Voice Capacity of IEEE 802.11b, 802.11a and 802.11g Wireless LANs

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Abstract- **IEEE 802.11 based Wireless Local Area Networks (WLANs) are becoming popular in home, enterprise and public access areas primarily due to their low cost, simplicity of installation and high data rates. While WLANs continue to be predominantly data centric, there is growing interest in using WLANs for voice, especially in enterprise markets. This paper presents new analytical and simulation results for the conversational speech capacity of WLANs and compares the different WLAN technologies in that regard. Specifically, we consider IEEE 802.11b, 802.11a and 802.11g systems in the infrastructure mode and find that voice capacity is a strong function of the channel bandwidth, codec packetization interval, data traffic and the packet size used by data. For the IEEE 802.11g system, we find that capacity depends on the CTS-to-self and RTS-CTS legacy protection mechanisms, with the RTS-CTS mechanism achieving lower capacity. We show that the analytical results are in close agreement with those from simulations and conclude the paper by highlighting some key factors that dictate the capacity of WLANs.**

I. INTRODUCTION

IEEE 802.11b based Wireless LANs (also referred to as Wi-Fi) [8, 9] are seeing a remarkable increase in their usage. One of the main reasons for the success of Wi-Fi has been shrinking equipment costs, the simplicity of setup and the high data rates (up to 11 Mb/s). While 802.11b devices operate in the 2.4 GHz ISM bands, newer technologies such as IEEE 802.11a [10] promise higher data rates (up to 54 Mb/s) in the less crowded 5 GHz U-NII bands. IEEE 802.11g based WLANs [11] also support rates of up to 54 Mb/s, however, they operate in the ISM band and are backwards compatible with 802.11b.

WLANs predominantly carry packet data traffic generated by applications such as web browsing and email. Recently, there has been growing interest in using WLANs to support voice communications, especially in an office environment [1, 2]. Thus, employees can engage in voice communications while still being mobile. A natural question to then ask is: How many voice calls can be supported in a WLAN? As shown in previous works on VoIP capacity of WLAN networks [1, 2] and as will be re-iterated in this paper, this question does not have any unique answer. Rather, we will see that the capacity is a strong function of the channel bandwidth, voice codec packetization interval and the data traffic in the system. The IEEE 802.11 standard defines two modes of Medium Access Control (MAC) – DCF (Distributed Co-ordination Function) and PCF (Point Co-ordination Function). DCF allows for highly distributed medium access using CSMA/CA (Carrier Sense Multiple Access/Collision Avoidance) and is the default mode supported by all WLAN devices. The PCF mode was designed with the intent of supporting isochronous traffic such as voice and has been the subject of some prior research [5, 6]. However, since it is specified as optional in the standard, it is seldom implemented in commonly available 802.11b devices. Further, some previous studies have indicated that PCF provides marginal improvement in performance when compared with DCF [5]. In addition to the MAC modes, the standard also specifies two management modes called the Infrastructure and Ad Hoc modes. The Infrastructure mode requires one node to bridge all communications, while the Ad Hoc mode allows direct communication between peers. The Ad Hoc mode is not widespread. The Infrastructure mode, on the other hand, is very widely deployed. Therefore, evaluating the voice capacity of a WLAN in infrastructure mode with DCF is of significant interest. Experimental and analytical results for such a deployment have first been reported in [1]. Other interesting studies reported include experimentation with prioritization of voice traffic at the AP [2] as well as ones with emphasis on delay limited capacity [13]. Performance of DCF protocol itself has been the subject of much prior research and [4] presents a particularly useful analytical model that we will partly use in our analysis.

The main contributions of this paper are three fold. First, we obtain the conversational speech capacity of WLANs via both analysis and simulation. Second, we compare the performance of IEEE 802.11b with that of 802.11a and 802.11g. Third, we consider the cases of 802.11g only WLANs as well as mixed 802.11b and 802.11g WLANs. To the best of our knowledge none of the three have been addressed in prior work in this area. This paper is organized as follows. Section II describes the system model while Section III describes the capacity analysis. Section IV quantitatively compares the capacity of the three systems and we conclude with some key observations from the results in Section V.

