a function of c_n and c_a. Note that (1) and (2) are based on the fact that a wireless node can collide with one of the remaining $N-2$ wireless nodes or the AP, whereas the AP can only collide with one of the $N-1$ wireless nodes. To complete the fixed point equations, we first devise the average service time of a station and then determine the function $\rho_i \tau_i,\ i \in \{n, a\}$.

As in [2], the average packet service time of a wireless node can be decomposed into three components: (i) the actual collision and successful transmission time of the packet; (ii) the interruptions to the backoff due to collisions and successful transmissions by the remaining $N-2$ wireless nodes and the AP; and (iii) the average backoff during which the channel is sensed idle.

For the first component, the collision and successful transmission time of a voice packet is defined as

$$T_c = T_p + T_{ACK_{TO}} + T_{AIFS}, \quad (3)$$

$$T_s = T_{AIFS} + T_p + T_{SIFS} + T_{ACK}, \quad (4)$$

where T_p, T_{ACK} are the transmission times of the packet and acknowledgement, $T_{ACK_{TO}}$ is the ACK timeout period of an unsuccessful transmission, and T_{AIFS}, T_{SIFS} are the duration of the arbitrary and short inter-frame spaces in $[\mu s]$, respectively. As a packet can collide several times before its successful reception, the average collision time caused by a single wireless node during a packet service time $1/\mu_n$ is given by

$$\frac{\bar{t}_n}{2} = \frac{1}{2}\sum_{i=1}^{R} c_n^i (1 - c_n) i T_c \underset{R\to\infty}{\approx} \frac{T_c\ c_n}{2(1 - c_n)}. \quad (5)$$

Note that the $1/2$ factor in the above expression is based on the assumption that a collision is due to simultaneous transmissions from two stations only, and thus the average collision time caused by a wireless node is half of the total collision time experienced by all stations. Similarly, the average collision time caused by the AP is given by

$$\frac{\bar{t}_a}{2} = \frac{T_c\ c_a}{2(1 - c_a)}, \quad (6)$$

where T_c is defined as in (3) because only the first packet in the transmitted TXOP-frame from the AP would experience a collision.

For the second component, the packet service time of a wireless node $1/\mu_n$ consists of, on average,

$$(N-2)\lambda_n \frac{1}{\mu_n} T_s + \frac{\lambda_a}{\eta}\frac{1}{\mu_n}(T_s + (\eta - 1)T_s^*) \quad (7)$$

successful transmission time, and

$$(N-2)\lambda_n \frac{1}{\mu_n}\frac{\bar{t}_n}{2} + \frac{\lambda_a}{\eta}\frac{1}{\mu_n}\frac{\bar{t}_a}{2} \quad (8)$$

collision time from other $(N-2)$ wireless nodes and the AP, where $T_s^* = T_p + 2T_{SIFS} + T_{ACK}$ and $\bar{t}_a/2$ is defined in (6). Note that the term λ_a/η in (7) and (8) is due to the fact that for each channel access the AP can send up to η packets in its TXOP-frame.

For the third component, the average backoff of a wireless node is given as [2]

$$\overline{w}_n = \sum_{i=0}^{R-2}(1 - c_n)c_n^i \frac{\left(\prod_{j=1}^{i}\alpha_j W\right) - 1}{2} + c_n^{R-1}\frac{2^m W - 1}{2}, \quad (9)$$

where $\alpha_j = 2$ if $j \leq m$, and 1 otherwise. Note that $\overline{w}_n$ in (9) is given in backoff slots, each of a length of $\sigma[\mu s]$.

Based on (9) the conditional attempt rate per slot of a wireless node can be immediately derived as

$$\tau_n = \frac{\sum_{i=0}^{R} c_n^i}{\overline{w}_n}, \quad (10)$$

where the numerator is the average number of transmission attempts per packet of the wireless node during its backoff period.

However, in contrast to the model in [2], the actual length in μs of the average backoff is not simply $\overline{w}_n \sigma$ because the backoff counter is managed differently in EDCA [14]. In particular, after every channel activities, the backoff counter in EDCA is resumed one slot time before the AIFS timer elapses. On average, during the backoff of a wireless node, the number of time the channel is sensed busy is given by

$$\epsilon_n = 2\rho_n\left((N-2) + \frac{N-1}{\eta}\right), \quad (11)$$

which can be devised based on the average number of successful transmissions and collisions from (7), (8) and the fact that arrival of packets at the AP is the superposition of many individual voice traffic in that network, i.e.,

$$\lambda_a = (N-1)\lambda_n. \tag{12}$$

Furthermore, according to the EDCA mechanism, a station has to wait either an additional slot when the medium is idle or an additional AIFS period when the medium is sensed busy, before attempting to transmit. The former occurs with a probability $1-c_n$, while the latter is with a probability c_n. Thus the third component in $[\mu s]$ can be expressed as

