Volume 2, No. 6, Nov-Dec 2011
ISSN No. 0976-5697
International Journal of Advanced Research in Computer Science
RESEARCH PAPER
Available Online at www.ijarcs.info

Analytical Modeling for Performance of Multimedia Applications in WLAN

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Abstract: In the context of the IEEE 802.11e standard for WLANs, we provide an analytical model for obtaining the maximum number of VoIP calls that can be supported on HCCA, such that the delay QoS constraint of the accepted calls is met, when TCP downloads are coexistent on EDCA. In this scenario, we derive the TCP download throughput by using an analytical model for the case where only TCP sessions are present in the WLAN. We extend the modeling heuristic of to evaluate the performance of an IEEE 802.11e infrastructure network carrying packet telephone calls, streaming video sessions and TCP controlled file downloads, using Enhanced Distributed Channel Access (EDCA). We identify the time boundaries of activities on the channel (called channel slot boundaries) and derive a Markov Renewal Process of the contending nodes on these epochs. This is achieved by the use of attempt probabilities of the contending nodes as those obtained from the saturation fixed point analysis.

Keywords: analytical modeling, performance, multimedia applications, WLAN

I. INTRODUCTION

The IEEE 802.11e [1] standard has been introduced in order to provide differentiated services to different traffic flows in an IEEE 802.11 WLAN. The 802.11e standard defines a new coordination function called hybrid coordination function (HCF). HCF has two modes of operation: enhanced distributed channel access (EDCA) which is a contention-based channel access function, and HCF controlled channel access (HCCA) which is based on a polling mechanism controlled by the hybrid co-ordinator (HC), which is normally resident in the QoS aware access point (QAP). EDCA and HCCA can operate concurrently in IEEE 802.11e WLANs. In this paper we provide an analysis of HCF with VoIP calls being carried on HCCA, and TCP file transfer downloads on EDCA. Each VoIP call comprises a wireless QSTA (QoS aware wireless STAtion) engaged in a VoIP call with a wired client via the QAP. In the case of TCP sessions, each STA engaged in a TCP transfer is downloading a long file from a server on the wired LAN via the QAP. There have only been a few attempts to model and analyze the IEEE 802.11e MAC when subjected to actual Internet traffic traffic loads.

II. THE MODELING APPROACH

We study the performance of a single cell infrastructure 802.11e WLAN that uses EDCA, when AC 3, AC 2 and AC 1are used for voice, video and data respectively. The modeling approach follows that of [1] and can be briefly explained as follows:

a. Embed the number of contending nodes (i.e., those that have non empty queues) at *channel slot boundaries*. The *channel slot boundaries* are those instants of time when an activity ends or there is a back off slot after which no node attempts. The activity could be a successful transmission or a collision.
b. Use the heuristic that, if n nodes are contending at a channel slot boundary, their attempt probabilities are those obtained from fixed point analysis of [2] with n saturated nodes.
c. Use the thus obtained attempt probabilities to model the evolution of the number of contending nodes Channel slot boundaries. Since the channel slot durations depend on the activity, this yields a Markov renewal process [8]

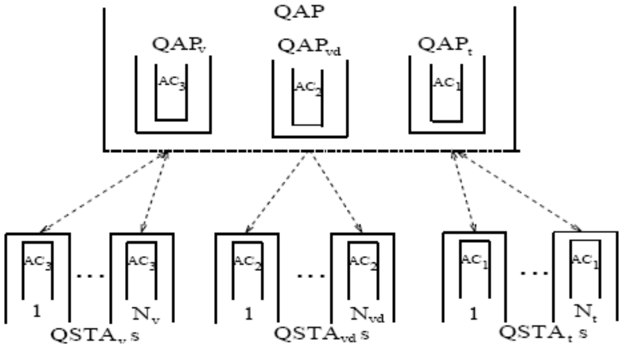

Figure 1: IEEE 802.11e WLAN model for VoIP calls, streaming video sessions and TCP traffic on EDCA

d. Obtain the stationary probability vector _ of the embedded Markov chain of the Markov renewal process.
e. Use a Markov regenerative argument to obtain the performance measures

