Multimedia Capacity Analysis of the IEEE 802.11e Contention-based Infrastructure Basic Service Set †

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***Abstract*—We propose a novel and simple multimedia capacity analysis framework for the IEEE 802.11e Enhanced Distributed Channel Access (EDCA) function. We design a simple model-based admission control scheme which is based on the accurate queue utilization ratio we calculate by appropriately weighing the service time predictions of our previously developed saturation cycle time model for different number of active stations. We show that the proposed call admission control algorithm maintains satisfactory user-perceived quality for coexisting voice and video connections in an infrastructure Basic Service Set (BSS) and does not present over- or under-admission problems of previously proposed models in the literature.**

I. Introduction

The IEEE 802.11e standard [1] specifies the Hybrid Coordination Function (HCF) which enables prioritized and parameterized Quality-of-Service (QoS) services at the MAC layer. The HCF combines a distributed contention-based channel access mechanism, referred to as Enhanced Distributed Channel Access (EDCA), and a centralized polling-based channel access mechanism, referred to as HCF Controlled Channel Access (HCCA). In this paper, we confine our analysis to the EDCA scheme, which uses Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) and slotted Binary Exponential Backoff (BEB) mechanism as the basic access method. The EDCA defines multiple Access Categories (ACs) with AC-specific Contention Window (CW) sizes, Arbitration Interframe Space (AIFS) values, and Transmit Opportunity (TXOP) limits to support QoS [1].

Due to its contention-based nature, EDCA cannot provide parameterized QoS for real-time applications that require strict QoS guarantees, if the network load and parameters are not tuned such that the network is operating in nonsaturated state [2],[3]. The use of an admission control algorithm is recommended in [1] to limit the network load for QoS provisioning although one is not explicitly specified. A loose capacity estimation is harmful for admission control, since the quality of ongoing flows will be jeopardized. Conversely, an underestimation of the network capacity results in a fewer number of admitted flows than the network can support.

In this paper, we design a novel, simple, and accurate analytical framework which directly employs the results of our previously developed simple saturation analysis to estimate the network capacity for multimedia traffic. An approximate station- and AC-specific average service time is calculated by weighing the average service time calculated for different number of active stations in saturation (i.e., every AC has always a packet ready for transmission and is actively contending for access). Given the average station- and AC-specific traffic load, the average service time is directly translated into a station- and AC-specific queue utilization ratio (note that since all the measures are station- and AC-specific, the proposed framework considers the potential unbalanced traffic load in the 802.11e infrastructure Basic Service Set (BSS) uplink and downlink). Next, we design a novel centralized EDCA admission control algorithm the admission decisions of which are based on the estimated queue utilization ratio. The key point is that the delay guarantee of real-time applications is only possible when the queue utilization ratio of active multimedia flows is smaller than 1 (i.e., the queue is stable).

It is important to note that our approach in this paper in estimating the multimedia capacity is employing the results of a relatively simpler saturation analysis in a novel and simple framework rather than designing a relatively complex nonsaturation model. Comparing the theoretical results with simulations, we show that the proposed call admission control algorithm maintains satisfactory user-perceived quality for coexisting voice and video connections in an infrastructure BSS by limiting the maximum number of admitted flows of each multimedia traffic type. Comparison with extensive simulation results also reveals that the proposed analysis does not result in an overestimation or a significant underestimation of the network capacity.

II. Background

In this section, we first give a brief overview of the EDCA function. Next, we provide the related literature on capacity analysis and admission control for EDCA. Finally, we summarize the main points about the EDCA cycle time analysis for saturation, and refer to the related work for details.

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A. EDCA Overview

The IEEE 802.11e EDCA is a QoS extension of IEEE 802.11 Distributed Coordination Function (DCF) [1], [4]. The major enhancement to support QoS is that EDCA differentiates packets using different priorities and maps them to specific ACs that are buffered in separate queues at a station. Each AC_i within a station ($0 \leq i \leq i_{max}$, $i_{max} = 3$ in [1]) having its own EDCA parameters contends for the channel independently of the others. Following the convention of [1], the larger the index i is, the higher the priority of the AC is. Levels of services are provided through different assignments of the AC-specific EDCA parameters; AIFS, CW, and TXOP limits.

