Admission Control for VoIP and Data Traffic in IEEE 802.11 WLANs

(Invited Paper)

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ABSTRACT

This paper presents an novel analytical model for calculation of throughput of the IEEE 802.11 Distributed Coordination Function (DCF) under variable packet length assuming ideal channel conditions. The proposed model considers basic access scheme but can be easily applied to RTS/CTS reservation scheme. Moreover, we use this model to present a simple algorithm to compute the voice capacity of an IEEE 802.11b WLAN and propose a simple admission control algorithm in order to maintain voice quality in a WLAN. Comparison of analytical results produced utilizing our proposed scheme against other admission control algorithms found in the literature shows that our model is fairly accurate.

Categories and Subject Descriptors

C.2.1 [**Computer-Communication Networks**]: Network Architecture and Design – *network communications, wireless communication, distributed networks.*

General Terms

Measurement, Performance.

Keywords

IEEE 802.11; DCF; throughput; variable packet length; VoIP.

1. INTRODUCTION

IEEE 802.11 is the dominant protocol used in wireless local area networks (WLANs). IEEE 802.11 specifies Medium Access Control (MAC) and Physical (PHY) layers for WLANs [1]. IEEE 802.11 MAC specifies a primary mechanism called Distributed Coordination Function (DCF) to access the medium. DCF makes use of the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) scheme and describes two techniques to transmit data packets; a two-way handshaking (DATA - ACK) called basic access and an optional four-way handshaking (RTS - CTS - DATA - ACK) called Request-To-Send/Clear-To-Send (RTS/CTS) access method.

1.1 Related Work

There is a lot of research work in modelling the behavior [2]-[6] as well as improving the performance of IEEE 802.11 DCF [7]-[10]. The bi-dimensional Markov chain modeling, first introduced by Bianchi in [2], has become the most common method for calculating the saturated performance of the IEEE 802.11 protocol. Wu in [3] has built on this model and presented a mathematical analysis that takes into account packet retry limits. In [4] we developed a new performance analysis based on the Markov chain model of [3] that allowed the calculation of the average packet delay and other performance metrics for IEEE 802.11 DCF. Work in [5] and [6] utilizes a different modelling approach of IEEE 802.11 DCF by employing elementary conditional probability arguments rather than bi-dimensional Markov chains.

A lot of research has been conducted on improving the performance of IEEE 802.11 DCF by utilizing many different approaches; nevertheless most of them do not consider Voice over WLAN (VoWLAN). More specifically, in [7] we have extended the mathematical model of [4] by considering packet bursting, a technique in which a station transmits more than one data packets when it gets hold of the medium and, thus, improves considerably protocol performance. In [8] we have studied an appropriate tuning of the backoff algorithm by proposing three sets of parameter values for initial contention window size, retry limit and number of backoff stages in order to achieve better performance on particular metrics for specific communication needs. Banchs in [9] proposed an admission control algorithm that provides end-to-end delay guarantees based on the computation of the cumulative delay distribution function. Cai in [10] proposed a performance analysis under asymmetric traffic and unsaturated conditions in order to compute the voice capacity of a WLAN (defined as the maximum number of voice sessions a WLAN can support simultaneously at specific quality limitations [13]). Nevertheless, the proposed admission control schemes of [9] and [10] are rather complicated. However, all previous works mentioned except of [9], assume that all stations in a wireless network transmit data packets of the same fixed size.

1.2 Overview

The main limitations of all previous proposals that attempt to improve IEEE 802.11 performance is that either they do not study VoWLAN or their approach is rather complex. Furthermore, most of the research work presented in the literature assumes that all stations transmit data packets of the same constant size. However, this assumption is not always true in a realistic environment and

MODENETs 2008, October 24, 2008, Athens, GREECE.
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* Dr. P. Chatzimisios is supported by a post-doctoral fellowship of the State Scholarship Foundation (I.K.Y.) of Greece.

the effect of various packet lengths on performance should be studied.