$$(\overline{w}_n - \epsilon_n)\sigma + (1-c_n)\sigma + c_n T_{AIFS}. \tag{13}$$

Based on (4), (5), (7), (8) and (13) the average packet service time of a wireless node is given by

$$\begin{aligned}\frac{1}{\mu_n} &= ((N-2)\rho_n + 1)\left(T_s + \frac{\bar{t}_n}{2}\right) \\ &+ \frac{N-1}{\eta}\rho_n\left(T_s + (\eta-1)T_s^* + \frac{\bar{t}_a}{2}\right) \\ &+ (\overline{w}_n - \epsilon_n + 1 - c_n)\sigma + c_n T_{AIFS}. \end{aligned}\tag{14}$$

It remains to find the attempt probability and average service time of the AP. By replacing c_n with c_a in (10) the conditional attempt rate per slot of the AP immediately follows as

$$\tau_a = \frac{\sum_{i=0}^{R} c_a^i}{\overline{w}_a}, \tag{15}$$

where $\overline{w}_a$ is the average backoff of the AP given by

$$\begin{aligned}\overline{w}_a = \sum_{i=0}^{R-2}(1-c_a)c_a^i \frac{\left(\prod_{j=1}^{i}\alpha_j W\right) - 1}{2} \\ + c_a^{R-1}\frac{2^m W - 1}{2}.\end{aligned}\tag{16}$$

Because the AP is allowed to send up to $\eta \geq 1$ packets consecutively, the average service time of the TXOP-frame consists of two parts: (i) the average service time of the first packet in the frame ($1/\mu_{a1}$), and (ii) the total average service time of all the subsequent packets in that frame ($1/\mu_{a2}$).

The average service time of the first packet in a TXOP-frame is calculated similar to (14) with appropriately using the collision probability c_a instead of c_n. Therefore the first part of the average service time of the TXOP-frame is given by

$$\begin{aligned}\frac{1}{\mu_{a1}} &= (N-1)\frac{\lambda_n}{\mu_a}\left(T_s + \frac{\bar{t}_n}{2}\right) \\ &+ \left(T_s + \frac{\bar{t}_a}{2}\right) \\ &+ (\overline{w}_a - \epsilon_a + 1 - c_a)\sigma + c_a T_{AIFS}, \end{aligned}\tag{17}$$

where $\bar{t}_a/2$ is given in (6), $\overline{w}_a$ in (16) and ϵ_a is defined as

$$\epsilon_a = 2(N-1)\frac{\lambda_n}{\mu_a}. \tag{18}$$

After the first packet of the TXOP-frame is served, all the subsequent packets in that frame have the same service time of T_s^* $[\mu s]$. It is because the rest of packets in the frame do not need to content for the channel access. Thus

$$\frac{1}{\mu_{a2}} = (\eta-1)T_s^*, \tag{19}$$

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

$$\frac{1}{\mu_a} = \frac{1}{\eta}\left(\frac{1}{\mu_{a1}} + \frac{1}{\mu_{a2}}\right). \tag{20}$$

Equations (1), (2), (10), (14), (15) and (20) constitute a non-linear system of equations that can be solved iteratively to obtain the collision probabilities c_i and the conditional attempt probability τ_i, as well as $\rho_i = \lambda_i/\mu_i,\ i \in \{n, a\}$.

Having obtained $\rho_i,\ i \in \{n, a\}$, we require the average packet loss κ of a voice call to be less than 2% to have an acceptable level of quality [15]. The maximum number of supported voice calls (C) is the number of calls such that the packet loss probability (p) of a voice call is kept 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

$$p_a \approx \frac{(1-\rho_a)\rho_a^K}{1-\rho_a^{K+1}}. \tag{21}$$

To obtain C, we repeatedly solve the above non-linear system of equations with incremental number of voice calls. Although the voice quality depends on both the average packet loss and the end-to-end delay, capacity C is calculated based on the packet loss only. It is because when the AP is the bottleneck, packet loss will be the main factor causing degradation in voice quality. We will look at the delay as well when this bottleneck is shifted as showed in the next section. Note that (21) is a blocking probability of an M/M/1/K queue [16] by assuming exponential service time at the AP and thus it is only an approximation for the packet loss probability. Also we assume here that packet loss is only due to the buffer overflow at the AP as explained in Section II, and not caused by packet collision or channel errors. In the next section, however, we will show that this approximation is reasonable and matches well with the packet loss observed from the simulations.