If there is a packet ready for transmission in the MAC queue of AC_i, the EDCA function must sense the channel to be idle for a complete AIFS_i before it can start the transmission. If the channel is detected to be busy at the packet arrival or during AIFS_i, a backoff procedure is completed following the completion of AIFS before the transmission of this packet. A uniformly distributed random integer, namely a backoff value, is selected from the range $[0, W_i]$. The backoff counter is decremented at the slot boundary if the previous time slot is idle. Should the channel be sensed busy at any time slot during AIFS_i or backoff, the backoff procedure is suspended at the current backoff value. The backoff resumes as soon as the channel is sensed to be idle for AIFS_i again. When the backoff counter reaches zero, the packet is transmitted.

The value of W_i depends on the number of retransmissions the current packet experienced. The initial value of W_i is set to the AC-specific $CW_{min,i}$. If the transmitter cannot receive an Acknowledgment (ACK) packet from the receiver in a timeout interval, the transmission is labeled as unsuccessful and the packet is scheduled for retransmission. At each unsuccessful transmission, the value of W_i is doubled until the maximum AC-specific $CW_{max,i}$ limit is reached. The value of W_i is reset to the AC-specific $CW_{min,i}$ if the transmission is successful, or the retry limit is reached thus the packet is dropped.

The higher priority ACs are assigned smaller AIFS. Therefore, the higher priority ACs can either transmit or decrement their backoff counters while lower priority ACs are still waiting in AIFS. This results in higher priority ACs enjoying a relatively faster progress through backoff slots. Moreover, the ACs with higher priority may select backoff values from a comparably smaller CW range (a smaller CW value means a smaller backoff delay).

An internal (virtual) collision within a station is handled by granting the access to the AC with the highest priority. The ACs with lower priority run the collision procedure as if an outside collision has occured [1].

B. Related Work

Assuming constant collision probability for each station, Bianchi [5] developed a simple Discrete-Time Markov Chain (DTMC) and the saturation throughput is obtained by applying regenerative analysis to a generic slot time. This analysis is extended by several researchers to enable capacity analysis of the DCF or the EDCA in nonsaturation [6]–[9]. Such approaches result in complex Markov analysis. A number of queueing models have also been proposed to analyze delay performance of a station or an AC under the assumption that the traffic is uniformly distributed [2], [3], [10]–[12].

There are also studies on capacity analysis and admission control considering the infrastructure BSS where the Access Point (AP) usually has a higher load in the downlink than the stations serving traffic in the uplink. A group of studies mainly concentrated on capacity analysis of only Voice-over-IP traffic for DCF and did not consider traffic differentiation [13]–[18]. Gao *et al.* [19] and Cheng *et al.* [20] calculated VoIP capacity of the WLAN when CW differentiation among uplink and downlink flows are used. A Markov renewal framework for the scenarios where the downlink is always the bottleneck is proposed in [21].

Another group of studies defines parameter adaptation algorithms for QoS enhancement and defines measurement-assisted call admission control algorithms [22]–[26].

C. EDCA Cycle Time Analysis

A random access system is known to exhibit cyclic behavior. A cycle time is defined as the duration in which an arbitrary tagged user successfully transmits one packet on average. We have previously derived the explicit mathematical expression for the AC-specific EDCA cycle time in saturation [27]. Our derivation considers the AIFS and CW differentiation by employing a simple average collision probability analysis. The cycle time analysis is used to predict the saturation throughput, the average service time, and the packet loss probability.