A. The Network Scenario and Modeling Observations:

We consider an infrastructure IEEE 802.11e WLAN, which has VoIP, downlink video streaming and TCP controlled file download traffic, serviced on EDCA. While IEEE 802.11e also defines EDCA TXOPs for transmission of more than one MSDUs (MAC Service Data Unit) when a node obtains the opportunity to transmit [3, Section 9.1.3.1], we use the default value that the sender can send not more than one MSDU in an EDCA TXOP. Let Nv be the number of full duplex CBR VoIP calls, Nvd be the number of simplex CBR download video streaming sessions and Nt be the number of TCP controlled file transfers in the WLAN. We carry forward the following assumptions from [1]:

A1 There are no hidden nodes in the WLAN, there are no bit errors, and packets in the channel are lost only due to collisions. A2 The VoIP traffic, video streaming traffic and TCP traffic all originate from different QSTAs. Thisimplies that each QSTA has only one type of traffic. Denote the QSTAs with VoIP traffic (AC 3 queue) as QSTAv, the QSTAs with video streaming traffic(AC 2 queue) as QSTAvd and QSTAs with TCPcontrolled file transfers (AC 1 queue) as QSTAt.A3 The QAP can be viewed as three nodes: QAPv, an AC 3 queue, for downlink VoIP traffic of all VoIPcalls, QAPvd, an AC 2 queue, for downlink video streaming traffic of all video streaming sessions, and QAPt, an AC 1 queue, for all TCP downloads.Assumptions A2 and A3 are simplifying implications of animportant observation in [2], viz, "with increase in the number of nodes, the performance of the multiple queues per node case coincides with the performance of the single queue per node case, each node with one queue of the original system ". This model is illustrated in Figure 2. Note that at any time the WLAN in Figure 2 can be seen to consist of Nv+Nvd+Nt+3 nodes.

B. VoIP Traffic:

We consider non-synchronized CBR duplex VoIP calls from codecs that generate VoIP packets every 20 ms. As a QoS requirement we demand that the probability that a packet is transmitted successfully within 20 ms is close to 1 [12]. Following are the assumptions that we carry forward from [1] and are justified in [1] and [12]:A4 The buffer of every QSTAv has a queue length of at most one packet A5 New packets arriving to the QSTAvs arrive only at empty queues. This assumption implies that if there are k QSTAvs with voice packets then, a new voice packet arrival comes to a (k + 1)th QSTAv. A6 QAPv is the capacity bottleneck for voice traffic, since, there can be up to Nv packets of different calls in the QAPv. Therefore to obtain the VoIP capacity of theWLAN, we consider QAPv saturated. But when we need to evaluate the throughputs of streaming video sessions and TCP download streams, we model the arriving VoIP traffic at QAPv. As mentioned earlier, packets arrive every 20 ms in every stream. We use this model in our simulations. However, since our analytical approach is via Markov chains, to model theVoIP traffic, we assume that the probability that a voice call generates a packet in an interval of length l slots is $p_l = 1 - (1 - \lambda)^l$, where λ is obtained as follows. Each system slot is of 20μs duration (hereafter denoted as δ). Thus in 1000 system slots there is one arrival. Therefore, for the 802.11b PHY we take λ = 0.001. This simplification turns out to yield a good approximation.

C. TCP Controlled File Downloads:

Each QSTAt has a single TCP connection to download a large file from a local file server. Hence, the QAPt delivers TCP data packets towards the QSTAts, while the QSTAts return TCP ACKs. [1] [12] A7 The QAPt and the QSTAts have buffers large enough so that TCP data packets or ACKs are not lost due to buffer overflows. A8 Each QSTAt can have a maximum of one TCP ACK packet queued up. This assumption implies two things. First, after an QSTAt's successful transmission, the number of active QSTAts reduces by one. Second, each successful transmission from the QAPt activates a new QSTAt. A9 QAPt is the traffic bottleneck and hence saturated and always contends for the channel.