In this paper, we extend previous works in [2]-[6] and present an accurate mathematical model to calculate throughput performance under variable packet length assuming saturation conditions. The proposed model applies to basic access. The RTS/CTS access is not appropriate for transmission of voice packets as it includes long control frames. Furthermore, we propose a simple admission control algorithm for IEEE 802.11 WLANs that support both voice and data traffic. This admission control algorithm is based on the minimum throughput that a single station is required to have in order to fulfill the application's requirements. Although, the algorithm was evaluated for the mixed scenario (coexistence of voice and data stations), it can be easily applied to any type of traffic such as video and/or audio. Comparison against other admission control algorithms shows that our model is fairly accurate even though less complex and can enhance the provided services for VoWLAN. Finally, we show how the voice capacity is increased by employing the packet length of data stations as a parameter. More specifically, we study the decrease of the packet length of data stations in order to favor voice traffic.

The rest of the paper is outlined as follows. Section 2 briefly provides background information for the Distributed Coordination Function (DCF). Section 3 describes our Markov chain based model and derives mathematical expressions for saturation throughput of IEEE 802.11 MAC. Section 4 provides a simple mathematical analysis for the case of variable packet length. Sections 5 and 6 present the considered VoIP (voice over IP) system and evaluate the performance of the proposed admission control scheme. Finally, Section 7 presents our conclusions and future work.

2. IEEE 802.11 DCF

According to basic access, if a station has a DATA packet to transmit, it detects whether the medium is idle or not. If it detects that the medium is occupied by another station, it defers the packet transmission until the medium becomes available for the transmission. After the station detects that the medium is idle for a certain period of time, which is the DCF Interframe Space (DIFS), it starts a backoff operation with a randomly-selected backoff counter value. This random backoff counter is stopped when the channel is busy, is resumed when the channel is sensed idle again for more than DIFS and is decreased by one for each idle slot time. When the backoff counter becomes zero, the station transmits the data packet. When the destination station receives this data packet successfully, it transmits an ACK packet back to the source station after a Short Interframe Space (SIFS) interval. If the transmitting station does not receive an ACK packet within a specified ACK timeout interval, the data packet is assumed to have been lost and the station schedules a retransmission. When the source station receives the ACK packet, the transmission operation of that packet is finally completed. The IEEE 802.11 standard also specifies retransmission limits for each packet; Each station holds a retry counter that is increased by one each time a data packet is unsuccessfully transmitted. If the counter reaches the retransmission limit *R* the packet is discarded.

The backoff time counter is chosen uniformly in the range [0,W_i-1], where *i* is the backoff stage and Wi is the current contention window (*CW*) size. The contention window at the first transmission of a packet is set equal to $CW_{min}=W$. After an unsuccessful packet transmission the contention window *CW* is doubled up to a maximum value $CW_{max}=2^m W$ (where m is the number of *CW* sizes). Once *CW* reaches CW_{max} it remains in this value until it is reset. The *CW* is reset to CW_{min} after a successful packet transmission or if the packet's retransmission limit *R* is reached.

3. MATHEMATICAL ANALYSIS

We assume that: (i) the network consists of n contending stations under ideal channel conditions (no transmission errors and no hidden stations exist), (ii) each station has always a packet available for transmission (saturation conditions).

3.1 Markov Chain Model

Let *b(t)* and *s(t)* be the stochastic processes representing the backoff time counter and the backoff stage (0,…*R*) respectively for a given station at time *t*.

We utilize the same discrete-time Markov chain with [3] in order to model the two-dimensional process {*s(t)*, *b(t)*}. The key approximation in this model is that each packet transmission collides with constant and independent probability *p* regardless of the backoff stage. The probability τ that a station transmits a packet in a randomly chosen slot time can be expressed as:

$$\tau = \frac{2\cdot(1-2p)\cdot(1-p^{R+1})}{W\cdot(1-(2p)^{m+1})\cdot(1-p)+(1-2p)\cdot[(1-p^{R+1})+W\cdot 2^m\cdot p^{m+1}\cdot(1-p^{R-m})]} \quad \text{for } R \geq m \quad (1)$$

The probability *p* that a transmitted packet encounters a collision is given by:

$$p = 1-(1-\tau)^{n-1} \quad (2)$$

Equations (1) and (2) represent a non-linear system with two unknown τ and *p*, which can be solved using numerical methods and has a unique solution.

