Voice capacity of IEEE 802.11b WLANs

D. S. Amanatiadis[1], V. Vitsas[1], A. Manitsaris[2], I. Mavridis[2], P. Chatzimisios[1] and A.C. Boucouvalas[3]

[1]Department of Information Technology
Technological Educational Institution, Thessaloniki, Greece,
{dima, vitsas, peris}@it.teithe.gr

[2]Department of Applied Informatics
University of Macedonia, Thessaloniki, Greece,
{manits, mavridis}@uom.gr

[3]School Design, Engineering & Computing
Bournemouth University
Fern Barrow, Poole, UK
tboucouv@bournemouth.ac.uk

***Abstract*-There is a tremendous growth in the deployment and usage of Wireless Local Area Networks (WLANs) in recent years. There is also an increasing interest in supporting voice applications over WLANs. In this paper we use analytical models presented in the literature for saturation throughput, average packet delay and packet drop probability and simulative results for jitter to evaluate voice traffic over WLANs. We propose a methodology that determines the voice capacity of WLANs and we examine the effect of packetization interval and transmission rate on voice capacity using G.711 codec. Analytical results for voice capacity match simulative results presented in the literature using extensive simulations.**

I. INTRODUCTION

Over the past few years the popularity of Wireless LANs has grown significantly. WLANs provide principal value in environments with a large and dynamic set of end users allowing easy access to the network. The main reasons for the success of WLANs are the simplicity of setup, high data rates and shrinking equipment costs. Advances in technology are enabling the delivery of laptop computers with enhanced and integrated WLAN capabilities.

Recently, there has been growing interest in using WLANs to support voice communications, especially in an office environment. Thus, employees can be engaged in voice communications while still being mobile. Furthermore, as WLAN applications share a common transmission medium, voice applications must compete with other voice applications as well as with data applications for bandwidth. Therefore, voice quality and capacity can be significantly affected by the simultaneous transmission of data and other voice traffic in WLANs.

IEEE has developed the 802.11 [1] protocol family for WLANs. IEEE 802.11b [2] is the most popular and operates in the 2.4 GHz Industrial/Scientific/Medical (ISM) band with data rates up to 11 Mb/s. IEEE 802.11a [3] operates at higher data rates (up to 54 Mb/s) in the less crowded 5 GHz ISM band. The IEEE 802.11 standard defines two modes of Medium Access Control (MAC) operation: a mandatory Distributed Coordination Function (DCF) mode and an optional Point Coordination Function (PCF) mode. DCF allows for highly distributed medium access using Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) techniques. PCF uses a polling mechanism controlled by a point coordinator operating at the Access Point (AP) of each Basic Service Set (BSS). The PCF mode was designed for real-time services such as voice. Many manufacturers are choosing not to implement PCF, claiming that it inhibits interoperability with other APs [4] and does not, in fact, always allocate bandwidth better than DCF [5]. Furthermore, authors in [4], claim that PCF provides marginal improvement in performance when compared with DCF for real-time services such as voice.

In addition to the MAC modes, the standard also specifies two network configurations called the Infrastructure and Ad Hoc. The Infrastructure mode requires one node to coordinate all communications and is based on PCF, while the Ad Hoc mode provides direct communication between peers and is based on DCF. Evaluating voice traffic under DCF Ad Hoc configuration is the subject of our interest.

As defined in the literature, voice capacity is the maximum number of voice sessions a WLAN can support simultaneously at specific quality limitations [6]. Experimental and analytical results for voice quality using DCF in Infrastructure configuration have first been reported in [7]. Authors in [7] proposed an approximate analytical model which notably illustrates the importance of choosing a voice packet size that is as large as possible. The voice capacity of a single AP is quantified in [6] which also examines the effect of codec and packetization intervals on voice capacity for IEEE 802.11b networks. In [8] the speech capacity is evaluated using the same conversational model as in [6], comparing analysis and simulation for 802.11a, 802.11b and 802.11g. It was concluded in [8] that channel capacity strongly depends on the channel bandwidth, voice codec packetization interval and the data traffic in the system. Authors in [9] proposed solutions to performance problems using a multiplex-multicast scheme for overcoming the large overhead effect of VoIP over WLAN. In [10] the co-existence of data and voice is being investigated. An analysis of the performance is provided for WLANs which use TCP/IP for data and UDP/IP protocol for the voice packets.