II. SYSTEM MODEL

We consider an 802.11 DCF system in infrastructure mode. We assume full duplex VoIP calls, where each call is between a wireless station and a wired station as shown in Figure 1. Each voice call is modeled according to the ITU recommendation for generating conversational speech [3]. We provide the key details of the traffic model here for easy reference. The voice codec used is the G.711 codec which generates voice packets at a rate of 64 Kb/s. We assume there

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are zero delays incurred in the wired part of the system and ignore all propagation delays.

A. Conversational Speech Model

We use the conversational speech model as specified in the ITU P.59 recommendation [3]. The important feature of the recommendation is that it models the conversation between two users A and B as a four state Markov chain with states being: (a) A talking B silent, (b) A silent B talking, (c) both talking, (d) both silent. This is depicted in Figure 2. The durations of states are mutually independent and identically distributed Exponential random variables with means 854 ms, 854 ms, 226 ms and 456 ms respectively. In our study, we assume that voice packets are generated only when a user is in the talking state. In other words, we make the simplifying assumption of silence suppression where no voice packets are generated when a user is silent.

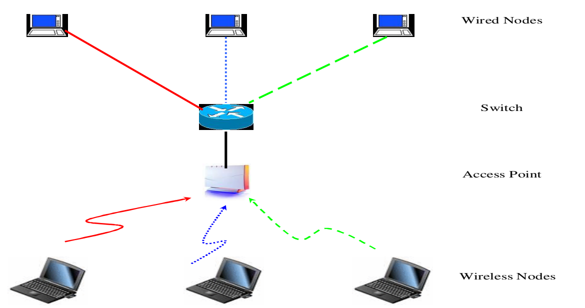

Figure 1 Architectural View of the Wireless LAN System

B. Protocol Layers and Headers

Considering a codec packetisation interval of 10 ms, the raw voice packet is 80 bytes. The headers are composed of 28 bytes for MAC, 20 bytes for UDP, 8 bytes for IP and 12 bytes for RTP. No other header is added to increase the packet size; however, a physical layer header (192 µs for 802.11b, 20 µs for 802.11a and an additional 6 µs for 802.11g) is incurred for every packet transmission.

C. MAC Layer Description

The DCF is governed by a “listen-before-talk” protocol. Every station that needs to send a packet first senses the channel for at least a duration of DIFS (Distributed Inter Frame Spacing). If the channel is sensed idle, the station chooses a random back-off counter value uniformly distributed in the range of [0, CW], where CW stands for Contention Window. CW is maintained in units of SLOT and is initially set to CWmin. Once the back-off counter value is chosen, it is decremented by one for each slot the channel is sensed idle. If the channel is sensed busy before the counter reaches zero, the decrementing process is frozen and is resumed only when the channel is sensed idle for a DIFS period again. After transmission, the sender expects to receive an acknowledgement (ACK) within a SIFS (Short Inter Frame Spacing) period. If an ACK is not received within ACKtimeout period, the packet is assumed lost. Each time a packet is lost (either due to collision or to channel errors), the

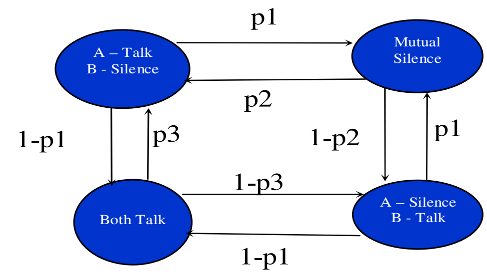

Figure 2 Conversational Speech Modeled as 4 state Markov Chain [3], p1=0.4, p2=p3=0.5

contention window is doubled until the maximum value of CWmax is reached. The station makes limited attempts to retransmit a packet, as specified by the RETRY_LIMIT parameter. Upon successful transmission of a packet, the CW is reset to CWmin.