3.2 Saturation Throughput

Let q_{tr} be the probability with that at least one station (out of *n*) transmits in a considered slot time:

$$q_{tr} = 1-(1-\tau)^n \quad (3)$$

Let q_s be the probability that an ongoing transmission is successful. This can be easily computed as the probability that only one station transmits (the *n*-1 remaining stations remain silent), given the condition that a transmission occurs on the channel:

$$q_s = \frac{n\cdot\tau\cdot(1-\tau)^{n-1}}{q_{tr}} = \frac{n\cdot\tau\cdot(1-\tau)^{n-1}}{1-(1-\tau)^n} \quad (4)$$

The average length of a slot time *E*[*slot*] is equal to:

$$E[slot] = (1-q_{tr})\sigma + q_{tr}q_s T_s + q_{tr}(1-q_s)T_c \quad (5)$$

where σ is the duration of an empty slot, and T_s and T_c are the time durations the channel is sensed busy during a successful transmission and a collision, respectively.

The time duration of T_s and T_c depends on the channel access method employed. For the basic access method we have:

$$T_s^{bas} = T_c^{bas} = O^{bas} + (l/C) \quad (6)$$

where O^{bas} is the associated overhead for basic access scheme:

$$O^{bas} = DIFS + T_H + 2\delta + SIFS + T_{ACK} \quad (7)$$

l is the packet length, T_H, and T_{ACK} represent the transmission time for the header (equal to the sum of MAC and physical header), and ACK, respectively; δ is the propagation delay and C is the channel bit rate.

The saturation throughput S is calculated as the ratio of the successfully transmitted payload information in a slot time:

$$S = \frac{q_{tr} \cdot q_s \cdot E[l]}{E[slot]} = \frac{q_{tr} \cdot q_s \cdot E[l]}{(1-q_{tr})\sigma + q_{tr} q_s T_s + q_{tr}(1-q_s) T_c} \quad (8)$$

where $E[l]$ is the average packet length when stations transmit packets of different size. If all stations transmit the same fixed packet length then $E[l]$ is equal to l.

4. VARIABLE PACKET LENGTH

We consider that a packet length takes a value l of the set L with probability $P_L(l)$, where L is the set of all possible packet lengths with corresponding probabilities in the set P_L. For simplicity, we assume that all stations pick the packet length from the same distribution set $(l, P_L(l))$ (the analysis would be similar in the case when the stations pick the packet length from different distribution sets).

Let $E[l]$ be the weighted average packet length of all possible l in the set L. $E[l]$ is computed as following:

$$E[l] = \sum_{l \in L} l \cdot P_L(l) \quad (9)$$

Let P_k be the probability that exactly k stations are involved in one collision:

$$P_k = \binom{n}{k} \cdot \frac{\tau^k (1-\tau)^{n-k}}{q_{tr}(1-q_s)} \quad \text{for } k \geq 2 \quad (10)$$

Let P_y be the sum of probabilities that corresponds to packet lengths that are shorter or equal to the longest data packet l in a collision:

$$P_y(l) = \sum_{h \in L | h < l} P_L(h) \quad (11)$$

Let $P_{l,k}$ be the probability that a packet with length l is the longest data packet in a collision when k stations are involved in a collision:

$$P_{l,k} = \sum_{r=1}^{k} (-1)^{r+1} \binom{k}{r} (P_L(l))^r P_y(l)^{k-r} \quad \text{for } 2 \leq k \leq n \quad (12)$$

Utilizing equation (12) we can compute the probability that any number of stations can be involved in one collision.

