Analytical Model for an IEEE 802.11 WLAN using DCF with Two Types of VoIP Calls

Sri Harsha, Anurag Kumar and Vinod Sharma
Department of Electrical Communication Engineering,
Indian Institute of Science, Bangalore, 560012.
Email: harshas, anurag, vinod@ece.iisc.ernet.in

Abstract — **We formulate an analytical model for capacity evaluation of an infrastructure IEEE 802.11 based network carrying full-duplex packet telephone calls when different types of codecs are used for voice calls. The analysis of the model utilizes the attempt rate results from a well known fixed-point based saturation analysis. The performance estimates obtained match very well with ns2 simulations.**

The network comprises wireless STAs establishing voice calls with wired STAs on a high speed local area network to which the access point is connected. We consider packet voice calls from two different types of coders, with N_1 calls of Type 1 and N_2 calls of Type 2, from a high speed local area network, terminating at N_1 and N_2 wireless STAs through an AP. We model the number of STAs that have an up-link voice packet as a Markov renewal process embedded at so called channel slot boundaries. Analysis of the evolution over the channel slot is done using saturation analysis. We find that the AP is the bottleneck, and the system can support (in the sense of a bound on the probability of delay exceeding a given value) a number of calls less than that at which the arrival rate into the AP exceeds the average service rate applied to a saturated AP.

I. Introduction

The convenience of tetherless high speed Internet access has resulted in rapid growth of IEEE 802.11 WLANs [3] in enterprises, campuses, public buildings and houses. The period in which WLAN installations have grown has also seen the remarkable success of packet voice telephony or VoIP (Voice over IP). Packet telephony over the Internet has now become a standard offering over all forms of wired access: ethernet, DSL or packet cable. WLANs however were originally designed for carrying bursty data services. Yet, since they are so ubiquitous, the need for carrying voice services on them has begun to be realized. This has given rise to much interest in the capacity of WLANs for carrying packet voice calls. In this paper we extend the analytical model developed in [6] to determine voice calls capacity of an 802.11b WLAN in a new situation. Our ongoing and further work will extend this analysis to 802.11e WLANs, that have QoS support for real time services.

As in [6] IEEE 802.11 stations (STAs) access a high speed local area network via an access point (AP). Our analysis yields answer to the question: "When two different types of codecs are used, how many packet telephone calls can be set up to different STAs such that voice call QoS is met?" As in [6] we take the QoS objective to be "the probability of packet delay over the WLAN exceeds (say) 20 ms with probability no more than 1%".

We consider voice calls with two types of codecs. Type 1 voice calls have a larger packet size than Type 2 calls. Type 1 calls use the G.711 codec that generates a voice packet of 160 bytes every 20 ms. To this we add the RTP+UDP+IP header of 40 bytes (without using header compression). Therefore, we model the voice traffic as generating 200 (= 40 + 160) bytes per 20 ms. Similarly, we consider the Type 2 calls that use G.729 codec. We model this voice traffic as generating 60 (=20 + 40) bytes per 20 ms. We obtain an analytical approximation for the number of calls of each type that can be admitted so that QoS is met. The analytical modeling we provide in our work helps in a deeper understanding of the "physics" of the system and will also be useful in designing on-line admission control algorithms.

Related Literature: The modeling of IEEE 802.11 DCF has been a research focus since the standard has been proposed. Many studies are focussed on saturation analysis of TCP and there are only a few attempts to characterize the 802.11 MAC protocol behavior when subjected to voice traffic. Analytical performance modeling of packet voice telephony to estimate the call capacity over 802.11 WLANs has been done in [1], [2], [7] and [6]. While [1], [2] and [7] involve approximations, [6] models the behaviour more accurately. In our work we extend the model developed in [6]. The studies have considered only a single type of voice calls. In practical environment it is natural to expect calls originating from more than one type of codecs. Our approach for two types of codecs, discussed in this paper, can be easily extended to more than 2 types of codecs also.