Table 1 Summary of important constants in 802.11b, 802.11a and 802.11g

Parameter	Value			
	802.11b	802.11a	802.11g only	802.11g + legacy
SLOT	20 µs	9 µs	9 µs	20 µs
SIFS	10 µs	16 µs	10 µs	10 µs
DIFS (SIFS + 2xSLOT)	50 µs	34 µs	28 µs	50 µs
Physical Layer Header Length	192 µs [long] 96 µs [short]	20 µs	20 µs	20 µs
Min. Mandatory Data Rate [Mb/s]	1	6	6	1
RTS Size (Bytes)	20	20	20	20
CTS Size (Bytes)	14	14	14	14
ACK Packet Size (Bytes)	14	14	14	14
CWmin (units of SLOT)	31	15	15	15
CWmax (units of SLOT)	1023	1023	1023	1023
Signal Extension	N/A	N/A	6 µs	6 µs

It is possible that two stations that are not within transmission range of each other can cause collisions at neighboring nodes due to the well known hidden terminal problem [7]. The standard specifies RTS (Request-To-Send) and CTS (Clear-To-Send) message exchanges to reduce these occurrences. In our study, we assume all stations are within radio range of each other and hence dispense with RTS/CTS. ACKtimeout was taken to be the conservative value of SIFS plus time to transmit the ACK at lowest mandatory rate. Table 1 above summarizes the important constants related to the IEEE 802.11b, 802.11a and 802.11g systems. Some comments are in order with respect to the 802.11g system. While 802.11g operates in the same 2.4 GHz band as 802.11b, 802.11b terminals cannot decode the OFDM modulated high rate 802.11g transmissions and hence there can be an increased number of collisions. The 802.11g standard specifies two mechanisms to minimize the cross-talk between high rate 802.11g users and low rate 802.11b users. These mechanisms are initiated as soon as a legacy 802.11b user registers with an 802.11g access point. The last column in the table above reflects the changes induced in the several system variables. The first effect of legacy terminals is to increase the SLOT duration from 9 µs to 20 µs. The second effect is to introduce additional message exchange cycles at the MAC layer which promote peaceful co-existence between 802.11g and 802.11b users. We now describe the two protection methods used at the MAC layer:

CTS-to-self: In this scheme, after the usual sensing time of DIFS and the random back-off time, the sender transmits a CTS message (with its own address) to inform all the neighboring 802.11b nodes of an upcoming packet transmission. Following the CTS message, the sender waits for SIFS duration and then transmits the payload packet and expects an ACK within the SIFS time as usual.

RTS-CTS Exchange: If 802.11g terminals experience significant packet loss in spite of using the CTS-to-self procedure, they have the option of using the full RTS-CTS exchange cycle. Here again, after the initial channel sensing and random back-off, the station sends an RTS message and expects the CTS after a SIFS duration. Once the CTS packet is received, the payload packet is sent after SIFS duration. This RTS/CTS exchange is the standard RTS/CTS exchange.

In both the schemes, the CTS and RTS messages must be sent at a rate understood by all terminals, so 11 Mb/s is the maximum rate for these packets. In addition, the long PLCP header needs to be used. As we would expect, the complete RTS/CTS cycle reduces the capacity of the 802.11g system beyond the CTS-to-self cycle. It is important to note that once either mechanism is completed, the 802.11g terminals use almost the same 54 Mb/s OFDM physical layer as 802.11a terminals. The only difference is the longer SLOT duration of 20 μs.

D. *Physical Layer Model*

The supported data rates are clearly a function of the standard of interest. Although data rate is simply a parameter in our study, the quantitative results we provide are for the highest physical rate supported. IEEE 802.11b physical layer supports 1, 2, 5.5 and 11 Mb/s, all four of which are mandatory. IEEE 802.11a has an OFDM based physical layer and supports 6, 9, 12, 18, 24, 36, 48 and 54 Mb/s, of which 6, 12 and 24 Mb/s are mandatory. IEEE 802.11g physical layer supports data rates of both 802.11b and 802.11a, of which the mandatory rates are 1, 2, 5.5, 11, 6, 12 and 24 Mb/s. We consider that the devices are stationary and the channel introduces no errors, however packets can still be lost due to collisions. Moreover, we assume that all users are in identical RF locations and within range of each other.

III. Capacity Analysis

We now describe the methodology we use to analytically compute the voice capacity of WLANs.