We calculate next the time durations T_s and T_c for the basic access scheme. The time T_s for a successful transmission is the sum of packet's overhead O^{bas} plus the average payload:

$$T_s = O^{bas} + E[l]/C \quad (13)$$

In the basic access collisions occur between data packets. The time duration of a collision is the time of the longest data packet involved in a collision:

$$T_c = O^{bas} + \frac{1}{C} \cdot \sum_{l \in L} \left(l \cdot \sum_{k=2}^{n} P_k P_{l,k} \right) \quad (14)$$

The partial throughput S_l for stations that transmit packets of length l is computed as follows:

$$S_l(l) = S \cdot \frac{l \cdot P_L(l)}{\sum_{l \in L} l \cdot P_L(l)} \quad \text{or} \quad S_l(l) = \frac{q_{tr} \cdot q_s \cdot l \cdot P_L(l)}{E[slot]} \quad (15)$$

where S is the saturation throughput of (8). The T_s and T_c times of (8) are computed with (13) and (14) respectively.

The throughput of a single station S_s that transmits packets of length l is:

$$S_s(l) = \frac{S_l(l)}{n \cdot P_L(l)} \quad (16)$$

In the case where all stations in a WLAN transmit packets of the same length, the throughput of a single station S_s is:

$$S_s = S / n \quad (17)$$

5. VALIDATION

The model is validated by comparing the analytical results with that taken from simulation outcome. The parameter values used for both simulation and analytical results follow the values specified for the Direct Spread Sequence Spectrum (DSSS) employed in the IEEE 802.11b standard and are shown in Table 1.

Table 1. System parameter values

Channel bit rate C	11Mbit/s
MAC header	224 bits at 11Mbit/s
Physical header PHY	192 bits at 1Mbit/s
ACK	112 bits at 11Mbit/s+PHY
Propagation delay, δ	1 µs
Slot time, σ	20 µs
SIFS	10 µs
DIFS	50 µs
Minimum CW, W_0	32
Number of CW sizes, m	5
Retransmission limit, R	6

The packet length is taken from the set $L = [1000, 8000, 12000]$ bits with corresponding probabilities $P_L = [0.7, 0.2, 0.1]$.

Figure 1 plots the saturation throughput (total and for packets of 1000, 8000, 12000 bits long) versus number of stations for basic access. The model is accurate as the analytical results (lines) match the simulation results (symbols). All simulation results are taken with a 95% confidence interval lower than 0.005, while the difference between the analytical results and the simulation results is lower than 1%. The control frames are transmitted at 1Mbit/s and the data packets at 11Mbit/s.

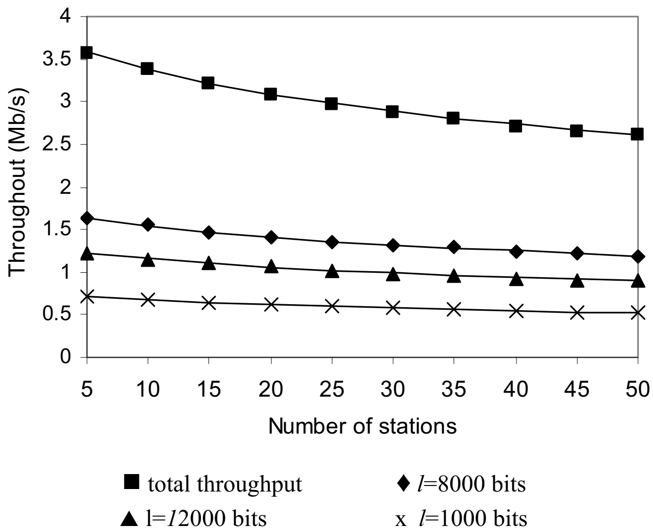


Figure 1. Saturation throughput vs. number of stations.

Table 2. Popular Voice Codecs

Voice codec:		*G.711*	*G.729*	*G.723.1*
Bit rate:		***64Kbps***	***8Kbps***	***5.3/6.3Kbps***
Sample period (ms)	***frames/sec***	***Payload (byte)***	***Payload (byte)***	***Payload (byte)***
10	100	80	10	
20	50	160	20	
30	33.33	240	30	20/24
40	25	320	40	
50	20	400	50	
60	16.67	480	60	40/48

6. VoIP

The Voice over Internet Protocol (VoIP) technology allows you to make telephone calls via an Internet connection instead of a regular phone line. Wireless VoIP combines VoIP with wireless networks. A VoIP component called codec converts an analog voice signals to digital voice data. The digital voice data are then packed into voice packets at a Constant-Bit-Rate (CBR) or Variable-Bit-Rate (VBR) by the packetizer component. In this paper we only consider CBR.