IV. Validation and Results Discussion

We validate our model in this section by comparing the analytical results with simulation, and briefly show the impact of the different backoff behavior of EDCA and DCF. Simulation is performed using NS-2 (version 2.28) with the EDCA extension from the TU Berlin [17]. To differentiate between the AP and the wireless nodes when EDCA is used, the AP was placed in a different access category where only the $TXOPLimit$ parameter is set to be different. Voice traffic is generated as a periodic stream of packets for a given voice codec to study the sensitivity of the Poisson approximation used in the analytical model. Table I provides a summary of parameters used in our analysis and simulation. In the following, for validation purposes, we set the buffer size at the AP and wireless nodes all equal to 50 packets as indicated

TABLE I
NETWORK PARAMETERS FOR AN 802.11 WIRELESS LAN

Channel rate	11MBit/s
Basic rate	1MBit/s
Backup Slot length σ	20 μs
CW_{min}	32
CW_{max}	1024
T_{SIFS}	10 μs
T_{AIFS}	50μs
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 Table I. The impact of the different buffer values will be studied in Section V.

In Fig. 3 we show the average packet service rate of the AP for a G.729 voice call with 10 ms sampling rate using an EDCA and DCF backoff process with a default TXOP value. Assuming $T_{AIFS} = T_{DIFS} = 50\mu s$, then the AP achieves a slightly higher average packet service rate when EDCA is used. Results obtained in this section below are based on this increased service rate using the EDCA mechanism.

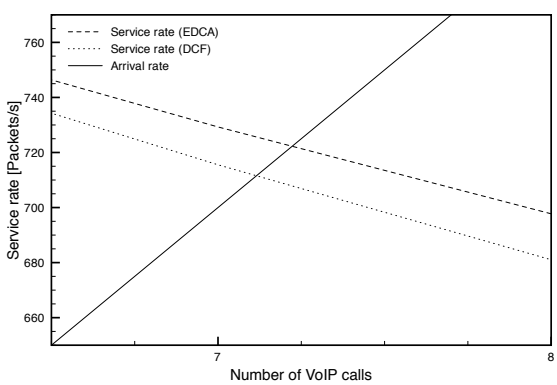

Fig. 3. Average packet service rate of the AP using the DCF and EDCA backoff for a G.729 voice call with a 10 ms sampling rate and default TXOP parameter ($\eta = 1$)

In addition to the analytical and simulations results, measurements obtained from a WLAN testbed are also presented in this section. The testbed is an infrastructure WLAN and consists of one AP and multiple wireless nodes as depicted in Fig. 1. In this testbed, a desktop PC is used as an AP and the wireless nodes are a combination of desktop PCs, Netbooks as well as embedded devices. Each system is equipped with an Atheros 802.11 wireless card using a version of the MADWIFI [18] wireless driver. The MADWIFI wireless drivers are used because of their support of the 802.11e standard [10]. With the exception of the embedded devices, each station runs Linux 2.6.32. The embedded devices are configured with TinyBSD 6.2. Each station is connected to a control station via a wired Ethernet link. The wired connection allows the control of the stations without interfering the wireless traffic. For the measurements the wireless channel rate is fixed to 11MBit/s on a pre-selected channel using 802.11b. The RTS/CTS mechanism has been deactivated along with other manufacturers features such as *fast-frames* or *bursting*. Table II provides an overview of the system hardware. The 802.11e MAC parameter are consistent with those shown in Table I.

TABLE II
SYSTEM OVERVIEW

System	OS	WLAN	Chipset
3 x Desktop PC	Linux 2.6.32	802.11a/b/g	AR5413
5 x Acer Netbooks	Linux 2.6.32	802.11b/g	AR928X
3 x Embedded devices	TinyBSD 6.2-Release	802.11a/b/g	AR5212

In Fig. 4 we show the packet loss probability at the AP using (21) for G.729 and G.711 voice calls with 10 ms sampling rate and selected values of TXOP parameter η. It can be observed that for $\eta = 1$ (default TXOP) the network can accommodate up to 7 voice calls using a G.729 codec before the packet loss threshold $\kappa = 0.02$ is exceeded. For $\eta = 5$ the voice capacity is almost doubled with 12 voice calls using the same codec. Similar number of voice calls are reported in [10], but with a larger TXOP value where the authors set the TXOP value equal to the number of expected voice calls. Also observe that the measurement results of the testbed for a G.711 voice call with a 10 ms sampling rate match the analytical results closely. Note that due to a limited number of testbed equipments, experimental results are only shown for TXOP = 1 and TXOP = 2 cases.

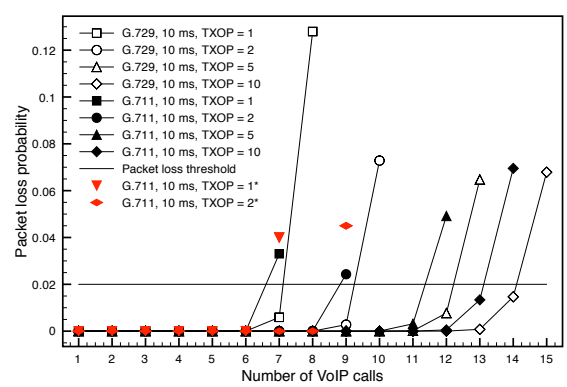

Fig. 4. Packet loss probability at the AP for G.729 and G.711 voice calls with a 10 ms sampling rate and selected values of TXOP. Note that items marked by * are results obtained through measurements in a WLAN testbed

In Figs. 5 and 6 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 there is a one call difference between the analytical results and values obtained from simulation due to rounding errors. Also the approximation in (21) is

conservative but does not lead to excessively conservative dimensioning.