We identify an embedded Markov chain which we study to obtain the parameters of interest. The MAC protocol (CSMA/CA) employed in 802.11 DCF is complicated and does not really lead to a Markov system. But we replace it with a system where each station transmits its packet (if it has one) in every slot with a probability that depends only on the number of stations contending for the channel at that time. This attempt probability is approximated using the saturation analysis in [5]. The intervals between the instants at which Markov chain is embedded are random, but together these constitute a Markov renewal process. We will see that the resulting stochastic model provides a good approximation

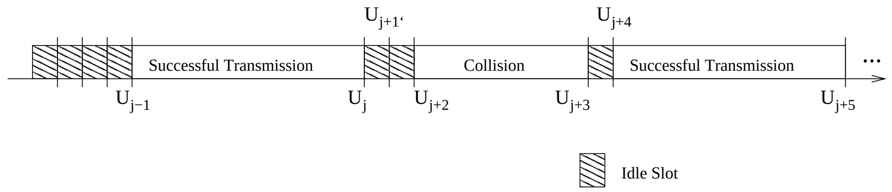

Fig. 1. An evolution of the back-offs and channel activity. $U_j,\ j \in 0, 1, 2, 3, \ldots$ is the instant where the j^{th} channel slot ends.

Parameter	Symbol	Value
PHY data rate	C_d	11 Mbps
Control rate	C_c	2 Mbps
G711 packet size	L_{voice1}	200 Bytes
G729 packet size	L_{voice2}	60 Bytes
MAC - layer ACK Packet Size	L_{ACK}	112 bits
MAC Header size	L_{MAC}	272 bits
PLCP preamble time	T_P	$144\mu s$
PHY Header time	T_{PHY}	$48\mu s$
DIFS Time	T_{DIFS}	$50\mu s$
SIFS Time	T_{SIFS}	$10\mu s$
EIFS Time	T_{EIFS}	$364\mu s$
Min. Contention Window	CW_{min}	31
Max. Contention Window	CW_{max}	1023

Tab. 1: Various parameters used in analysis and simulation, using IEEE 802.11b

to the actual system. We consider IEEE 802.11b WLAN at 11Mbps to show a comparison of analysis and simulation results.

II. Overview of the IEEE 802.11 DCF

In this section, we briefly summarize the key features of the IEEE 802.11 standard which are relevant to our purpose. A complete specification can be found in [3]. In IEEE 802.11 based wireless systems, traffic originates and terminates at stations (STAs) or access points (APs). Data transfer is possible by a two-way handshake of DATA-ACK exchange called the *Basic Access* mechanism, or a four-way handshake of RTS-CTS-DATA-ACK exchange called the *RTS/CTS* mechanism.

We assume that the basic access mechanism is used for voice calls. This is due to a small size of the packets involved. A transmitting STA infers a collision if either a packet is not received correctly or an ACK frame is not received correctly within the $ACKTimeout$. After each unsuccessful attempt (either due to collision or transmission errors), a retransmission attempt is scheduled with non decreasing mean back off, upto a specified number of times called the *retry count*. The retry count depends upon the physical (PHY) layer being used. When the number of unsuccessful attempts exceeds the retry count, the packet is dropped and transmission of the next head of the line (HOL) packet is scheduled.

After an erroneous frame is received (either due to collisions, transmission errors or insufficient power), a STA must defer channel access at least for a duration called Extended Inter Frame Space (EIFS). The EIFS interval begins when the PHY indicates a medium IDLE condition at the end of the transmission of the erroneous frame. The value of EIFS is defined in the IEEE 802.11 standard as

$$T_{EIFS} := T_{SIFS} + T_{ACK} + T_{DIFS}.$$

where T_{ACK} is the time required for the transmission of an ACK frame, T_{SIFS} and T_{DIFS} are different inter frame spaces (IFS) defined to provide priority levels of access to the wireless media. IFSs for IEEE 802.11b are given in Table 1 and for details see [4].