Table 2 lists some of the most popular codecs and their corresponding packetization intervals. For instance, a G.711 codec samples the voice at a fixed bit rate of 64Kbps. For packetization interval of 10ms generates 100 packets/sec which corresponds to 80 bytes (64000/(100*8) of payload. Moreover, a 40 bytes RTP/UDP/IP header is attached to each voice packet.

6.1 Model

We consider the following configuration of the IEEE 802.11 DCF stations:

- Stations can either transmit voice (voice stations) or data packets (data stations)
- Voice stations generate payload packets l_v according to the voice codecs of Table 2
- Data stations have always a packet ready for transmission (saturation conditions). The maximum data packets payload is 1500 bytes.
- The basic access mode is used for both voice and data packets.
- Physical header is transmitted utilizing a data rate of 1 Mbit/s. On the other hand, ACK packets, MAC and RTP/UDP/IP headers are transmitted at 11 Mbit/s.
- Voice and data payload are transmitted at 11Mbit/s.

For simplicity, we consider that all voice stations operate with the same codec, bit rate and the same sample period.

6.2 Admission control algorithm

Real-time applications such as voice applications require that all or most of the packets arrive at the destination with a delay below a maximum value. This maximum delay is the time needed for a codec to prepare a new voice packet. If the sampling bit rate of a codec is at 64Kbps and the packetization interval is 20ms then the maximum delay of a packet to reach destination should be 20ms. In such case the throughput of a single voice station should be 64Kbps or higher. In the proposed analysis we consider codec's sampling bit rate as a threshold value. If the throughput of a single voice station exceeds this threshold value then the quality of the transmitted voice is acceptable.

We consider a WLAN is operating with n stations N_v of them are voice and N_d data stations ($n = N_v + N_d$). We proceed as follows in the case where a new voice or data station issues a request for admission. We compute the Ts (13), Tc (14) and the throughput for n+1 stations (8); and then the partial throughput for voice stations (15) and finally, the throughput of a single voice station (16). If the throughput of a single voice station is higher or equal to codec's sampling bit rate then the new station will be admitted otherwise not.

6.3 Performance evaluation

In this section we evaluate our proposed admission control algorithm. The parameter values used follow the values specified for the Direct Spread Sequence Spectrum (DSSS) employed in the IEEE 802.11b standard and are shown in Table 1.

6.3.1 Only voice stations

In this scenario all stations are voice stations and transmit voice packets of length l_v. We compute the throughput (8) using Ts and Tc of (6) or we compute Ts (13) and Tc (14) with $L = [l_v]$ and $P_L = [1.0]$ and then the throughput (8), (15) and (16).

Table 3 tabulates the capacity of voice sessions that our admission control algorithm computes for 802.11b WLANs and for the most popular codecs. Table 3 shows also results for voice capacity of [10] and [11]. Our algorithm performs pretty well in comparison to [10] and [11] up to packetazation interval of 40ms. Although [10] use different network model than our model, we get results close to [10] as far the AP in [10] can serve voice traffic with no packets in queue. For packetization intervals 50, 60 ms the capacity in [10] and [11] are higher than our results as they consider that longer delays than 50-60 ms still give good voice quality.