A mathematical analysis based on a Markov chain model has been developed in [11] for a WLAN operating in *saturation* conditions, i.e., the transmission queue of each station is assumed to be always nonempty. The analysis calculates the saturation throughput which is defined as the maximum load that the system can carry in stable conditions [11] and evaluates the performance for different scenarios and parameter values for unlimited packet retries. The throughput model is extended in [12] to take into account packet dropping when the packet retransmission limit is reached. In [13] the same model is expanded to calculate average packet delay, packet drop probability and the packet drop time.

Research work presented in the literature [6], [7], [8] calculates the voice capacity using extensive simulations. As analytical models are much easier to use, this work calculates the voice capacity of a WLAN based on the mathematical model of [12], which calculates the maximum WLAN throughput performance assuming saturation conditions. An ad hoc network scenario consisting of a finite number of stations is assumed. We also assume ideal physical channel conditions, (no transmission errors, no hidden stations) and that all stations transmit packets of constant length. Analytical models presented in [13] that calculate average packet delay and packet drop probability are utilized to examine whether the voice quality requirements are met for all considered scenarios. Our analysis identifies that, in ad hoc WLAN voice communications, the average packet delay should not exceed the packetization interval, i.e. a voice packet should be successfully received before a new voice packet is generated.

This paper is organized as follows: Section II describes the DCF method for channel access and the backoff procedure. Section III reviews the mathematical model that calculates throughput, average delay and drop probability. Section IV describes the considered network scenario, the voice quality metrics and the methodology utilized to calculate voice capacity. Analytical results for voice capacity and comparison with results presented in the literature are placed in Section V. Finally, section VI presents the conclusions.

II. Distributed Coordination Function (DCF)

DCF is based on CSMA/CA techniques and adopts a slotted Binary Exponential Backoff (BEB) scheme to reduce collisions due to stations transmitting simultaneously. Each node with a packet to transmit first senses the medium to ascertain whether it is in use. If the medium is sensed to be idle for a time interval greater than the Distributed Inter-Frame Space (DIFS), the station proceeds with the packet transmission. If the medium is sensed busy, the station defers transmission and initializes its random backoff interval. This backoff timer is decremented when the medium is idle and is frozen when the medium is sensed busy. After a busy period, the backoff resumes only after the medium has been idle for longer than DIFS. A station transmits its packet when its backoff counter reaches zero.

After the successful reception of a packet, the destination station replies with an immediate positive acknowledgment (ACK) after a time interval equal to Short Inter-Frame Space (SIFS). Since SIFS is shorter than DIFS, the station sending the ACK transmits before stations attempting to send new packets and hence takes priority. If the source station does not receive an ACK packet within a specified ACK timeout interval, the data packet is assumed to have been lost and a retransmission is scheduled. Each station holds a retry counter that is increased by one each time a data packet is transmitted unsuccessfully. If the counter reaches the retransmission limit m the packet is discarded.

The random number of empty slots for the backoff timer is chosen in the interval $[0,W_i-1]$, where i is the backoff stage, $i\in[0,m]$ and W_i is the current contention window (CW) size. The value of CW depends on the number of failed transmissions of a packet. At the first transmission attempt, CW is set equal to W which is the minimum contention window size $CW_{min}=W=W_0$. After each retransmission due to a collision, CW is doubled up to a maximum value $W_{m'}=CW_{max}=2^{m'}\cdot CW_{min}$ where $W_{m'}$ is the largest contention window size. Once the CW reaches CW_{max}, it will remain at this value until it is reset. We have:

$$\begin{cases} W_i = 2^i \cdot W & i \le m' \\ W_i = 2^{m'} \cdot W & i > m' \end{cases} \tag{1}$$

Here m represents the maximum backoff stage while m' represents the number of different window sizes. The model considers that m can have a value larger or smaller than m'. The 802.11b standard [2] specifies that $m = 6$ and $m' = 5$.