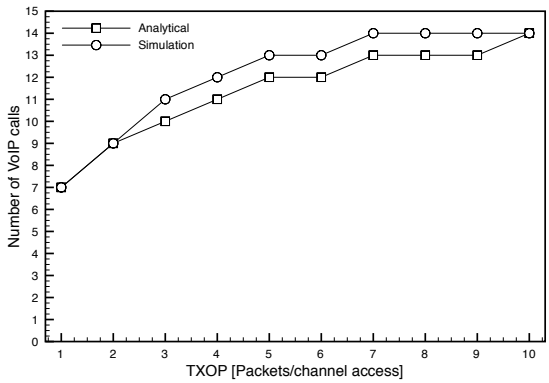

Fig. 5. Analytical and simulation results for G.729 voice calls with a 10 ms sampling rate and increasing values of TXOP parameter

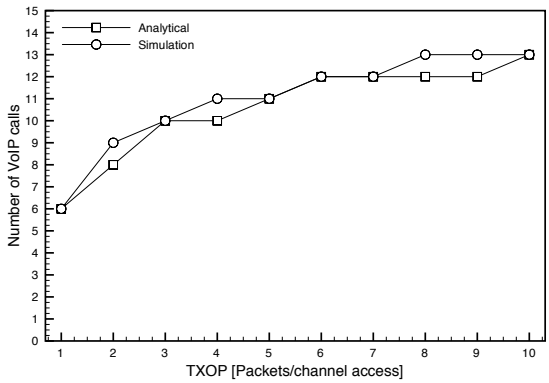

Fig. 6. Analytical and simulation results for G.711 voice calls with a 10 ms sampling rate and increasing values of TXOP parameter

Even though setting larger TXOP parameter at the AP can improve significantly the voice capacity, the maximum number of voice calls is limited. Figure 7 shows the asymptotic value for the number of voice calls when $\eta \gg 1$. For example, using G.729 voice codec with 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 period of time before gaining channel access which then results in long delay and excessive packet loss. In particular, we have identified that the bottleneck shift occurs when $\eta > C_1$ where C_1 is the number of calls which can be accommodated with a default TXOP value ($\eta = 1$).

The shift of the bottleneck at some large TXOP value is demonstrated by monitoring the average loss and delay that a wireless node is experiencing with increasing TXOP value. The average packet loss of the AP and a wireless node for increasing values of TXOP parameter is shown in Fig. 8. It can be seen that for small values of TXOP parameter, i.e. $\eta = 5$, only the AP experiences packet loss (on the downlink). In contrast, when $\eta = 10$, it is shown that the wireless node has

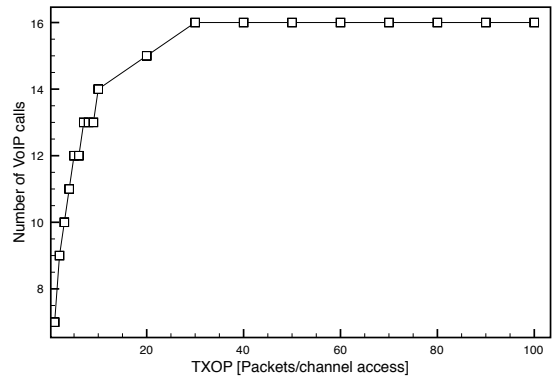

Fig. 7. Number of G.729 voice calls (asymptotic) with a 10 ms sampling rate for large values of TXOP parameter

experienced excessive packet loss (on the uplink) before the AP starts to lose any of its packets. The discrepancy in packet loss is due to the use of the $M/M/1/K$ model rather than a model with periodic arrivals and deterministic service time. The bottleneck shift also contributes to the difference between analysis and simulation seen in Fig. 8. However, note that despite this inaccuracy, the voice capacity estimation remains reasonably accurate (accounting for rounding errors) as long as the loss is less than 2%.

In Fig. 9 we show the average end-to-end network delay experienced by the wireless nodes. Observe that for $\eta < C_1$ the average end-to-end network delay as seen by the wireless nodes (uplink) is small. For $\eta = 10$ however, this delay rapidly increases and exceeds a maximum delay bound, beyond which the quality of a voice call can not be maintained. Here the delay bound is set to 150 ms as in [6]. The observation that with increasing TXOP value the wireless nodes experience excessive loss and long delay indicates that the AP is no longer the bottleneck of the network. It also shows that $\eta = C_1$ is an optimal parameter setting to obtain the maximum voice capacity in 802.11 WLAN.