A. *Voice only scenario*

Voice capacity (K) is computed analytically by computing the resources consumed by each call (λ). The fraction of channel time needed to support a single call is taken to be the resource consumed by a call [12]. Roughly speaking, it is the product of number of packets generated per second and average time needed to successfully transmit a packet (including back-off time and other overhead). We show that λ can be computed, to a reasonable approximation, by assuming that the WLAN consists of only the station of interest. Let us denote P_{col} as the collision probability and N_{max} as the maximum number of retransmissions. Similarly, T_P, T_{Layers} and T_{ACK} denote, respectively, the time required for transmitting the raw voice packet, the RTP/UDP/IP/MAC/PHY header and the ACK. D be the codec packetization interval and *DIFS, SIFS, EIFS, CWmin, CWmax* and *SLOT* are as defined in the 802.11b/a standards. Let T_n represent the time for successful reception of a packet, given n retransmissions were needed. If we denote the two users in a voice call as A and B then, $T_{AB}, T_{BA}, T_{BT}, T_{MS}$ are the average durations of "A-talk, B-silent", "B-talk, A-silent", "Both talking", and "Both silent" states and are obtained by solving the steady state probabilities of the Markov chain in [4]. They are found to be 213.5ms, 213.5ms, 67.8ms and 91.2ms respectively. Note that these times are solely a function of the voice traffic model.

The minimum time needed to successfully transmit a packet is $T_0^b = T_P + T_{Layers} + DIFS + SLOT \times \frac{CWmin}{2} + SIFS + T_{ACK}$ for 802.11b. In 802.11g with CTS-to-self, $T_0^c = T_0^b + CTS/R + SIFS$ while in 802.11g with RTS/CTS $T_0^r = T_0^c + RTS/R + SIFS$. The total time for transmission, given n retransmissions are needed can be written in the compact form as:

$$T_n = (n+1)T_0 + \left\{\sum_{k=1}^{n} \min\left\{2^k CWmin, CWmax\right\}\right\} \times \frac{SLOT}{2} + n\left(ACKtimeout - CWmin \times \frac{SLOT}{2}\right)$$

If we take the collisions to be independent, number of tries needed for successful transmission becomes a truncated Geometric random variable. One can compute the average time for successful transmission as by weighting with the probability as:

$$E[T] = \frac{1}{\sum_{n=0}^{N_{max}} P_{col}^n (1-P_{col})} \sum_{n=0}^{N_{max}} T_n \times P_{col}^n (1-P_{col})$$

The average resources consumed by the voice call is obtained by taking the product of number of packets generated in each state of the Markov chain, weighted by the average time needed to transmit a packet under that state. If we fix the time horizon of interest as the sum of mean durations of the four states, normalized channel usage time (resources consumed) is

$$\lambda = \frac{1 \times \left(\frac{T_{AB}}{D}\right) T_0 + 1 \times \left(\frac{T_{BA}}{D}\right) T_0 + 2 \times \left(\frac{T_{BT}}{D}\right) E[T] + 0 \times \left(\frac{T_{MS}}{D}\right)}{T_{AB} + T_{BA} + T_{BT} + T_{MS}}.$$

Where we have used the fact that in BT state, both wired and wireless nodes generate packets. The capacity is then the number of calls that can be supported with each call using a fraction of channel time equal to λ, i.e., $K = \lfloor 1/\lambda \rfloor$. Although P_{col} is in general a complex function of number of users in the system, traffic characteristics, etc. we will approximate it to 1/(*CWmin*+1). The rationale behind this approximation is that collision probability in infrastructure based WLANs is seen to be small, especially for VoIP traffic [13]. Further, the dominant portion of the overhead per transmission is seen to be the protocol headers, back-off time, etc. and not collisions [1]. Note that since we compute λ by assuming that it is the only call in the system, collisions, if they occur, can only occur between the AP and station. Hence, E[T] is used for BT state and T_0 for AB and BA states.

B. *Voice and Video traffic*

So far we have only focused on voice only traffic. We now attempt to address the question: Given there is a CBR video source of R b/s and using packets of size P bits, how many voice calls can be supported in the residual capacity? To that end, we start by computing the resources consumed by the video source (λ_D), as number of packets per second times average time taken per packet, i.e. $(R/P)T_0$. We obtain the residual capacity ($1-\lambda_D$) and from that the voice capacity, $K_V = \lfloor (1-\lambda_D)/\lambda_V \rfloor$. The video traffic is assumed to be from a single source and the protocol headers (RTP, UDP, IP) are taken to be the same for both voice and video traffic.