III. Mathematical Model

This study assumes ideal channel conditions (no transmission errors or hidden stations), the contenting stations are of fixed number n and each station has always a packet available for transmission of the same fixed size.

Let $b(t)$ and $s(t)$ be the stochastic processes representing the backoff time counter and the backoff stage $(0,\ldots m)$ respectively for a given station at time t. We utilize the same discrete-time Markov chain with [12][13] in order to model the bi-dimensional process $\{s(t), b(t)\}$. The key approximation in this model is that each packet collides with constant and independent probability p regardless of the number of retransmissions suffered. Let $b_{i,k} = \lim_{t\to\infty} P\{s(t)=i, b(t)=k\}$ be the stationary distribution of the Markov chain, where $i\in[0,m]$, $k\in[0,W_i-1]$. The probability τ that a station transmits a packet in a randomly chosen slot time can be expressed as [12]:

$$\tau=\begin{cases} \dfrac{2\cdot(1-2p)\cdot(1-p^{m+1})}{W\cdot(1-(2p)^{m+1})\cdot(1-p)+(1-2p)\cdot(1-p^{m+1})} & , m\le m' \\[2ex] \dfrac{2\cdot(1-2p)\cdot(1-p^{m+1})}{W\cdot(1-(2p)^{m'+1})\cdot(1-p)+(1-2p)\cdot(1-p^{m+1})+W\cdot 2^{m'}\cdot p^{m'+1}\cdot(1-2p)\cdot(1-p^{m-m'})} & , m>m' \end{cases} \tag{2}$$

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

$$p = 1-(1-\tau)^{n-1} \tag{3}$$

Equations (2) and (3) represent a non-linear system with two unknown τ and p, which can be solved using numerical methods and has a unique solution [12].

A. Saturation Throughput

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

$$P_{tr} = 1 - (1-\tau)^n \tag{4}$$

The probability P_s that an occurring packet transmission is successful is given by the probability that exactly one station transmits and the remaining n-1 stations defer transmission, conditioned on the fact that at least one station transmits:

$$P_s = \frac{n \cdot \tau \cdot (1-\tau)^{n-1}}{1-(1-\tau)^n} \tag{5}$$

Considering that a random slot is empty with probability $(1-P_{tr})$, contains a successful transmission with probability $P_{tr} \cdot P_S$ and a collision with probability $P_{tr} \cdot (1-P_S)$, the saturation throughput S is given by:

$$S = \frac{P_{tr} \cdot P_S \cdot l}{(1-P_{tr}) \cdot \sigma + P_{tr} \cdot P_S \cdot T_S + P_{tr} \cdot (1-P_S) \cdot T_C} \tag{6}$$

where l is the length of the transmitted packet, σ the duration of an empty slot time, T_s and T_c are the average times that the medium is sensed busy due a successful transmission or a collision respectively. The values of T_s and T_c depend on the channel access mechanism and in the case of basic access:

$$\begin{cases} T_S^{bas} = DIFS + H + l + SIFS + ACK \\ T_C^{bas} = DIFS + H + l + SIFS + ACK \end{cases}$$

where $H = MAC_{hdr} + PHY_{hdr}$ is the packet header.

B. Packet drop probability

The packet drop probability is defined, as the probability that a packet is dropped when the retry limit is reached and it is equal to:

$$P_{drop} = p^{m+1} \tag{7}$$

since a packet is dropped if it encounters m+1 collisions.

C. Average packet delay

The delay D for a successfully transmitted packet is defined to be the time interval from the time the packet is at the head of its MAC queue ready to be transmitted, until an acknowledgement for this packet is received. If a packet is dropped because it has reached the specified retry limit, the delay time for this packet will not be included into the calculation of the average delay.

The average packet delay $E[D]$, is given by:

$$E[D] = E[X] \cdot E[slot] \tag{8}$$

where $E[slot]$ is the denominator of (6) and $E[X]$ is the average number of slot times required for successfully transmitting a packet and is given by:

$$E[X] = \sum_{i=0}^{m} \left[\frac{W_i + 1}{2} \cdot \frac{(p^i - p^{m+1}) \cdot}{1 - p^{m+1}} \right] \tag{9}$$

where $(1-p^{m+1})$ is the probability that the packet is not dropped and $(p^i - p^{m+1})/(1-p^{m+1})$ is the probability that a packet that is not dropped reaches the i stage.