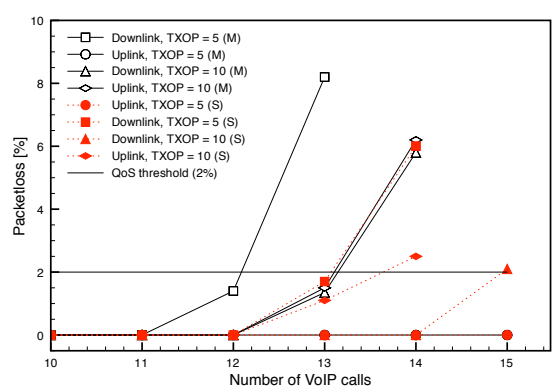

Fig. 8. Average packet loss experienced by the AP and a wireless node for selected values of TXOP parameter and a G.729 codec with a 10 ms sampling rate (M = analytical model, S = NS-2 simulation)

Note that even though CW_{min} can be used in conjunction with $TXOPLimit$, using our model we find that only

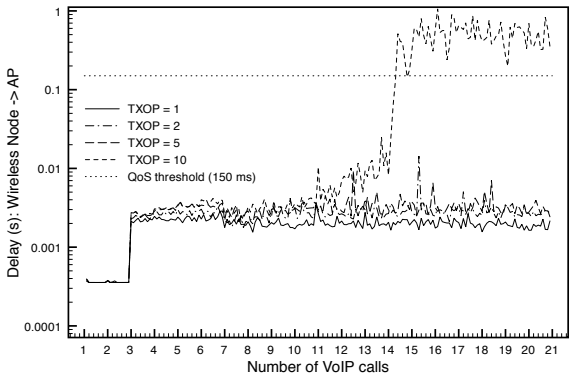

Fig. 9. Average uplink delay experienced by the wireless nodes for selected values of TXOP parameter (NS-2 simulation) and a G.729 codec with a 10 ms sampling rate

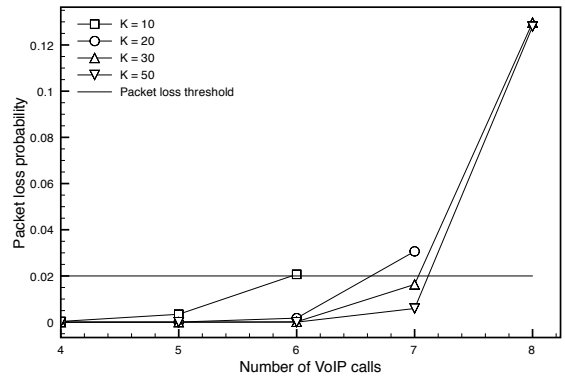

Fig. 10. Packet loss probability at the AP for a G.729 voice call with a 10 ms sampling rate, TXOP = 1 and variable buffer size K

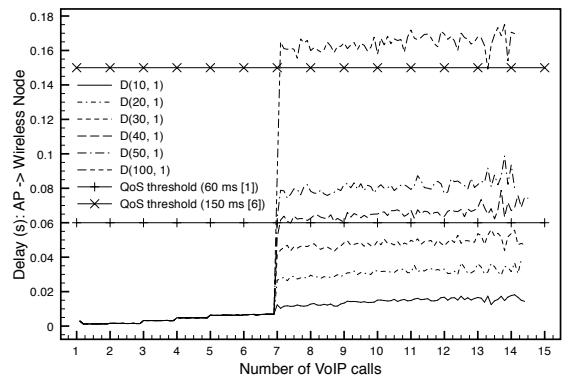

Fig. 11. Average downlink network delay for different buffer sizes K with default TXOP parameter and a G.729 codec with a 10 ms sampling rate (D(K, η))

marginal increase in voice capacity is achieved when both parameters are used to give the AP advantages in accessing the channel. The results concerning CW_{min} is in line with previous results reported in [19], [20], [21], [22].