IV. NUMERICAL RESULTS

In this section, we provide quantitative results for the voice capacity obtained via simulations and analysis. In our simulations, voice capacity refers to the maximum number of calls supported while meeting a round-trip delay budget of 200 ms and a packet loss rate of 2%. In the infrastructure mode, the downlink (AP to mobiles) is the bottleneck and routinely dictates the capacity [2]. All numerical values for the system parameters are provided in Table 1. In addition, we assume the highest channel rate for data packets and ACK packets, i.e., 11 Mb/s for 802.11b and 54 Mb/s for 802.11a and 802.11g. The retry limit was set to 5. In case of 802.11b we use long PLCP header. Further, we do not consider the RTS/CTS exchange for 802.11a and 802.11b. In case of 802.11g with legacy users, we assume that at least one of the legacy users has associated with the access point. However, they do not send any data and hence all the system resources are assumed to be available for 802.11g users. All the capacity results for 802.11g thus denote the maximum number of 802.11g voice calls supported with zero traffic from legacy terminals. Recall that when no legacy terminals have associated with an 802.11g access point, we have an 802.11g only system.

Codec Interval	**802.11b**		**802.11a**		**802.11g only**	
	Simulation	Analysis	Simulation	Analysis	Simulation	Analysis
10 ms	11	11	55	54	55	54
20 ms	21	22	105	102	105	102
30 ms	30	31	149	145	149	145
40 ms	38	39	185	183	185	183
50 ms	44	46	220	217	220	217

Table 2 Voice capacity with no data traffic for 802.11b, 802.11a and 802.11g only systems

A. *Voice only scenario*

In Tables 3 and 4, we provide both analytical and simulation voice capacity results for the different systems as a function of the codec packetization interval. First, we note the good match between analytical results and simulation results. We also immediately observe that larger packetization intervals achieve higher capacity. This result should be intuitive when one considers the overhead incurred in transmitting a single packet. We see that for 10 ms codec the actual payload of 80 bytes takes about 58.2 µs at 11 Mb/s. With approximately 100 packets generated per second λ for a single voice call would be 100x58.2 and capacity should be $\lfloor 1/0.00582 \rfloor$=171 calls. The actual capacity, however, is about 17 times less than this. This large disparity can be explained by taking a more detailed look into the time taken to successfully transmit a packet.

In 802.11b $T_P = 58.2\mu s$, while successful packet transmission requires at least 918 µs. Of the 860 µs overhead, $T_{Layers} = 241\mu s$, $SLOT \times CWmin/2 = 310\mu s$, $T_{ACK} = 248\mu s$, and $DIFS$ + $SIFS$ = 60 µs. The most dominant overhead is the average back-off time, even when the value of CW is at the minimum value. Further, the physical layer header of 192 µs is added twice, for the data packet and for the ACK. Thus, close to 50% of the overhead is due to the physical layer header. When we increase the packetization interval, D increases and the channel efficiency improves (λ decreases). However, as one would expect, the efficiency tends to saturate as the packetization interval is increased. Moreover, increasing the packetization interval beyond 80-100ms would start to drastically interfere with the end-to-end delay budget of 200ms. Comparing the average time needed for successful delivery without collisions (918 µs) with the average time needed after conditioning on retransmissions ($E[T]$ =967 µs), one sees that the cost incurred solely due to retransmissions is rather small. Similar calculations for a 40ms packetization interval gives $T_P = 232.8\mu s$, while the overhead of 860 µs remains the same, resulting in $E[T]$=1.15 ms. Thus, when compared with 10ms codecs, the channel efficiency, defined as $T_p / E[T]$, increased from 6% to 20.2%. In addition, the resource consumption (λ) of a single voice call is reduced approximately by a factor of 3.4, thus immediately explaining the corresponding increase in capacity by going from 10ms to 40ms.