IV. Network system model

A. Description

We consider an 802.11b DCF system in ad hoc mode. We assume full duplex VoIP calls between the stations. We also assume that:

- the physical channel conditions are ideal, (no transmission errors, and no hidden stations) and we ignore all propagation delays.
- the source always has a packet to send, either a voice packet or a data packet of the same size,
- all stations are within range of each other and hence achieve the maximum data rate.

Voice is typically sampled at a fixed rate, and collected bits are assembled into packets at fixed intervals. The voice codec used is the G.711 codec which generates voice packets at a Constant Bit Rate (CBR) of 64 kbps and assemble packets every 10, 20, 30 or 40 ms, resulting in packets with payloads of 80, 160, 240 and 320 bytes, respectively. Each call results in two Real-time Transport Protocol (RTP) streams. Each packet is attached a 68-byte packet header (12 bytes RTP, 8 bytes UDP, 20 bytes IP, and 28 bytes MAC layer header). Each MAC frame has a physical layer header (preamble) of 24 bytes, transmitted always at 1 Mbps. The MAC layer acknowledgement (ACK) is 14 bytes long and is transmitted at variable rate.

The time required to transmit the packet payload depends on the size of the packet, and thus on the packetization interval of the voice codec. For example if voice is generated at 64 kbps and packetized every 10 ms, each packet payload is 640 bits and takes 58 μs to transmit at 11 Mbps rate. While the headers (RTP, UDP, IP, MAC) are 544 bits and ACK is 112 bits, the additional time needed to transmit them at 11 Mbps is almost 60 μs. In addition, we need 60 μs for DIFS+SIFS and 192 μs for transmitting the physical packet headers and 192 μs for the physical packet headers of the MAC ACK packet. It is easily seen that for 80 bytes (58 μs) of payload, the overhead takes about ten times the time needed to transmit this packet at 11 Mbps. Thus, when voice is packetized less frequently, the result is proportionately larger payloads and more time to transmit, but the number of packets required to convey a fixed payload scales inversely. Therefore much of the packet overhead also scales inversely with respect to the payload when packetization intervals are increased, and the medium is used more efficiently.

B. Quality metrics

The human ear is sensitive to voice time delays larger than 200 ms [6, 10]. Thus, if a voice packet is delayed more 200 ms, it should be considered as a lost packet. In addition, a packet is dropped when the retransmission limit is reached. Caputo in [14] defines the packet drop rate limit, the delay limit and the jitter (in literature it is defined also as delay variation) which are tabulated in table I for good voice quality.

TABLE I
REQUIREMENTS FOR GOOD VOICE QUALITY

Requirement	Limit
Delay	150 ms
Jitter	75 ms
Drop rate	3 %

In our model we take in account these quality metrics as these values are commonly accepted by many vendors and researchers.

C. Methodology

In this paper, we try to calculate the maximum number of VoIP clients which can be supported with good voice quality in IEEE WLANs. As defined in the literature, capacity for voice traffic is the maximum number of voice sessions than can be supported simultaneously at specific quality limitations [6]. We use the mathematical model presented in section III to calculate average packet delay and packet drop probability and examine whether the voice quality requirements presented in section IV-B are met.

We calculate voice capacity assuming that the average packet delay should not exceed the threshold of packetization interval. In other words, a voice packet should be successfully received before a new packet is generated i.e. if the packetization interval is 30 ms, the maximum number of stations that have average packet delay less than 30 ms is the voice capacity of the considered scenario. This number depends on the maximum throughput the channel can achieve, which is a function of the packet size. Other factors affecting the channel throughput include the packet overheads and the transmission rate.

V. RESULTS

Unless otherwise specified, the values reported in the following figures have been obtained using the system parameters in table II and are based on the Direct Spread Sequence Spectrum (DSSS) physical layer used in 802.11b standard.