V. Impact of the buffer size on the voice capacity

In the following the analytical model developed in Sec. III is used to investigate the impact of the AP's buffer on the voice capacity while allowing simultaneously the use of TXOP parameter. Furthermore, the buffer of the wireless nodes are set to be the same as that of the AP through this section. The impact of buffer on the voice capacity has previously been studied in [10], [23]. Using throughput as an indicator for the voice capacity, it was shown in [23] that larger buffer increases the voice capacity. However, this increase is only marginal, because the transition from a low-loss and low-delay environment to a high-loss and high-delay environment is rapid. With adjustable CW_{min} and TXOP parameters the authors in [10] suggested that the size of the AP's buffer should be proportional to the number of accommodated voice calls in a WLAN. Our findings below, however, indicate a different insight via calculating the voice capacity using various TXOP values and increasing AP's buffer sizes. In Table III and IV we show results obtained by our analytical model and simulation for the G.729 and G.711 voice codecs using 10 ms sampling rate, respectively. Observe that for a range of TXOP values, there is a minimum buffer size ($K_{min} = 30$) at which the maximum voice capacity is achieved. For buffers with smaller size than K_{min}, however, the number of voice calls which can be accommodated is reduced due to excessive packet loss at the AP, as shown in Fig. 10. It can also be seen in Fig. 10 that with increasing the buffer beyond K_{min} the packet loss at the AP is reduced slightly, however, the overall capacity does not change.

In addition, we show in Fig. 11, the average end-to-end delay of a voice call on the downlink for different buffer sizes K and default TXOP value ($\eta = 1$). Observe that with increasing buffer size the delay on the downlink increases. This increasing delay with larger buffer size is not unexpected, however, for a buffer size of K_{min}, observe that the average delay is still below the strict delay bound of 60 ms used in [1] and well below the delay bound of 150 ms applied in [6]. This shows that the voice capacity in this case is no longer limited by packet loss, but is now limited by the end-to-end delay of an individual call. The delay threshold, however, depends on the users perception of voice quality and what is the tolerable delay. Our results also show that K_{min} is an optimal buffer size to achieve the maximum voice capacity, which also satisfies strict delay constraints.

TABLE III
Comparison of analytical and simulation results of the maximum number of G.729 voice calls with a 10 ms sampling rate for different buffer sizes K and selected values of TXOP parameter η (M = analytical model, S = NS-2 simulation)

η	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

$$f(\eta) = \frac{1}{2}\sqrt{\left(\frac{\alpha + (\eta - 1)\, T_s^*}{\beta}\right)^2 + \frac{4\ \eta}{\beta}} - \frac{\alpha + (\eta - 1)\, T_s^*}{2\beta}, \quad \text{where} \tag{22}$$

$$\alpha = \left(T_s + (\overline{w}_a - \epsilon_a + 1 - c_a)\sigma + c_a T_{AIFS} + \frac{\bar{t}_a}{2}\right)\lambda_n, \quad \beta = \left(\frac{\lambda_n}{\mu_a} T_s + \frac{\lambda_n \bar{t}_n}{2\mu_a}\right)\lambda_n.$$

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

η	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

The results reported in Table III and IV also show that as long as the bottleneck is at the AP, 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. By solving equation $\lambda_a = \mu_a$ for N based on (12) and (20), the closed form expression for voice capacity as a function of η, denoted as $f(\eta)$, is given in (22).

We study the accuracy of the $M/G/1/\infty$ model by comparing the maximum number of calls calculated using (22) with results obtained from (21) which is based on the $M/G/1/K$ model. Results for G.729 and G.711 codecs with 10 ms and 20 ms sampling rate based on both models are depicted in Figs. 12 and 13. 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 latter we do have a closed form expression for the maximum number of calls.

VI. VOIP CAPACITY APPROXIMATION

In this section we propose a simple yet accurate approximation 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 estimate the voice capacity in WLAN, as it does not

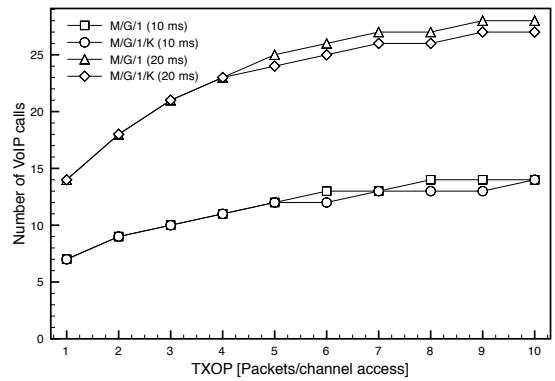

Fig. 12. Comparison of the maximum number of G.729 VoIP calls with a 10 ms and 20 ms sampling rate for the two different queueing models

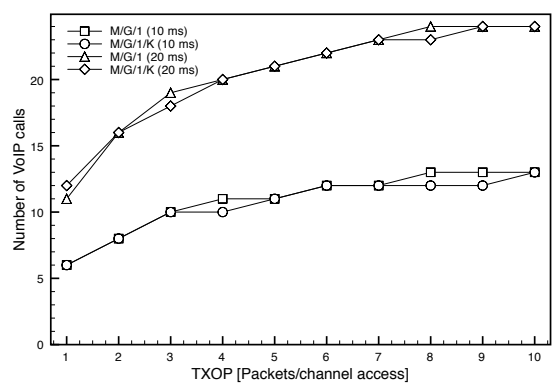

Fig. 13. Comparison of the maximum number of G.711 VoIP calls with a 10 ms and 20 ms sampling rate for the two different queueing models

require the repeated calculation of the fixed-point formulation developed in Section III. This formula also allows us to gain further insight into the voice capacity as a function of TXOP parameter. In particular, we show that the voice capacity only depends on the TXOP parameter and C_1. Furthermore, we use the approximation to obtain the optimal value of TXOP parameter that maximizes the voice capacity. We argue that this optimal value is also a threshold value where the AP is no longer the bottleneck limiting the number of voice calls the WLAN can support.