The performance of EDCA depends on the number of active ACs within the same station as well as the number of active ACs at the other stations due to the fact that the EDCA function acts differently in the case of an internal or an external collision. In [28], we extend our cycle time model in [27] by considering both the possibility of a station running multiple ACs (thus the possibility of internal collisions) and the possibility of the number of active ACs varying from station to station. For example, consider a simple WLAN scenario where an AP runs 2 downlink ACs, namely AC_1 and AC_2. Similarly, assume n_1 stations ($STA_1, \ldots, STA_{n_1}$, $n_1 > 0$) only run AC_1 and n_2 other stations ($STA_{n_1+1}, \ldots, STA_{n_1+n_2}$, $n_2 > 0$) only have AC_2 in the uplink. Although there are 2 distinct ACs active in the system, the downlink AC_i and the uplink AC_i ($i = \{1, 2\}$ for the running example) is not expected to have the exact same performance due to internal and external collision differentiation [1]. In this case, the performance analysis should be carried out individually for 4 different Traffic Classes (TCs) which are uplink AC_1, downlink AC_1, uplink AC_2, and downlink AC_2. Therefore, each TC is distinguished by the other active ACs within a station. Following the same method as we propose in [27], a generalized formulation for the EDCA cycle time is derived for an arbitrary distribution of active number of ACs [28].

Let $E_j[t_{cyc}]$ be the average cycle time (renewal cycle) for a TC_j user. Then, the average protocol service time is

$$E_j[t_{srv}] = (1 - p_{j,drop})E_j[t_{cyc}] \tag{1}$$

where $p_{j,drop}$ is the average packet drop probability for a TC_j user [28]. Due to space limitations, we cannot provide the generalized cycle time formulation in this paper. The reader is referred to [28] for details.

Although we use our cycle time model to calculate the service time parameters necessary for the proposed capacity analysis in Section III, we also note that any other accurate saturation analysis method provided in the literature should be applicable. We stick with our cycle time model since *i)* it incorporates accurate AIFS and CW differentiation, *ii)* it considers the possibility of the number of active ACs varying from station to station, and *iii)* in the meantime, it is simple.

III. Multimedia Capacity Analysis for 802.11e Infrastructure BSS

We propose a novel framework where we calculate TC-specific average frame service rate μ via a weighted summation of saturation service rate $E[t_{srv}]$ over varying number of active stations. Defining a TC-specific average queue utilization ratio ρ, we design a simple call admission control algorithm which limits the number of admitted real-time multimedia flows in the 802.11e infrastructure BSS in order to prevent the corresponding TC queues going into saturation. Admitted real-time multimedia flows can be served with QoS guarantees, since low transmission delay and packet loss rate can be maintained when the 802.11e WLAN is in nonsaturation [3],[28],[29]. Comparing with simulation results, we show that not only does the proposed admission control algorithm prevent the so-called over-admission or under-admission problems but also efficiently utilizes the network capacity.

A. TC-specific Average Queue Utilization Ratio

Each station runs a QoS reservation procedure with the AP for all of its traffic streams that need parameterized QoS support. The Station Management Entity (SME) at the AP decides whether the Traffic Stream (TS) is admitted or not regarding the Traffic Specification (TSPEC) in the Add Traffic Stream (ADDTS) request provided by the station. The TSPEC specifies the traffic parameters such as the mean/peak data rate (R/R_{peak}), the mean/maximum packet size(L/L_{max}), the Delay Bound (DB), the physical layer rate, the User Priority (UP), etc [1].

Let average frame service rate for TC$_j$ be denoted as μ_j. Also let the average packet arrival rate for TC$_j$ be denoted as λ_j. For simplicity, in the sequel, we assume that TCs at different stations are running TSs with equal TSPEC values (so all traffic parameters remain TC-specific). Though all work in this section can be generalized for varying traffic load and parameters within a TC at different stations, we opted not to present this out-of-scope generalization since it would make the model more difficult to understand.