III. Model for 2 Types of Voice Codecs

A. Modeling Assumptions

Packets arrive at the STAs every 20 ms. As a QoS requirement we demand that the probability that a packet is transmitted successfully within 20 ms is greater than 0.99. Since the packets will experience delays in the rest of the network also, this is a reasonable target to achieve. Then, if the target is met, whenever a new packet arrives at an STA, it will find the queue empty. Thus the following two assumptions will be acceptable in the region where we want to operate: (1) the buffer of every STA has a queue length of at most one packet, and (2) new packets arriving to the STAs see empty queues. The latter assumption implies that if there are k STAs with voice packets then a new voice packet arrival comes to a $(k+1)^{th}$ STA. Since the AP handles packets from $N\ (= N_1 + N_2)$ streams we expect that it is the bottleneck and we assume that it will contend at all times. This is a realistic assumption near the system capacity.

As mentioned earlier, packets arrive every 20 ms in every stream. To simplify our analysis we assume that the arrival process at each node is Bernoulli with rate λ per system slot. The value of λ can be calculated as follows. Each system slot in 802.11b is of $20\mu s$ duration. Thus in 1000 system slots there is one arrival. Therefore, on matching the arrival rate per slot we obtain $\lambda = 0.001$.

B. Stochastic Modeling

The evolution of the back-offs and channel activity in the network is as in [6]. We allow a voice call to use one of the two coders: G711 and G729. We extend here the voice modeling of [6]. Figure 1 shows the evolution of the back-offs and channel activity in the network. Let the system slot be δ (for IEEE 802.11b, $\delta = 20\mu s$). $U_j,\ j \in 0, 1, 2, 3, \ldots,$ are the random instants where either an idle slot, or a successful transmission, or a collision ends.

$$\text{Prob}\ \left(B_j^{(1)} = b/(Y_j^{(1)}, Y_j^{(2)}) = (y_1, y_2); L_j = l\right) = \binom{N_1 - y_1}{b}\ (p_l)^b\ (1-p_l)^{N_1 - y_1 - b} \tag{1}$$

$$\text{Prob}\ \left(B_j^{(2)} = b/(Y_j^{(1)}, Y_j^{(2)}) = (y_1, y_2); L_j = l\right) = \binom{N_2 - y_2}{b}\ (p_l)^b\ (1-p_l)^{N_2 - y_2 - b} \tag{2}$$

$$\zeta_1\left(Y_j^{(1)}, Y_j^{(2)}\right) = \begin{cases} p_1 \beta_{Y_j+1} \sum_{l_2=1}^{Y_j^{(2)}} \binom{Y_j^{(2)}}{l_2} \beta_{Y_j+1}^{l_2} (1-\beta_{Y_j+1})^{Y_j - l_2} + \sum_{l_1=2}^{Y_j^{(1)}} \binom{Y_j^{(1)}}{l_1} \beta_{Y_j+1}^{l_1} (1-\beta_{Y_j+1})^{Y_j+1-l_1} \\ + \sum_{l_1=1}^{Y_j^{(1)}} \sum_{l_2=1}^{Y_j^{(2)}+1} \binom{Y_j^{(1)}}{l_1} \beta_{Y_j+1}^{l_1} \binom{Y_j^{(2)}}{l_2} \beta_{Y_j+1}^{l_2} (1-\beta_{Y_j+1})^{Y_j+1-l_1-l_2} \end{cases} \tag{3}$$

$$\zeta_2\left(Y_j^{(1)}, Y_j^{(2)}\right) = \quad p_2 \beta_{Y_j+1} \sum_{l_2=1}^{Y_j^{(2)}} \binom{Y_j^{(2)}}{l_2} \beta_{Y_j+1}^{l_2} (1-\beta_{Y_j+1})^{Y_j - l_2} + \sum_{l_2=2}^{Y_j^{(2)}} \binom{Y_j^{(2)}}{l_2} \beta_{Y_j+1}^{l_2} (1-\beta_{Y_j+1})^{Y_j+1-l_2} \tag{4}$$