TABLE II
IMPORTANT PARAMETERS IN 802.11B

Parameter	Value
Slot	20 μs
SIFS	10 μs
DIFS	50 μs
PHY layer header	192 bits
MAC layer header	224 bits
ACK packet	112 bits + PHY layer header
CW_{min}	32
Number of CW sizes, m′	5
Retry limit	6
Packet payload	Depends on codec
Channel bit rate	1, 2, 5.5, 11 Mbps

Fig. 1 plots the average packet delay against the number of stations for different packetization intervals using G.711. In the figure, the horizontal lines are the limits for packets assembled every 10, 20, 30 and 40 ms. The point where the packet delay for a specific packetization interval crosses the horizontal line of this packetization interval indicates the voice capacity of the WLAN if the G.711 codec implements the specific packetization interval. Voice capacity values are tabulated in Table III.

TABLE III
MAXIMUM NUMBER OF VOICE STATIONS AND VOICE SESSIONS

Packetization Interval	Number of stations	Voice Capacity (number of sessions)
10 ms	13	6
20 ms	24	12
30 ms	33	16
40 ms	41	20

The number of sessions is rounded to an even value indicating that when the number of stations is an odd number, the last session is assumed as incomplete. The voice capacity results of Table III match the experimental results presented in [7] indicating the correctness of the analytical methodology presented in this work.

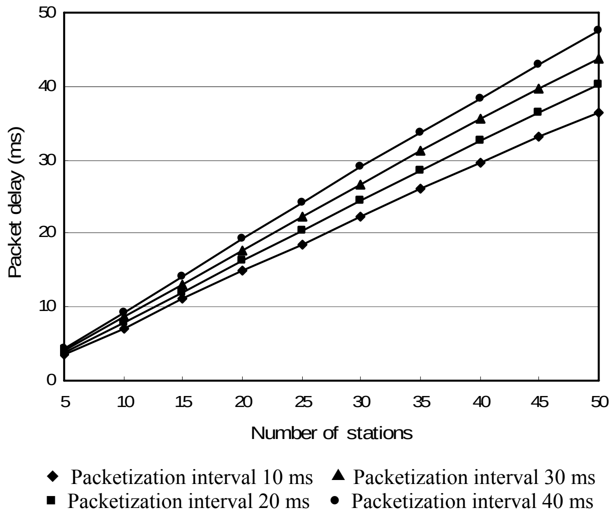

Figure 1 Average packet delay against number of stations at 11 Mbps

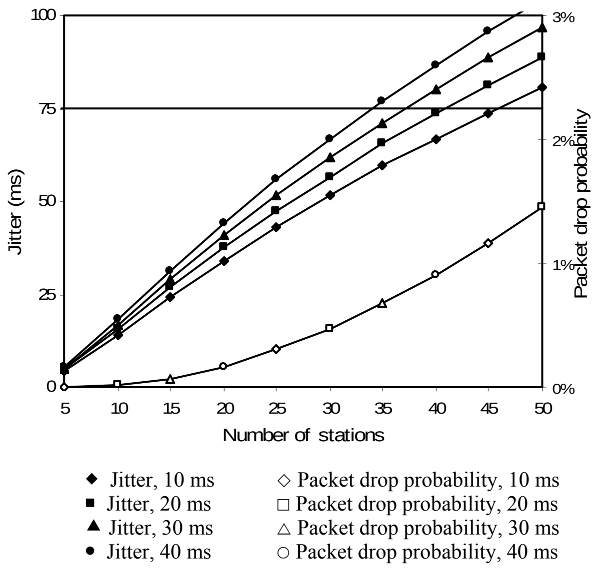

Figure 2. Packet drop probability and jitter versus number of stations at 11 Mbps

In Fig. 2 we plot jitter and drop probability for the same packetization intervals as in previous figure. Figure indicates that packet drop probability is independent of the packetization interval. This can easily be explained by considering equation (7), which indicates that p_{drop} depends only on m and p; p is calculated by solving the non-linear system of (2) and (3), thus p depends on W, m' and m. Figure indicates that the packet drop rate quality limit of 3% is never reached in the considered configuration and thus, packet dropping does not affect the voice capacity of WLANs in this kind of setup. Figure 2 also plots simulative results for jitter as no analytical model for jitter is known to us in the literature for the considered configuration. The jitter quality limit of 75 ms (Table I) is exceeded for high n and the average delay quality limit is reached before the jitter quality limit for the packetization intervals 20 ms and 30 ms. The jitter results for the 40 ms packetization interval indicate that jitter exceeds the quality limit of 75 ms for a WLAN consisting of 34 or more transmitting stations. Thus, jitter limits the voice capacity to 17 sessions, a value lower than the voice capacity considering the average packet delay (fig. 1, Table III).