Codec Interval	**802.11g with legacy**			
	CTS-to-self		RTS-CTS	
	Simulation	Analysis	Simulation	Analysis
10 ms	18	20	11	14
20 ms	37	39	24	27
30 ms	56	57	36	41
40 ms	72	75	48	54
50 ms	90	92	60	66

Table 3 Voice capacity of 802.11g system when legacy terminals are present

As a cross-technology comparison, the capacity of 802.11a and 802.11g (without legacy users) systems is seen to be significantly higher than 802.11b. This can be attributed to the fact that the peak rate of 802.11a and 802.11g is approximately 5 times higher than 802.11b, the average back-off time is approximately 1/4th (CWmin is 15 as opposed to 31, and SLOT duration is 9 µs as opposed to 20 µs) and the physical layer header is about 8 times smaller. However, since the minimum contention window is half the size in 802.11b, the probability of collision is twice that of 802.11b. The net effect yields approximately a 5-fold increase in capacity. The results in Table 3 reflect these calculations. It is interesting to note that 802.11g without legacy achieves exactly the same capacity as 802.11a in spite of the shorter SIFS duration (10 µs compared with 16 µs). The reason is that the 6 µs Signal Extension time exactly compensates for the difference in SIFS, resulting in identical overhead per transmission and identical capacities. When legacy terminals are present, both the CTS-to-self and RTS/CTS protection schemes drastically reduce the capacity. This is explained by the increased overhead due to either the CTS with long PLCP header or the RTS/CTS transmissions with long PLCP header. As expected, with the RTS/CTS protection, 802.11g systems achieve a capacity close to the 802.11b system (without RTS/CTS exchange), especially at smaller packetization intervals.

B. *Voice and Video traffic*

The capacity results are summarized in Table 5 for different rates and a voice packetization interval of 30ms. Capacity reduces significantly when video is present. For a constant data rate, decreasing the packet size decreases the inter-arrival time. Based on the analysis of voice only results, it should be apparent that decreasing the packet size results in poor channel efficiency. Hence voice capacity is lowest when the packet size is 500 B and increases as the packet size is increased. However, as noticed in the voice only results, the efficiency tends to saturate as packet size is increased. This is especially clear in the 802.11a results.

Packet Size	Standard	Maximum Number of Voice Calls Supported					
		No Video	1 Mb/s	2 Mb/s	3 Mb/s	4 Mb/s	5 Mb/s
500 B	802.11b	31	22	12	3	0	0
	802.11a	145	136	128	119	110	102
1000 B	802.11b	31	25	19	13	7	1
	802.11a	145	139	134	128	122	116
1500 B	802.11b	31	26	21	16	11	6
	802.11a	145	140	136	131	126	122

Table 4 Voice capacity with CBR video (30ms codec)

The effect of small packet size is obvious for higher streaming rates. For example, no voice calls could be supported in the 802.11b system with a 4 Mb/s streaming rate with 500 B packet sizes. However, 11 voice calls can be supported if a 1500 B packet size is used. The effect of video traffic is much less adverse for 802.11a, primarily because of low transmission overhead and high bandwidth.

V. CONCLUSIONS

In this paper, we have obtained call carrying capacity of 802.11b, 802.11a and 802.11g systems for conversational speech through both simulations and analysis. Based on example calculations, we showed that increasing the packetization interval increases channel efficiency by decreasing the number of packets generated per second, there by reducing the number of times the high overhead is incurred. We found that high bandwidth systems such as 802.11a and 802.11g offer a capacity which is approximately 4 times that of 802.11b and that 802.11g users only system offers the same capacity as 802.11a system. We considered the case of 802.11g terminals co-existing with legacy 802.11b terminals and studied the impact of CTS-to-Self and RTS-CTS protection methods on voice capacity. We found that CTS-to-self procedure dramatically reduces the capacity of 802.11g system and that RTS-CTS procedure further reduces the capacity to values comparable to those of 802.11b system. In all the cases, we obtained a close match between analytical and simulation results. We then considered voice capacity in the case when CBR video traffic is present in conjunction with voice traffic. We found that larger video packets increase the packet inter-arrival times for the same data rate, and hence leave a greater share of resources for voice users.

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