Let us define the time between two such successive instants as a *channel slot*. Thus the interval $[U_{j-1}, U_j)$ is called the j^{th} channel slot. We denote this channel slot by L_j. L_j can take five values (in number of system slots): 1 if it is an idle slot, T_{succ1} if it corresponds to a successful transmission of a station/AP with a Type 1 call, T_{succ2} if it corresponds to a successful transmission of a station/AP with a Type 2 call, T_{c-long} if it corresponds to a collision between one Type 1 call station and any other station/AP, and $T_{c-short}$ if it corresponds to a collision between one Type 2 call station and any other station/AP with Type 2 call.

Let N_1 and N_2 be the total number of calls of Type 1 and Type 2 respectively. Let $Y_j^{(1)}$ be the number of non-empty STAs of Type 1 and $Y_j^{(2)}$ be the number of non-empty STAs of Type 2 call stations at the instant U_j. Thus $0 \leq Y_j^{(1)} \leq N_1$ and $0 \leq Y_j^{(2)} \leq N_2$. Let $B_j^{(1)}$ and $B_j^{(1)}$ be the number of new packet arrivals of Type 1 and Type 2 calls respectively. Let $V_j^{(1)}$ and $V_j^{(2)}$ be the number of departures from STAs of Type 1 calls and Type 2 calls respectively in the j^{th} channel slot. At most one departure can happen in any channel slot. Thus,

$$\begin{array}{c} Y_{j+1}^{(1)} = Y_j^{(1)} - V_{j+1}^{(1)} + B_{j+1}^{(1)} \\ Y_{j+1}^{(2)} = Y_j^{(2)} - V_{j+1}^{(2)} + B_{j+1}^{(2)} \end{array}$$

and

$$0 \leq V_{j+1}^{(1)} + V_{j+1}^{(2)} \leq 1.$$

We now describe a key modeling approximation from [6]. In [5] an approximate saturation analysis of a single cell IEEE 802.11 WLAN has been provided. When there are n saturated nodes, denote the attempt probability of each node by β_n. This can be obtained from the fixed point analysis in [5]. The approximation that we employ here is that if n nodes are contending (i.e., have non empty queues), then the attempt probability is taken to be β_n. Thus when there are $Y_j^{(1)}$ Type 1 STAs and $Y_j^{(2)}$ Type 2 STAs contending, the total number of contending STAs is $Y_j := Y_j^{(1)} + Y_j^{(2)}$. Hence, including the AP we take the attempt probability to be β_{Y_j+1}. Now with the Bernoulli model for arrivals and the above state dependent probability of attempt, it is easily seen that $\{Y_j^{(1)}, Y_j^{(2)}; j \geq 0\}$ forms a finite irreducible two dimensional discrete time Markov chain on the channel slot boundaries and hence is positive recurrent. The stationary probabilities π_{n_1,n_2} of the Markov Chain $\{Y_j^{(1)}, Y_j^{(2)}; j \geq 0\}$ can then be determined using expressions of $B_j^{(1)}$, $B_j^{(2)}$, $V_j^{(1)}$ and $V_j^{(1)}$, that are obtained as follows.

Let the probability with which a packet arrives at a node in a slot be λ. Then the probability that at least one packet arrives in l slots will be $1-(1-\lambda)^l = p_l$. Since we assume that packets arrive at only empty STAs, $B_j^{(1)}$ and $B_j^{(2)}$ will be modeled as having a binomial distribution.

$$\begin{array}{l} B_j^{(1)} \sim Bin(N_1 - Y_j^{(1)}, 1-(1-\lambda)^{L_j}) \\ B_j^{(2)} \sim Bin(N_2 - Y_j^{(2)}, 1-(1-\lambda)^{L_j}) \end{array}$$

The probabilities $prob(B_{j+1}^{(1)}/Y_j, L_j)$ and $prob(B_{j+}^{(2)}/Y_j, L_j)$ are given by Eq 1 and Eq 2 respectively.