Table 3. Comparison of maximum capacity of voice sessions for 801.11b

Voice (ms)	*G.711*			*G.729*			*G.723.1*		
	PA	[10]	[11]	*PA*	[10]	[11]	*PA*	[10]	[11]
10	6	6	6	7	6	7			
20	11	11	12	13	13	14			
30	15	15	17	19	19	21	19	19	21
40	18	19	21	23	25	28			
50	20	22	25	28	31	34			
60	22	25	28	32	37	41	33	37	42

PA=proposed algorithm

6.3.2 One voice station

In this scenario we have only one voice station (N_v=1) and none data station ($N_d = n$-1) at the beginning. We increase gradually the data stations in order to see how many data stations can be admitted without to influence the voice quality. The voice packet payload is l_v = 80 bytes (G.711, 64Kbps, sample period of 10ms), and the data payload is l_d =1470 bytes. Figure 2 plots the throughput of the voice station versus number of stations. The throughput is decreasing rapidly as the number of data stations are increasing.

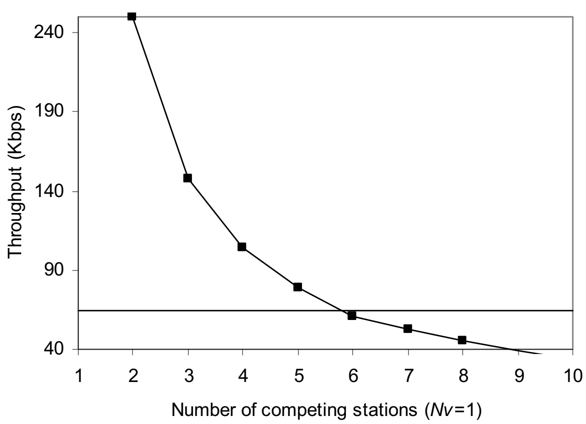


Figure 2. Throughput of a voice station as the competing data stations are increasing.

The results show that a 802.11b WLAN can support at maximum four data stations and one voice station with acceptable quality. If more than four data stations get admitted to the WLAN then the throughput falls below 64Kbps with immediate influence to voice quality.. This outcome is in agreement with an experiment in [12]. The straight line represents the threshold of 64Kbps.

6.3.3 Voice and data stations

In this section we consider a WLAN scenario with 75% of the stations are *voice* stations and the remaining 25% are *data* stations. Figure 3 shows how the throughput per voice station is decreasing as the voice and data stations are increasing according to 75%-25% above scenario. The figure shows that only 20 stations can be admitted (15 voice stations and 5 data stations) to maintain voice quality. If more than 20 stations get admitted then the quality of voice calls will degrade. The voice packet payload is l_v = 240 bytes (G.711, 64Kbps, sample period of 30ms), the data payload is l_d =1500bytes, the set L is $L=[l_v,l_d]$ with corresponding probabilities $P_L=[0.75, 0.25]$.

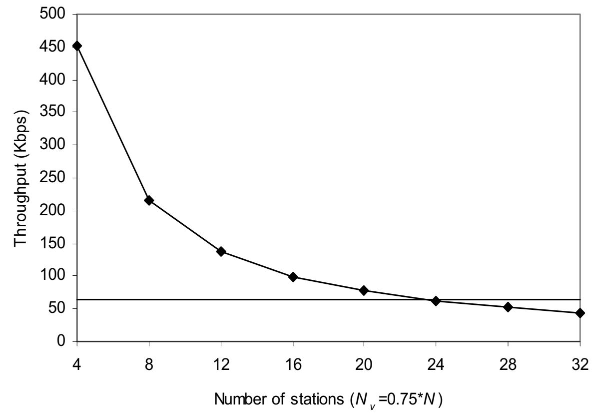


Figure 3. Throughput per voice station as the number of voice and data stations are increasing (voice scenario).