We define TC-specific average queue utilization ratio ρ as

$$\rho_j = \lambda_j/\mu_j, \ \forall j. \tag{2}$$

B. TC-specific Average Frame Service Rate

The TC-specific average queue utilization ratio ρ_j shows the percentage of time that TC$_j$ has a frame in service on average. In other words, ρ_j is the probability that TC$_j$ is active. Our novel approach in calculating μ_j is performing a weighted summation of $E_j[t_{srv}]$ for varying number of active TCs.

Let $P^j_{TC_0,TC_1,...,TC_j,...,TC_{J-1}}(f'_0, f'_1, ..., f'_j, ..., f'_{J-1})$ denote the joint conditional probability that f'_j stations using TC$_j$, $\forall j$, are active given that one TC$_j$ has a frame in service and when the total number of TCs is J. Also, let $E_j[t_{srv}(f'_0, f'_1, ..., f'_{J-1})]$ denote the average service time when f'_j stations using TC$_j$, $\forall j$, are active. We use the proposed cycle time model in [28] to calculate $E_j[t_{srv}(f'_0, f'_1, ..., f'_{J-1})]$. Then, the TC-specific average frame service rate μ_j is calculated as in (3).

We noticed that the distribution of the number of active TCs approximates the sum of independent Binomial distributions with parameters $f_{j'}$ and $\rho_{j'}$, $\forall j'$ as in (4) for the traffic models we used in this study. We confirm the validity of (4) via comparing the analytical estimations with simulation results in Section IV. On the other hand, we do not argue that the binomial activity distribution holds for any type of traffic model in any scenario. Our observation is that for widely used voice and video traffic models this approximation works well. The proposed framework is generic in the sense that any other activity distribution profile may be used to incorporate other traffic models in other network scenarios.

Note that when $\sum_{j'=0}^{J-1} f'_{j'} = 1$, i.e., there is only one active TC, the cycle time model in [28] is not applicable in calculating $E_j[t_{srv}(f'_0, f'_1, ..., f'_{J-1})]$. On the other hand, the cycle time calculation in this case is straightforward. Since no collisions can occur and no other station is active, the successful transmission is done at AIFS completion. Therefore, $E_j[t_{srv}(f'_0, f'_1, ..., f'_{J-1})] = T_{s_j}$ if $\sum_{j'=0}^{J-1} f'_{j'} = 1$ (T_{s_j} is the time spent in a successful transmission for TC$_j$ [28]).

The fixed-point equations (2)-(4) can numerically be solved for ρ_j and μ_j, $\forall j$.

C. Admission Control Procedure

Upon receiving the ADDTS request, the AP associates the TS with the AC and the TC using the value in the UP field of the TSPEC and the station MAC address. The traffic stream is admitted if and only if the following tests succeed

$$\rho_j < 1, \ \forall j. \tag{6}$$

The simple tests in (6) ensure that the average traffic arrival rate to all TCs is smaller than the average service rate that can be provided to them. Therefore, the MAC queues of all TCs can be considered to be stable (all TCs remain in nonsaturation) on average.

When a real-time flow ends, the source node transmits a Delete Traffic Stream (DELTS) request for the TS. The AP deletes the corresponding entry from the list of admitted flows.

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$$\frac{1}{\mu_j} = \sum_{0 \le f'_0 \le f_0} \cdots \sum_{1 \le f'_j \le f_j} \cdots \sum_{0 \le f'_{J-1} \le f_{J-1}} E_j[t_{srv}(f'_0, ..., f'_j, ..., f'_{J-1})] \cdot P^j_{TC_0,...,TC_j,...,TC_{J-1}}(f'_0, ..., f'_j, ..., f'_{J-1}) \quad (3)$$

$$P^j_{TC_0,TC_1,...,TC_j,...,TC_{J-1}}(f'_0, f'_1, ..., f'_j, ..., f'_{J-1}) = \binom{f_j - 1}{f'_j - 1} \rho_j^{f'_j - 1}(1-\rho_j)^{f_j - f'_j} \prod_{\forall j': j' \neq j} \binom{f_{j'}}{f'_{j'}} \rho_{j'}^{f'_{j'}} (1-\rho_{j'})^{f_{j'} - f'_{j'}} \quad (4)$$