$V_j^{(1)}$ is 1 if an STA with Type 1 call wins the contention for the channel and 0 otherwise. Similarly $V_j^{(2)}$ is 1 if an STA with Type 2 call wins the contention for the channel and 0 otherwise, i.e.,

$$V_{j+1}^{(1)} = \begin{cases} 1 & \text{w.p. } Y_j^{(1)} \beta_{Y_j+1} (1-\beta_{Y_j+1})^{Y_j} \\ 0 & \text{otherwise} \end{cases}$$

and

$$V_{j+1}^{(2)} = \begin{cases} 1 & \text{w.p. } Y_j^{(2)} \beta_{Y_j+1} (1-\beta_{Y_j+1})^{Y_j} \\ 0 & \text{otherwise} \end{cases}$$

The process $\{\ \{Y_j^{(1)}, Y_j^{(2)}; U_j\}, j = 0, 1, 2, \ldots\}$ can be seen to be a Markov Renewal process with L_j being the renewal cycle time. We make use of Markov regenerative framework to find the throughput of AP. In order to apply the well known Renewal Reward Theorem, we need the mean renewal cycle time and hence we identify the probabilities of L_j as follows:

Let $\eta(Y_j^{(1)}, Y_j^{(2)})$ be the probability of channel slot being idle, $\alpha_1(Y_j^{(1)}, Y_j^{(2)})$ be the probability that a STA with Type 1 packet succeeds, $\alpha_2(Y_j^{(1)}, Y_j^{(2)})$ be the probability that a STA with Type 2 packet succeeds, $\sigma_1(Y_j^{(1)}, Y_j^{(2)})$ be the probability that the AP succeeds and sends Type 1 packet, $\sigma_2(Y_j^{(1)}, Y_j^{(2)})$ be the probability that the AP succeeds and sends Type 2 packet, $\zeta_1(Y_j^{(1)}, Y_j^{(2)})$ be the probability that there is a long collision (involving at least one Type 1 packet) and $\zeta_2(Y_j^{(1)}, Y_j^{(2)})$ be the probability that there is a short collision (not involving a Type 1 packet). Then L_j takes the five values with the following probabilities.

$$L_{j+1} = \begin{cases} 1 & \text{w.p. } \eta(Y_j^{(1)}, Y_j^{(2)}) \\ T_{succ1} & \text{w.p. } \alpha_1(Y_j^{(1)}, Y_j^{(2)}) + \sigma_1(Y_j^{(1)}, Y_j^{(2)}) \\ T_{succ2} & \text{w.p. } \alpha_2(Y_j^{(1)}, Y_j^{(2)}) + \sigma_2(Y_j^{(1)}, Y_j^{(2)}) \\ T_{c-long} & \text{w.p. } \zeta_1(Y_j^{(1)}, Y_j^{(2)}) \\ T_{c-short} & \text{w.p. } \zeta_2(Y_j^{(1)}, Y_j^{(2)}) \end{cases}$$

where

$$\begin{aligned} \eta(Y_j^{(1)}, Y_j^{(2)}) &= (1-\beta_{Y_j+1})^{(Y_j+1)}, \\ \alpha_1(Y_j^{(1)}, Y_j^{(2)}) &= Y_j^{(1)}\beta_{Y_j+1}(1-\beta_{Y_j+1})^{Y_j}, \\ \alpha_2(Y_j^{(1)}, Y_j^{(2)}) &= Y_j^{(2)}\beta_{Y_j+1}(1-\beta_{Y_j+1})^{Y_j}, \\ \sigma_1(Y_j^{(1)}, Y_j^{(2)}) &= p_1\ \beta_{Y_j+1}(1-\beta_{Y_j+1})^{Y_j}, \\ \sigma_2(Y_j^{(1)}, Y_j^{(2)}) &= p_2\ \beta_{Y_j+1}(1-\beta_{Y_j+1})^{Y_j}, \\ \zeta_1(Y_j^{(1)}, Y_j^{(2)}) &\ \text{ and } \zeta_2(Y_j^{(1)}, Y_j^{(2)}) \text{ are given by Eq 3 and Eq 4;} \end{aligned}$$