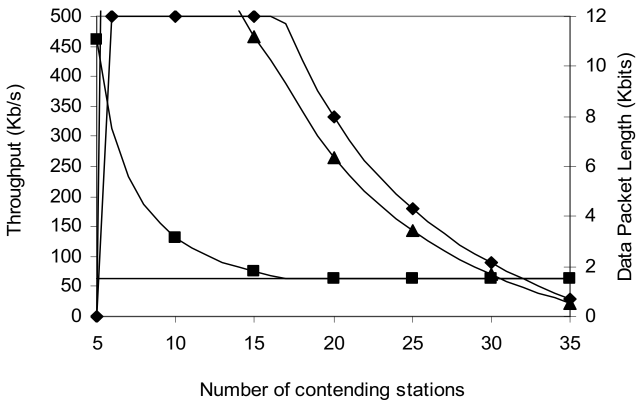
In [9] the authors consider three scenarios: *voice* scenario, with 75% of voice stations and 25% of data stations, *mixed* scenario, with 50% of the stations are voice and the other 50% data, and *data* scenario, with 75% of data stations and 25% of voice stations. We compare our proposed admission control algorithm with that found in [9], see Table 4. The codec used is the GSM 06.10 at bit rate of 13 Kbit/s, payload 260 bits, and sample period of 20 ms. Authors in [9] employ RTS/CTS combined with packet bursting (transmitting more one packets when gaining access on the medium) in order to improve voice performance. However, our algorithm gives the same results without employing packet bursting (the results of [9] in Table 4 correspond to probability P=0.9 and delay D=20ms). In order to get the comparison results we have used l_v = 260/8 bytes l_d=1500bytes the set $L=[l_v,l_d]$ with corresponding probabilities $P_L=[0.75,0.25]$ voice scenario, $P_L=[0.25,0.75]$ data scenario and $P_L=[0.5,0.5]$ mixed scenario.

Table 4. Comparison of maximum voice and data stations in different scenarios.

GSM 6.10		
Scenario	*PA*	*[9]*
Voice	16	16
Data	8	8
Mixed	12	12

PA=proposed algorithm

Our approach for increasing the number of the admitted voice stations is to reduce the packet length of the data stations. As the throughput requirement of a voice station (Table 2) is known, we use equations (15) and (16) for computing the suitable data packet length in order to admit new voice or data stations keeping steady the voice quality.



Throughput: ▲ per data station ■ per voice station.

♦ Packet length of the data stations

Figure 4. Throughput per voice station and per data station for fixed number of voice stations (N_v=5) and increasing the number of data stations.

Figure 4 plots throughput per voice station and per data station for fixed number of voice stations N_v=5 and increasing number of data stations. The figure also shows how the data station's packet length is reduced in order to maintain the quality of the voice stations as the data stations are increasing. The voice stations use G.711 at 30ms (l_v =1920 bits) and the data stations use payload l_d =12000 bits. For n=5 there are only voice stations and no data stations. Up to 16 contending stations (N_v=5 and N_d=11) the throughput of the data stations is very high and the throughput of the voice stations is higher than 64 Kbit/s and, thus, provides good voice quality. In order to increase the number of admitted data stations and at the same time to keep good quality of the five voice stations, we reduce the packet length of the data stations. For n=17 up to n=35 we reduce the packet length of the data stations resulting in the degradation of the throughput per data station nevertheless maintaining good voice quality since the throughput per voice station is kept stable at 64 Kbit/s. For n=30 the packet length of data stations is almost equal to a voice station's packet length (that is 15 voice sessions, which is in agreement with values in Table 2).

7. CONCLUSIONS AND FUTURE WORK

This paper proposes a simple but effective call admission control algorithm that is based on throughput analysis of IEEE 802.11 DCF with variable packet length under saturation conditions. We present simple mathematical relations that can compute the throughput of a single station in a WLAN where the stations transmit packets of variable length. The proposed admission control algorithm is based on the minimum throughput that a single station of a WLAN is required to have in order to fulfill the application's throughput requirements. The algorithm was evaluated with voice and data stations but can be applied to any type of traffic as video and/or audio. Comparison with other proposed admission control algorithms shows that our presented model is simple but at the same time provides fairly accurate results.

More sophisticated admission control algorithms that include additional metrics as average packet delay and delay jitter could be a future research direction. Moreover, our proposed approach could be combined with the Mean Score Opinion (MOS) in order to provide a measure of the quality of voice over WLANs.

8. ACKNOWLEDGMENTS

This work is funded by the Greek Ministry of Education (25%) and European Union (75%) under the EPEAEK II program "Archimedes".

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