D. Video Streaming Traffic:

We consider the scenario where the WLAN users connect toa video streaming server located in the wired network, through the QAP. A10 In our work, we assume that video packets are streamed over UDP between the streaming server and the wireless playout station, without any feedback traffic from the playing station. This assumption implies that the QTAvds do not have any uplink traffic and hence never contend for the channel. Li et al. [14] have studied the two dominant streaming multimedia products, RealNetworks RealPlayerTM and Microsoft MediaPlayerTM and their experiments for a low rate video stream using UDP show that 1) The sizes of MediaPlayer packets are concentrated around the mean packet size (of 900 bytes). The sizes of RealPlayer packets are spread more widely over a range from 0.6 to 1.8 of the mean normalized packet size. 2) The packet inter arrival times for RealPlayer varied over a range of 10 ms to 160 ms. In contrast, the packet inter arrival times for MediaPlayer are concentrated near 130 ms, indicating that most packets arrive at constant time intervals.

The packet inter arrrival times were mainly attributed to the property of the streaming server. Thus they draw the conclusion that the packet sizes and rates generated by MediaPlayer are essentially CBR while the packet sizes and rates generated by RealPlayer are more varied. A11 In the analysis we obtain the maximum service rate obtainable by the video streams by considering that the video queue is saturated. Thus QAPvd is saturated and always contends for the channel. A12 In simulations, we consider CBR video streams (one of the two choices as observed by Li et al. , discussed above) and consider a rate of 1.5 Mbps and packet size of 1500 bytes, for validation, since, when the SD -TV (Standard Definition Television) resolution video is coded with H.264 for an MoS of 4, the output streaming video rate is 1.5 Mbps [8].

III. AN EMBEDDED CHAIN

The evolution of the channel activity in the network is Uj, j Є 0,1,2,3,…. Are the random instants where either an idle slot, or a successful transmission, or a collision ends.

Let us define the time between two such successive instants as a channel slot. Thus the interval [Uj-1, Uj) is called the jth channel slot . Let the time length of jth slot be Lj

The implication of access differentiation through AIFS is the ACs with large4r AIFS values cannot counted in those slots that were preceded by some activity (i.e., successful transmission or collision). After every activity (successful transmission or collision) on the channel. AC1 nodes wait for an additional system slot before contending for the channel. When AC 1 nodes have still to wait for one more system slot to be able to contend. At other instants. U5, U8, U11, nodes with AC 3 or AC 1 can attempt.

The AC attempt probabilities obtained from [13] are conditioned on when an AC can attempt. We use the variable $Y_j^{(s)}$ to keep of which ACs are permitted to attempt in a channel slot. Let $Y_j^{(s)}$ =1 denote that the preceding channel slot had an activity and so in the beginning of the jth channel slot remained idle and hence, at the beginning of the jth channel slot any node can attempt. Thus $Y_j^{(s)}$ € {0, 1}.

Then we have the following dynamics.

$$Y_{j+1}^{(v)} = Y_j^{(v)} - V_{j+1}^{(vSTA)} + B_{j+1}^{(v)}$$

$$Y_{j+1}^{(t)} = Y_j^{(t)} - V_{j+1}^{(tSTA)} + B_{j+1}^{(tAP)}$$

With the condition that

$$V_{j+1}^{(vSTA)} + V_{j+1}^{(vAP)} + V_{j+1}^{(tSTA)} + V_{j+1}^{(tAP)} \in \{0,1\}.$$

Since the probability with which a packet arrives at a node in a channel slot of length l is pl (as in chapter 4) and we assume that packets arrive at only be empty QSTAvs, $B_j^{(v)}$ can be modeled as having a binomial distribution and conditioned probability

$$\text{prob}\left(B_{j+1}^{(v)} / Y_j^{(v)} = n_v; L_{j+1} = l\right) = \binom{N_v - n_v}{b} (pl)^b \left(1 - pl\right)^{N_v - n_v - b}$$

Where pl=1- (1- λ)I as before.