with

$$\begin{aligned} p_1 &= \frac{N_1}{N_1+N_2}, \\ p_2 &= \frac{N_2}{N_1+N_2} \end{aligned}$$

and

$$\begin{aligned} T_{succ1} &= T_P + T_{PHY} + \frac{L_{MAC}+L_{voice1}}{C_d} + T_{SIFS} + \\ &\qquad T_P + T_{PHY} + \frac{L_{ACK}}{C_c} + T_{DIFS}, \\ T_{succ2} &= T_P + T_{PHY} + \frac{L_{MAC}+L_{voice2}}{C_d} + T_{SIFS} + \\ &\qquad T_P + T_{PHY} + \frac{L_{ACK}}{C_c} + T_{DIFS}, \\ T_{c-long} &= T_P + T_{PHY} + \frac{L_{MAC}+L_{voice1}}{C_d} + T_{EIFS}, \\ T_{c-short} &= T_P + T_{PHY} + \frac{L_{MAC}+L_{voice2}}{C_d} + T_{EIFS}, \\ T_{EIFS} &= T_P + T_{PHY} + \frac{L_{ACK}}{C_c} + T_{SIFS} + T_{DIFS}, \end{aligned}$$

where C_d is the PHY data rate, C_c is the control rate, T_P is preamble transmission time, T_{PHY} is the PHY header transmission time, $L_{Lvoice1}$ is the length of G711 voice packet, $L_{Lvoice2}$ is the length of G729 voice packet, L_{MAC} is MAC header length and L_{ACK} is length of MAC layer ACK packet. See Table 1 for values of parameters. For IEEE 802.11b the channel slot values are $T_{succ1} = 34, T_{succ2} = 29, T_{c-long} = 37$ and $T_{c-short} = 32$ (all in system slot units).

C. Voice Call Capacity

Let A_j be the reward when the AP wins the channel contention. If there are n_1 STAs of Type 1 calls active and n_2 STAs of Type 2 calls active, then we have,

$$A_j = \begin{cases} 1 \text{ w.p. } \beta_{n+1}(1-\beta_{n+1})^n \\ 0 \text{ otherwise} \end{cases}$$

where $n = n_1 + n_2$.

Let $A(t)$ denote the cumulative reward of the AP until time t. Applying Markov regenerative analysis (or the renewal reward theorem) we obtain the service rate of the AP as

$$\Theta_{AP-voip}(N_1, N_2) = \lim_{t\to\infty} \frac{A(t)}{t} \overset{a.s.}{=} \frac{\sum_{n_1=0}^{N_1}\sum_{n_2=0}^{N_2} \pi_{n_1,n_2}\ E_{n_1,n_2}A}{\sum_{n_1=0}^{N_1}\sum_{n_2=0}^{N_2} \pi_{n_1,n_2}\ E_{n_1,n_2}L}$$

where, $E_{n_1,n_2}A = E(A_j/(Y_j^{(1)}, Y_j^{(2)}) = (n_1, n_2))$ and $E_{n_1,n_2}L = E(L_j/(Y_j^{(1)}, Y_j^{(2)}) = (n_1, n_2))$ and $\Theta_{AP-voip}(N_1, N_2)$ is in packets per slot.

Since the rate at which a single call sends data to the AP is λ, and the AP serves $N(=N_1+N_2)$ such calls the total arrival rate to the AP is $(N_1+N_2)\lambda$ (= $\gamma(N_1, N_2)$ say). Obviously, this rate should be less than $\Theta_{AP-voip}(N_1, N_2)$ for stability. Thus, for permissible combination of N_1 and N_2 calls we need

$$\Theta_{AP-voip}(N_1, N_2) > (N_1+N_2)\lambda$$

The above inequality defines the admission region.