We have shown earlier that for $\eta = 1$ the WLAN can accommodate C_1 duplex calls and that the AP is the bottleneck of the network. In other words, there are in average C_1 packets sent in each direction of the bidirectional traffic flow in the network. Increasing η from 1 to 2 enables the AP to send C_1 packets in $C_1/2$ channel accesses (i.e., 2 packets per each channel access) and thus has in average another $C_1/2$ channel

accesses left on the downlink. 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. $\eta = 2$) can be approximated as $\left(\frac{C_1}{2}\right)/2$. This is because each additional call adds an equal number of packets on the uplink and the downlink voice traffic. As a result, the total number of calls using $\eta = 2$ is $C_1 + \left\lceil \left(\frac{C_1}{2}\right)/2 \right\rceil$. The above approach is illustrated in Fig. 14 where C_1 is assumed to be four calls. In this scenario, increasing η from 1 to 2 reduces the number of packets sent by the AP by 50% which then enables us to add a new call as depicted in Fig. 14.

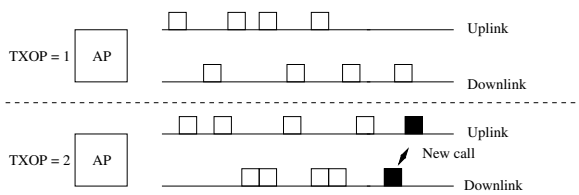

Fig. 14. Illustration of the concept used to develop the voice capacity approximation.

Based on similar arguments the voice capacity with increasing TXOP value can be approximated as follows:

$$\begin{aligned}
\eta &= 1 \rightarrow C_1, \\
\eta &= 2 \rightarrow C_1 + \left\lceil \left(\frac{C_1}{2}\right)/2 \right\rceil, \\
\eta &= 3 \rightarrow C_1 + \left\lceil \left(\frac{C_1}{2}\right)/2 \right\rceil + \left\lceil \left(\frac{C_1}{3}\right)/2 \right\rceil, \\
&\vdots \\
\eta &= n \rightarrow C_1 + \left\lceil \left(\frac{C_1}{2}\right)/2 \right\rceil + \ldots + \left\lceil \left(\frac{C_1}{n}\right)/2 \right\rceil, \\
&= C_1 + \sum_{\eta=2}^{n} \left\lceil \left(\frac{C_1}{\eta}\right)/2 \right\rceil. \qquad (23)
\end{aligned}$$

Based on (23) and the closed form expression (22) for voice capacity with default TXOP value ($\eta = 1$), the number of additional voice calls for arbitrary TXOP setting ($\eta > 1$) , denoted as Γ_η, can be approximated by

$$\Gamma_\eta = \left(\frac{f(\eta = 1)}{\eta}\right)/2. \qquad (24)$$

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

$$\tilde{f}(\eta) = \tilde{f}(\eta - 1) + \Gamma_\eta. \qquad (25)$$

In Figs. 15 and 16 we compare approximation results with results obtained by the analytical model for the G.729 and G.711 codecs using a variety of parameters. Observe that the approximation results match those of the analytical model closely even with large TXOP values (e.g. $\eta = 10$) despite the fact that the approximation may be too optimistic at those values. It is optimistic because the approximation is based on the argument that new calls can be initiated by increasing the TXOP value at the AP as long as the average rate of successful transmissions on the channel remains the same. This argument, however, ignores that some transmissions (i.e. those originated from the AP) are now involved multiple packets and therefore capture the channel much longer. On the other hand, further increasing η beyond the above value carries no benefit because of the bottleneck shift as discussed in the previous section. Thus the results in Figs. 15 and 16 show that the approximation is reasonably accurate in terms of the voice capacity for the range of the TXOP values that are of interest. The approximation formula provides a simple alternative to the complete analytical model that can be used to estimate the voice capacity attainable in WLAN. Note that the analytical results have been validated earlier in Sec. IV and thus simulation results are omitted in Figs. 15 and 16.