7.2.2 Markov Property via State Dependent Attempt Probabilities

For determining the expressions of $V_{j+1}^{(vSTA)}$, $V_{j+1}^{(vAP)}$, $V_{j+1}^{(tSTA)}$, $V_{j+1}^{(tAP)}$, we also use the attempt probabilities of [13]. Let $\beta_{n_v+1,n_t+1}^{(v)}$ be the attempt probability of node with AC 3 and $\beta_{n_v+1,n_t+1}^{(t)}$ be the attempt probability of node with AC 1, when there are nv VoIP calls and nt TCP sessions in the attempt. Note that the addition of one in the subscripts is so as to include the QAPv and QAPt. which by assumption always contend. The values, $\beta_{n_v+1,n_t+1}^{(v)}$ for AC 3 and $\beta_{n_v+1,n_t+1}^{(t)}$ for AC 1are obtained from saturation fixed point analysis of [38]for all combinations pf nv,nt. Our approximation is that the state dependent values of attempt probabilities from the saturated nodes case can be used for a WLAN where the nodes are not saturated, by keeping track of number of nonempty nodes in the WLAN and taking the state dependent attempt probabilities corresponding to this number of nonempty nodes.

We can express the conditional distributions $V_{j+1}^{(vSTA)}$, $V_{j+1}^{(vAP)}$, $V_{j+1}^{(tSTA)}$ and $V_{j+1}^{(tAP)}$, using these functions. $V_{j+1}^{(vSTA)}$ is 1 if a QSTAv wins the contention for the channel and 0 otherwise. Then

$$V_{j+1}^{(vSTA)} / Y_{j+1}^{(s)} = \begin{cases} 1\, w.p\; \alpha\left(Y_{j+1}^{(v)}, Y_{j+1}^{(t)}\right) \eta_t \left(Y_{j+1}^{(v)}, Y_{j+1}^{(t)}\right) & if\; Y_{j+1}^{(s)} = 0 \\ 1\, w.p\; \alpha\left(Y_{j+1}^{(v)}, Y_{j+1}^{(t)}\right) & if\; Y_{j+1}^{(s)} = 1 \\ 0 \quad otherwise \end{cases}$$

Similarly,

$$V_{j+1}^{(vAP)} / Y_{j+1}^{(s)} = \begin{cases} 1\, w.p\; \sigma\left(Y_{j+1}^{(v)}, Y_{j+1}^{(t)}\right) \eta_t \left(Y_{j+1}^{(v)}, Y_{j+1}^{(t)}\right) & if\; Y_{j+1}^{(s)} = 0 \\ 1\, w.p\; \sigma\left(Y_{j+1}^{(v)}, Y_{j+1}^{(t)}\right) & if\; Y_{j+1}^{(s)} = 1 \\ 0 \quad otherwise \end{cases}$$

$V_{j+1}^{(tSTA)}$ is 1 if QSTA with AC 1 wins the contention for the channel and 0 otherwise.

So $V_{j+1}^{(tSTA)}$

$$/ Y_{j+1}^{(s)} = \begin{cases} 1\, w.p\; \alpha\left(Y_{j+1}^{(v)}, Y_{j+1}^{(t)}\right) \eta_v \left(Y_{j+1}^{(v)}, Y_{j+1}^{(t)}\right) & if\; Y_{j+1}^{(s)} = 0 \\ 0 \quad otherwise \end{cases}$$

And $V_{j+1}^{(tAP)}$ is 1 if QAPt wins the contention for the channel and 0 otherwise. Therefore we have

$$V_{j+1}^{(tAP)} / Y_{j+1}^{(s)} = 0 = \begin{cases} 1\, w.p\; \sigma\left(Y_{j+1}^{(v)}, Y_{j+1}^{(t)}\right) \eta_v \left(Y_{j+1}^{(v)}, Y_{j+1}^{(t)}\right) & if\; Y_{j+1}^{(s)} = 0 \\ 0 \quad otherwise \end{cases}$$

When $Y_{j+1}^{(s)}$ =1, $V_{j+1}^{(tSTA)}$ ($Y_{j+1}^{(s)}$ =1)= $V_{j+1}^{(tAP)}$ ($Y_{j+1}^{(s)}$ =1)=0

$Y_{j+1}^{(s)}$ takes the values in {0, 1} with the following probabilities:

$$Y_{j+1}^{(s)} \begin{cases} 1\, w.p\; \eta_v \left(Y_j^{(v)}, Y_j^{(t)}\right) \eta_t \left(Y_j^{(v)}, Y_j^{(t)}\right) \\ 0 \quad otherwise \end{cases}$$

With the initial state, $Y_0^{(s)} = 0$.