$$\frac{1}{\mu_j} = \sum_{0 \le f'_1 \le f_1} \cdots \sum_{1 \le f'_j \le f_j} \cdots \sum_{0 \le f'_{J'} \le f_{J'}} E_j[t_{srv}(f'_1, ..., f'_j, ..., f'_{J'-1}, f_{J'}, ..., f_{J-1})] \cdot P^j_{TC_1,...,TC_{J'}}(f'_1, ..., f'_{J'}) \quad (5)$$

A few remarks on admission control and capacity analysis are as follows.

- The proposed capacity analysis and admission control can easily be extended for the case of some TCs running best-effort traffic. To do a worst-case analysis, the TCs that run best-effort traffic can be assumed to be always active. This generalizes (3) as in 5 where J' and $J - J'$ are the number of TCs that run multimedia and best-effort flows respectively. In this case, the tests in (6) are done only for $0 \le j \le J'$.
- Our capacity analysis and admission control scheme is based on mean values and does not consider the worst-case scenario (when all the admitted Variable Bit Rate (VBR) multimedia traffic may transmit at their peak rate, R_{peak}). To limit the channel utilization by multimedia flows thereby leaving room to accommodate bandwidth fluctuations caused by VBR traffic, the tests in (6) can be modified such that

$$\rho_j < \rho_{th}, \ \forall j \quad (7)$$

  where $\rho_{th} < 1$. A good value for admission threshold ρ_{th} can be decided based on R_{peak}/R ratios. Alternatively, R_{peak} may be used in (2) leaving the test requirement as in (6). The proposed framework is pretty flexible for such tweaks.
- The TSPEC of a TS may also specify a delay bound DB which denotes the maximum amount of time allowed to transport the frames across the wireless interface including the queueing delay. As also shown in [3, Table I], multimedia services should satisfy QoS requirements in terms of one-way transmission delay, delay variation, and packet loss rate. For example, for VoIP and videoconferencing, the excellent (acceptable) quality is satisfied if one-way transmission delay is smaller than 150 ms (400 ms) and the packet loss rate is smaller than $1\% - 3\%$. Note that packet loss rate includes the dropped packets at the playout buffer of the receiver when the packets are not received within the delay bound. Our capacity analysis does not explicitly consider these metrics in admission control. On the other hand, the proposed call admission control algorithm makes the multimedia TC queues remain stable (TC queues do not go into saturation) by limiting the number of admitted real-time flows. This provides low transmission delays and packet loss ratio due to the low collision probability in nonsaturation (as also confirmed via simulations).

IV. Analytical and Simulation Results

We consider an infrastructure BSS. All the stations including the AP use the following EDCA parameters (suggested parameters in [1]); $AIFSN_2 = 2$, $AIFSN_3 = 2$, $CW_{2,min} = 15$, $CW_{3,min} = 7$, $m_1 = m_3 = 1$, $r_1 = r_3 = 7$. The simulations consider two types of traffic sources; voice and video. The voice traffic model implements G.711 or G.729 VoIP application as Constant Bit Rate (CBR) traffic. The CBR traffic model is used for two reasons; *i)* it provides a worst-case upper bound for the case when the traffic presents on-off traffic characteristics (silence suppression) and *ii)* this enables comparison of voice capacity results with the models proposed in [13]–[19]. The voice codec parameters are selected as in [18]. For the video source models, we use the traces of real MPEG-4 video streams [30]. For the specific MPEG-4 codec used, the average payload is 821 bytes, and the codec bit rate is 174 kbps. The voice traffic uses AC_3 and the video traffic uses AC_2. The simulation results are reported for the wireless channel which is assumed to be not prone to any errors. The extension of the analysis for the errored channel is left for future study. All the stations have 802.11g Physical Layer (PHY) using 54 Mbps and 6 Mbps as the data and basic rate respectively.