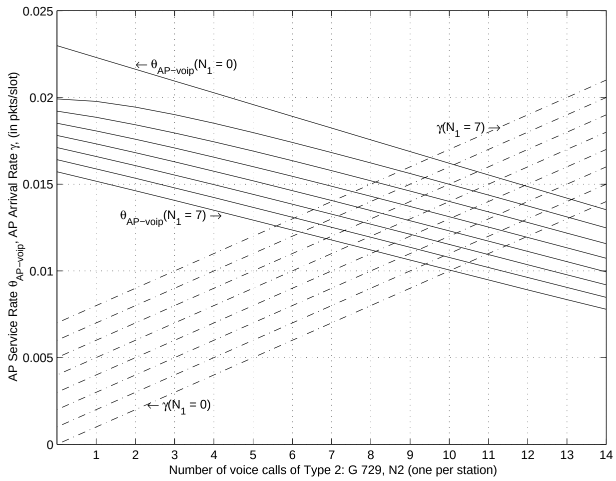

Fig. 2. Results from analysis: The service rate $\Theta(N_1, N_2)$ applied to the AP vs number of voice calls, N_2 for different values of N_1. Also shown are lines $\gamma(N_1, N_2) = (N_1+N_2)\lambda$ for different values of N_1. The point where the γ line crosses the curve for a fixed value of N_1 gives the maximum number of calls supported; N_1 use G711 Codec and N_2 use G729 Codec.

D. Numerical Results and Validation

We present our simulation results and compare them with results obtained from the simulation. The simulations were done using ns 2 [8]. In Figure 2 we plot the numerical results

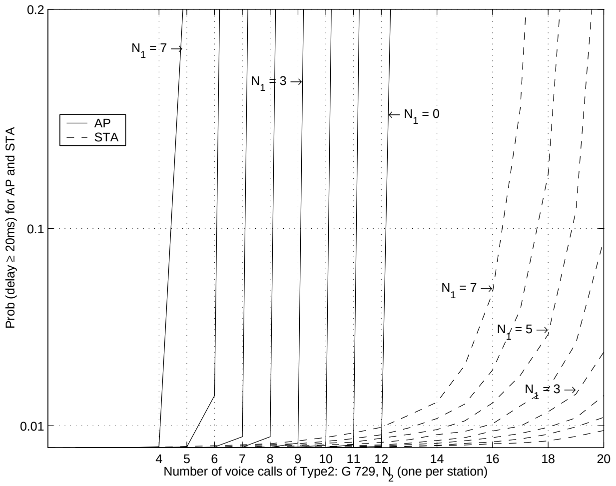

Fig. 3. Results from simulation: The $Prob(delay \geq 20ms)$ at AP and STA vs number of voice calls, N_2. N_1 use G711 Codec and N_2 use G729 Codec.

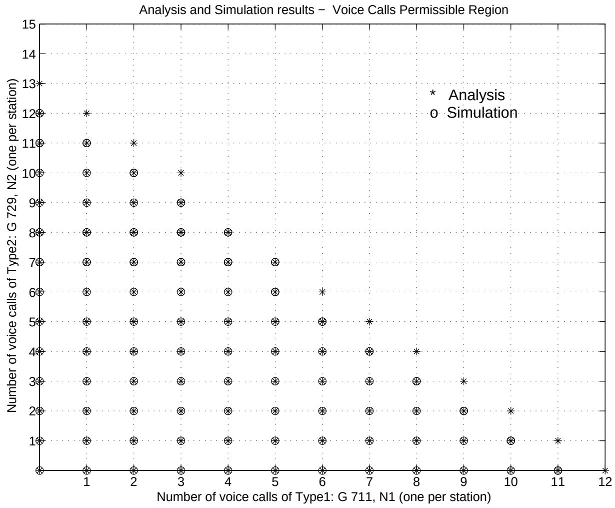

Fig. 4. Analysis and simulation results: The admissible combinations of Type 1 and Type 2 calls. N_1 use G711 Codec and N_2 use G729 Codec. The data rate is 11 Mbps.

for the AP service rate and load arrival rate at the AP vs values of N_2. The different curves correspond to different values of N_1 starting from 0. The simulation results for the QoS objective of $Prob(delay \geq 20ms)$ for the AP and the STAs are shown in Figure 3.