With the assumed binomial distribution for voice packet arrivals and the state dependent probabilities of attempt, it is easily seen that { $Y_j^{(v)}$, $Y_j^{(t)}$, $Y_j^{(s)}$; j ≥ 0}

forms a finite irreducible three dimensional discrete time Markov chain on the channel slot boundaries and hence is positive recurrent. The stationary probabilities $\pi_{n_v n_t n_s}$ of the Markov Chain { $Y_j^{(v)}$, $Y_j^{(t)}$, $Y_j^{(s)}$; j ≥ 0} can then be numerically determined using expressions for distributions of $\beta_j^{(v)}$, $V_j^{(vAP)}$, $V_j^{(vSTA)}$, and $V_j^{(tAP)}$, $V_j^{(tSTA)}$ and the probability functions defined above.

IV. THE MARKOV RENEWAL PROCESS

In this section we use the state dependent attempt probabilities to obtain the distribution of the channel slot duration. On combining this with the Markov Chain in Sec 7.2.2 we finally conclude that

{$Y_j^{(v)}$, $Y_j^{(t)}$, $Y_j^{(s)}$;Uj), j=0,1,2…} is a Marcov renewal process.

We use the basic access mechanism for the TCP traffic. This shall facilitate the validation of analytical results through simulations by the ns-2 with EDCA implementation [44] that supports only basic access mechanism and not RTS/CTS mechanism. However our Analysis can be worked out for RTS/CTS mechanism as well. When basic access mechanism is used, there shall be collision between three kinds of packets. The longest collision time is seen when QAPt packet collides with a packet of any other node. A smaller collision time is seen when VoIP packet collides with a packet of any other node except with packet of QAPt. The shortest collision time is seen when only packets of QSTAts collide. Then Lj (in system slots) takes one of the seven values: 1 if it is an idle slot. Ts-v if it corresponds to a successful transmission of a AC 3 node, Ts-tAP if it corresponds to a successful transmission of QAPt, Ts-tSTA if it corresponds to a successful transmission of QSTAt. Tc-short if it corresponds to a collision between QSTAts, Tc-voiceif it corresponds to a collision amongst nodes with AC 3 or between AC 3 nodes and any QSTAt and Tc-long if it corresponds to a collision between QAPt and Tc-long if it corresponds to a collision between QAPt and any other QSTA. The distribution of Lj is given in the Appendix. The various values of Ljare as follows:

$$T_{EIFS} = T_P + T_{PHY} + \frac{L_{ACK}}{C_c} + T_{SIFS},$$

$$T_{sv} = T_P + T_{PHY} + \frac{L_{MAC} + L_{voice}}{C_d} + T_{SIFS} + T_P + T_{PHY} + \frac{L_{ACK}}{C_c} + T_{AIFS(3)},$$

$$T_{stSTA} = T_P + T_{PHY} + \frac{L_{MAC} + L_{TCPACK}}{C_d} + T_{SIFS} + T_P + T_{PHY} + \frac{L_{ACK}}{C_c} + T_{AIFS(1)},$$

$$T_{stAP} = T_P + T_{PHY} + \frac{L_{MAC} + L_{IPH} + L_{TCPdata}}{C_d} + T_{SIFS} + T_P + T_{PHY} + \frac{L_{ACK}}{C_c} + T_{AIFS(1)}$$

$$T_{c-voive} = T_P + T_{PHY} + \frac{L_{MAC} + L_{voice}}{C_d} + T_{EIFS} + T_{AIFS(3)},$$

$$T_{c-short} = T_P + T_{PHY} + \frac{L_{MAC} + L_{TCPACK}}{C_d} + T_{EIFS} + T_{AIFS(1)},$$

$$T_{c-long} = T_P + T_{PHY} + \frac{L_{MAC} + L_{IPH} + L_{TCPdata}}{C_d} + T_{EIFS} + T_{AIFS(1)},$$

Where Cd is the data rate, Cc is the control rate, Tp is preamble transmission time TPHY is the PHY header transmission time, LMAC is MAC header length, LLvoice1is the length of G711 voice packet, LIPH is the length of TCP/IP Header, LTCPdata is the lengthOf MAC ACK packet. See table 8.1 for values of parameters. Table 7.2 for values of Lj for PHY data rate of 11 Mbps and control rate of 2Mbps.