We use a network topology in which a voice/video connection is initiated between a distinct party in the Internet and the WLAN. The traffic is relayed at the AP from (to) the wireless channel to (from) the wired link. The wired link delay is set to 20 ms for all connections.

1) Voice Capacity Analysis: In the first set of experiments, we investigate the VoIP capacity of 802.11e WLAN when no other type of traffic coexists. A two-way voice connection is established every ω ms, with the starting time randomly chosen over $[0, \omega]$ ms. We set ω equal to the packet interval duration of the voice codec used. Table I tabulates the maximum number of VoIP connections for different codecs. In the simulations, the maximum number of voice connections is obtained in such a way that one more connection results in the delay outage ratio (the ratio of the number of audio packets with end-to-end delay exceeding 150 ms over the total number of packets transmitted) larger than 1% [18]. As shown in Table I, the analytical results for the proposed model and the simulation results closely follow each other. The obtained analytical

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TABLE I
COMPARISON OF THE MAXIMUM NUMBER OF VOIP CONNECTIONS

Audio (ms)	G.711		G.729	
	Analysis/Simulation	[18]	Analysis/Simulation	[18]
10 ms	27/27	21	29/29	22
20 ms	49/49	38	56/56	43
30 ms	70/70	53	85/85	65
40 ms	87/87	67	112/112	85
50 ms	102/102	79	139/139	106
60 ms	115/115	89	166/166	128

TABLE II
COMPARISON OF THE MAXIMUM NUMBER OF VIDEO CONNECTIONS (ANALYSIS/SIMULATION) WHEN VOIP FLOWS COEXIST.

		Number of existing two-way G.711 (20 ms) connections					
		5	10	15	20	25	30
Number of admitted MPEG-4 flows	Downlink	109/110	98/100	87/88	76/78	64/65	52/54
	Uplink	88/89	67/69	57/57	48/48	37/37	28/28
	Two-way	54/55	46/47	41/41	34/34	28/28	19/19

bounds are much tighter when compared to the results obtained using the model in [18]. As the comparison of the performance with 802.11e MAC parameters in 802.11g PHY layer indicates the model in [18]* has significant under-admission problems for an arbitrary selection of MAC parameters (especially when the CW settings are small and the underlying PHY is 802.11g). We do not provide any comparison with [15]–[17] the over-admission problems of which are already shown in [18].

Fig. 1 show the probability density function (pdf) of active number of TCs given that the TC at the AP or at the non-AP station (denoted as STA in the figure) is active in a scenario consisting of only VoIP connections (G.711 VoIP codec with 10 ms packet intervals). As previously stated, we analytically calculate $P^{j}_{TC_0,TC_1,...,TC_j,...,TC_{J-1}}(f'_0, f'_1, ..., f'_j, ..., f'_{J-1})$ by assuming that the distribution of the number of active TCs approximates a Binomial distribution with parameters f_j and ρ_j. The comparison in Fig. 1 presents that the pdf of analytical calculation closely follows the pdf obtained through simulation. Although the results are not presented here due to space limitations, a similar discussion holds for other codecs with different packet interval values. Note that corresponding activity distribution profiles also depend on when the CBR flows start. Actually, it is shown in [3] that the capacity of the WLAN can be increased by randomizing the flow start times. The pdf results for simulations are obtained through averaging over several simulation runs with different random number generator seeds and randomized flow start times. Our capacity estimation results show that our assumption on the activity distribution is suitable in the case of practical traffic models we use for voice and video flows.