From Figure 2 we observe that for each value N_2, as we increase the value of N_1 the service rate available to the AP decreases. This is, of course, because more service needs to be given to the STAs as the number of calls increases. Observe that for $N_1 = 0$, the rate of packets arriving into the AP is $N_2\lambda$ packets per slot. This exceeds the curve $\theta_{AP-voip}(0, N_2)$ after $N_2 = 13$ but before $N_2 = 14$. Hence, from the analysis, we can conclude that the pair $(N_1 = 0, N_2 = 13)$ can be admitted. Looking at Figure 3, we find that for $N_1 = 0$, the $Prob(delay - AP \geq 20ms)$ shoots up after $N_2 = 12$. As in [6] we find that our analysis overestimates the capacity by 1 call. For $(N_1 = 0, N_2 = 12)$, the $Prob(delay - STA \geq 20ms)$ is close to 0, *confirming that the AP is the bottleneck*, as per our assumptions. Similarly, for $N_1 = 7$, the analysis says that we can permit $N_2 = 5$, whereas the simulations show that we can permit $N_2 = 4$.

These observations are also summarized in Figure 4 , where the $\circ$ symbols show the (N_1, N_2) pair, admissible by the simulations and the $*$ symbols show the call admission points obtained by analysis. Thus the analysis captures the admissible region very well, and in practice we can use the rule of thumb of accepting one call less than that given by the analysis.

IV. Summary

In this paper we have extended the packet voice analysis of [6] to obtain the admission region for two types of voice calls with different codecs in an IEEE 802.11b infrastructure WLAN. The analysis proceeds, as before, by modeling the evolution of the number of contending STAs at channel slot boundaries. This yields a Markov renewal process. A regenerative analysis then yields the service rate applied to the AP assuming that the AP is saturated. Comparison of this number with the load into the AP for each number of voice calls, yields the desired admission region. We obtain the two dimensional admission region. Our analysis captures the admission region well, overestimating it by just one call.

Our ongoing work will extend this analysis to the IEEE 802.11e standard and will also analyze the system with simultaneous VoIP and TCP transfers.

Acknowledgments

This work is based on research sponsored by Intel Technology, India.

References

[1] S. Garg and M. Kappes. Can I add a VoIP call? In *Proceedings of IEEE ICC 2003*, volume 2, pages 779–783, 11 - 15 May 2003.

[2] G. H. Hwang and D. H Cho. Voice capacity in IEEE 802.11 wireless LANs. volume 40, issue 18, pages 1137 – 1138, 2 Sept. 2004.

[3] IEEE 802.11 standard for Wireless Local Area Networks. *h*ttp://standards.ieee.org/getieee802/802.11.html.

[4] IEEE 802.11b standard for Wireless Local Area Networks. *h*ttp://standards.ieee.org/getieee802/802.11.html.

[5] A. Kumar, E. Altman, D. Miorandi, and M. Goyal. New Insights from a Fixed Point Analysis of Single Cell IEEE 802.11 WLANs. In *Proceedings of IEEE INFOCOM 2005, The 24th Annual Joint Conference of the IEEE Computer and Communications Societies.*, pages 1550 – 1561, 13 - 17 March 2005.

[6] G. Kuriakose. Master of Engg Thesis: Analytical Models for QoS Engineering of Wireless LANs, ECE Department, Indian Institue of Science, Bangalore, April 2005.

[7] K. Medepalli, P. Gopalakrishnan, D. Famolari, and T. Kodama. Voice Capacity of IEEE 802.11b, 802.11a and 802.11g Wireless LANs. In *Proceedings of IEEE GLOBECOM 2004*, volume 3, pages 1549–1553, 29 Nov - 3 Dec 2004.

[8] The Network Simulator ns2 http://www.isi.edu/nsnam/ns/.