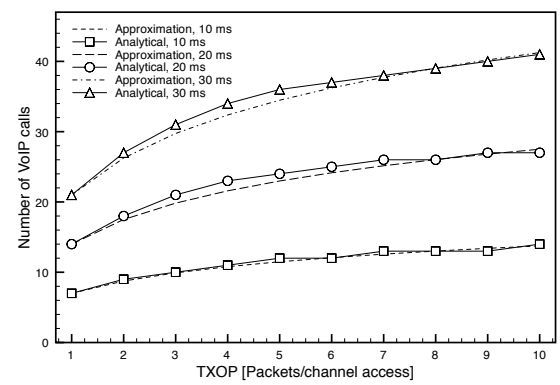

Fig. 15. Comparison of the number of G.729 voice calls obtained by (22) and (25) for different sampling rates and increasing TXOP parameter

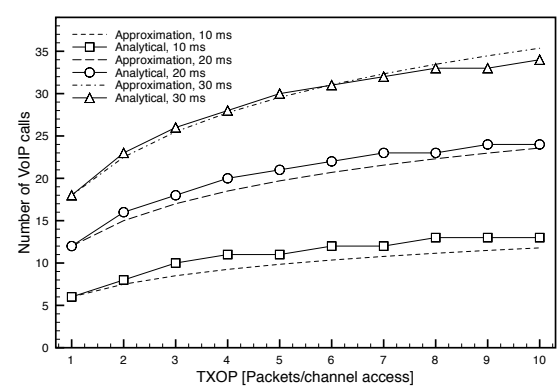

Fig. 16. Comparison of the number of G.711 voice calls obtained by (22) and (25) for different sampling rates and increasing TXOP parameter

The approximation formula in (25) also allows us to gain further insights into the voice capacity in WLAN and to study its relation to MAC parameters such as the TXOP parameter. In particular, from (24) and (25) it can be seen that the voice capacity is depending on the TXOP value and the initial voice capacity obtained using default parameter settings

TABLE V
COMPARISON OF OUR APPROACH TO EXISTING MODELS IN THE LITERATURE

	Analytical model	Simulation	Improved voice capacity using TXOP	Delay analysis	Impact of buffer size on voice capacity	Closed form expression for voice capacity	Voice capacity approximation
Shin et al [1]	-	-	-	√	-	-	-
Cai et al. [2]	√	√	-	√	-	-	-
Hedge et al. [3]	√	-	-	√	-	-	-
Dangerfield et al. [10]	-	-	√	√	√	-	-
Kawata et al. [24]	√	√	-	√	-	√	-
Harsha et al. [25]	√	√	-	-	-	-	-
Gao et al. [26]	√	-	√	√	-	-	-
Our analysis	√	√	√	√	√	√	√

(i.e. TXOP = 1). Also Γ_η in (25) tends to zero with an increasing value of TXOP parameter and thus $\tilde{f}(\eta) \approx \tilde{f}(\eta - 1)$ as TXOP $\rightarrow \infty$. This implies that the asymptotic value described in Sec. IV (Fig. 7) for the maximum number of voice calls still exists even though the AP's buffer is now infinite. And varying the buffer at the wireless nodes will only affect the shift in bottleneck but not the overall achievable voice capacity in the network. Furthermore, it can be observed in (23) that no additional packets can be gained on the downlink as TXOP value exceeds C_1 indicating that the AP buffer is no longer a bottleneck as described in Section IV. In summary, using TXOP significantly improves the voice capacity in WLANs but even then the capacity is limited. We show that once the capacity limit is reached, further increasing TXOP will not increase the VoIP capacity. We have also confirmed the voice capacity limit by simulation and showed that the maximum number of voice calls can be achieved when $\eta = f(\eta = 1)$. Increasing TXOP beyond this optimal value results in excessive packets loss and long delay at a wireless node. The approximation results also show that assumptions made in [21] that TXOP should be equal to the number of wireless nodes will not hold in general, specifically when $N > f(\eta = 1)$.

Finally in Table V we summarize and highlight the differences between our approach and other existing models in the literature.

VII. CONCLUSION

In this paper we have developed a detailed analytical model to evaluate the performance gain in terms of voice capacity 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 without any changes to hardware or protocol. We demonstrated that there exists an optimal TXOP value beyond which the voice capacity cannot be improved any further. We validated our model using extensive simulations and showed that any TXOP value greater than the optimal value causes the bottleneck to shift from the AP to the wireless nodes. Furthermore we have investigated the impact of the AP buffer on the maximum number of voice calls that the WLAN can accommodate. For a given TXOP value, we observed that there exists a minimum buffer size (K_{min}) where this capacity is reached. We showed that the voice capacity of a WLAN can either be calculated using an $M/G/1/K$ ($K \geq K_{min}$) or an $M/G/1/\infty$ queuing models where a closed form expression for a number of attainable voice calls was obtained using the latter. More importantly, we developed a simple yet accurate recursive approximation formula that provides the achievable voice capacity in WLAN for a given TXOP parameter. The approximation provides a simple way to determine the optimal TXOP value and can be used as a guideline for network design and voice admission control in WLAN.

ACKNOWLEDGEMENT

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