2) Voice and Video Capacity Analysis: In the second set of experiments, we investigate the capacity of 802.11e WLAN when both voice and video traffic coexist (using different ACs). Note that the analytical models of [14]–[21] do not provide such analysis capability. Table II shows the number of admitted uplink, downlink, and two-way MPEG-4 flows for increasing the number of VoIP connections. In this scenario, we use the G.711 codec with a 20 ms sample interval. In the simulations, the number of maximum video connections is obtained in such a way that one more connection results in delay outage ratio larger than 1%. As the results indicate, the analytical and simulation results closely follow each other. Such a comparison reveals that the proposed capacity prediction and admission control scheme is also effective when different classes of multimedia traffic coexist in the BSS.

V. CONCLUSION

We have designed a practical and simple multimedia capacity prediction and admission control algorithm to limit the number of admitted real-time multimedia flows in the 802.11e infrastructure BSS. Motivated by the previous findings in the literature such that the contention-based 802.11 MAC can achieve high throughput and low delay in nonsaturation, the proposed admission control algorithm is based on simple tests on station- and AC-specific queue utilization ratio estimates. Our novel approach is the calculation of the queue utilization ratio by weighing the average service time predictions of our previously developed cycle time saturation model among varying number of active stations. The proposed simple framework is effective in capacity estimation even in the case of coexisting voice and video flows using different ACs with arbitrarily selected MAC parameters. One of the key insights provided by this study is the accuracy of the proposed approximate

*Our implementation of the model in [18] duplicates the analytical results in [18] which are for a specific DCF MAC parameter set. Although the results are not provided in this paper, the analytical results for the proposed model and our simulation results also confirm the capacity prediction of [18] for the specific DCF scenario.

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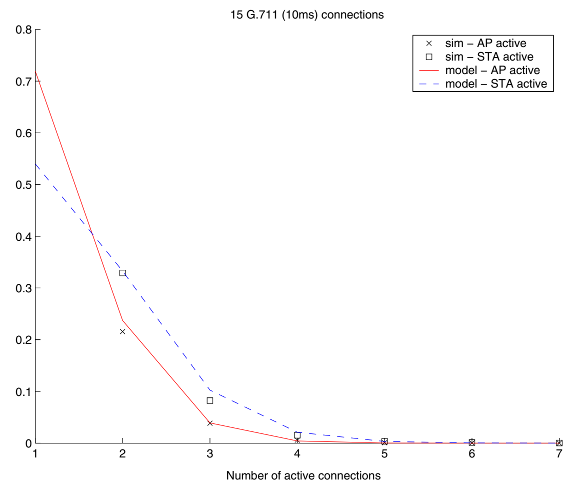
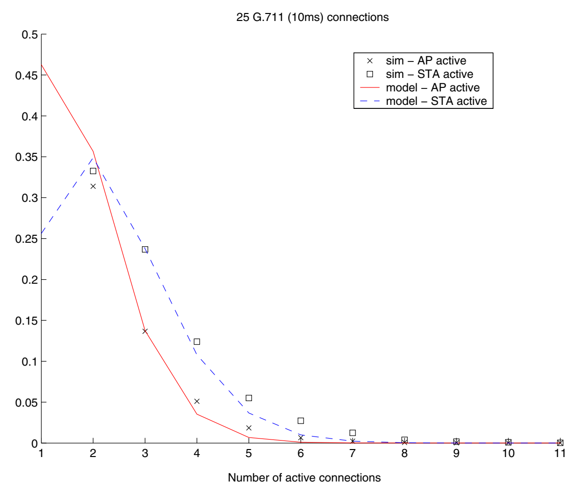


(a)



(b)

Fig. 1. The pdf of active number of TCs given that the TC at the AP or at the station (denoted as STA) is active in a scenario consisting of G.711 VoIP connections (10 ms packet intervals). (a) 15 connections. (b) 25 connections. Note that the figures do not present the whole x-axis (activity profile for large number of stations) for better clarity on the comparison of simulation and analysis (especially when the activity probability is not close to zero).

capacity estimation framework that uses the results of a relatively simpler saturation analysis rather than designing a more complex and hard to implement nonsaturation model.

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