Table 7.1 Parameters used in analysis and simulation for EDCA802.11e WLAN

Parameter	Symbol	Value
PHY data rate	Cd	11Mbps
Basic (control) rate	Cd	2Mbps
G711 packet size	Lvoice	200Bytes
Data packet size	LTCPdata	1500Bytes
PLCP preamble time	Tp	144µs
PHY Header time	TPHY	48 µs
MAC – layer ACK Packet Size	LACK	112bits
MAC Header Size	LMAC	288bits
AIFS(3) Time	TAIFS(3)	50 µs
AIFS(1) Time	TAIFS(1)	70 µs
SIFS Time	TSIFS	10 µs
Min. Contention Window for AC(3)	CWmin(AC(3)	7
Min. Contention Window for AC(3)	CWmin(AC(3)	15
Min. Contention Window for AC(1)	CWmin(AC(3)	31
Min. Contention Window for AC(1)	CWmin(AC(3)	1023
Idle slot / system slot (using 802.11b)	δ	20 µs

Table 7.2 Values of Lj using basic mechanism, VoIP packet size of 200 bytes, data packet size of 1500 bytes and PHY rate of 11Mbps and control rate of 2Mbps.

Parameter	Symbol	Value
AC(3) Successful transmission time	Tsv	34
AC 1 QAP Successful transmission time	TstAP	84
AC 1 QSTA Successful transmission time	L stSTA	29
Collision transmission involving AC 3 and any other QSTA except QAPt	Tc-voice	37 32
Collision transmission involving AC 1 only	Tc-short	
Collision transmission involving AC 1 and any other QAPt and any other QSTA	Tc-long	87
Extended His (without AIFS)	TEIFS	314µs

We thus see that

$$P\left(\begin{array}{l} Y_{j+1} = y, (U_{j+1} - U_j) \le l / (Y_0 = y_0, U_0 = u_0), (Y_1 = y_1, U_1 = u_1) \ldots (Y_j = y_y, U_j = u_j) \\ = P(Y_{j+1} = y, U_{j+1} - U_j) \le l (Y_j = y_j, U_j) \end{array} \right)$$

And so conclude that

{$Y_j^{(v)}$, $Y_j^{(t)}$, $Y_j^{(s)}$; (Uj),j=0,1,2…} is a Markov renewal process with Lj= Uj-1being the renewal cycle time.

V. CONCLUSION

In this paper, we evaluated the performance of EDCA WLAN, when the traffic consists of VoIP calls, streaming video sessions and TCP download transfers. The analysis proceeds by modeling the evolution of the number of contending QSTAs at channel slot boundaries. This yields a Markov renewal process. A regenerative analysis then yields the required performance measures like the VoIP capacity, video saturation throughput and the TCP aggregate download throughput. The model predicts the measures that compare closely with the simulation results.

By an effective bandwidth approach we obtained the buffer size of QAPvd that ensures the probability of loss of video packets to be within 1%. Our work provides the following modeling insights:

a. The idea of using saturation attempt probabilities as state dependent attempt rates yields an accurate model in the unsaturated case.
b. Using this approximation, an IEEE 802.11e infrastructure WLAN can be well modeled by a multidimensional Markov renewal process embedded at channel slot boundaries. We also obtain the following performance insights:
c. Unlike the original DCF, the EDCA mechanism supports the coexistence of VoIP connections, video

streams and TCP file transfers; but even one video streaming session and one TCP transfer reduces the VoIP capacity from 12 calls to 6 calls. Subsequently the VoIP capacity is independent of the number of video sessions and TCP transfers (see Figures 4 and 5).

d. For an 11 Mbps PHY, the net video throughput reduces linearly by 0.38 Mbps per additional VoIP call and when both VoIP and video sessions are present, the TCP file download throughput reduces linearly with the number of voice calls by 0.14 Mbps per additional VoIP call.

e. By using a small buffer for AC 2 of AP (about 75KB), the video packet loss probability can be kept within permissible limits (i.e., ≤ 0.01). In related work, we have also provided an analytical model for IEEE 802.11e infrastructure WLANs, with voice being carried in contention period using HCCA, and TCP data in the remaining time using EDCA (see [29]).

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