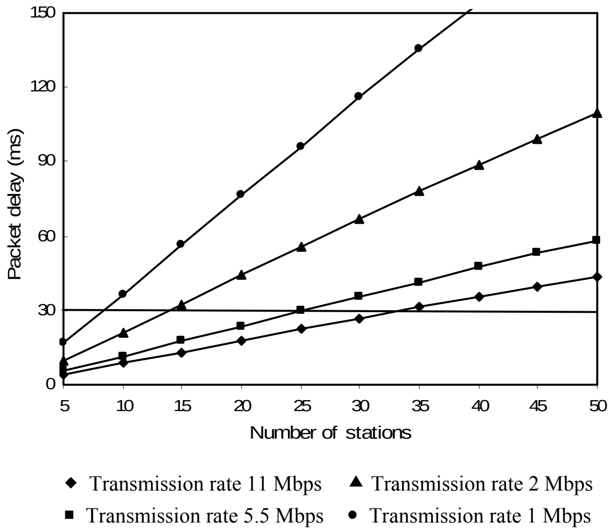

Figure 3 Packet delay for different transmission rates with l=1920 bits

Figure 3 examines the effect of transmission rate on voice capacity by plotting the average packet delay for all transmission rates of the 802.11b for a packet payload of 240 bytes (1920 bits) when G.711 is used for packetization every 30 ms. Our proposed threshold of 30 ms crosses the lines representing the transmission rates at different number of stations indicating the voice capacity value at specific transmission rate. At 1 Mbps there are only 8 stations, at 2 Mbps 13 stations, at 5.5 Mbps 25 stations and at 11 Mbps 33 stations. These results indicate that when transmission rate increases, voice capacity increases. When this packetization interval is used, as in most commercial implementations of IP phones, the voice capacity is reduced from 16 at 11 Mbps to only 4 at 1 Mbps. As it is shown in Figure 2 the packet drop probability and jitter limits are never exceeded for this packetization interval. The dominant factor is the packet delay limit which restricts the voice capacity for the specified packetization interval.

VI. CONCLUSIONS

In this paper, we present a methodology that calculates the voice capacity of ad hoc WLANs based on mathematical models presented in the literature. We present the requirements for good voice quality in terms of delay, jitter and drop probability. We propose that the average packet delay should not exceed the packetization interval indicating the voice capacity for the packetization interval of the codec used. The calculated results match results presented in the literature using simulations. We have also shown that the transmission rate has a significant effect on voice capacity. The number of users increases when the transmission rate increases.

ACKNOWLEDGMENT

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

REFERENCES

[1] IEEE standard for Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications, ISO/IEC 8802-11:1999(E), Aug. 1999.

[2] Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specification: High-Speed Physical Layer Extension in the 2.4 GHz Band, IEEE 802.11b Sept. 1999.

[3] Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specification: High-Speed Physical Layer in the 5 GHz Band, IEEE 802.11a, Sept. 1999.

[4] Koepsel A., Ebert J.-P., and Wolisz A., “A Performance Comparison of Point and Distributed Coordination Function of an IEEE 802.11 WLAN in the Presence of Real-Time Requirements,” in *Proc. of the 7thInternational. Workshop on Mobile Multimedia Communications* (MoMuC2000), Tokyo, October, 2000.

[5] Veeraraghavan M., Cocker N. and Moors T., “Support of Voice Services in IEEE 802.11 Wireless LANs,” in *Proc. 20th Annual Joint Conference of the IEEE Computer and Communications Societies* INFOCOM'2001, vol.1, pp. 488-497, 22-26 April, 2001.

[6] Eiger M., Elaoud M. and Famolari D., “The effect of packetization on voice capacity in IEEE 802.11b networks”, in *Proc. of Consumer Communications and Networking Conf.* (CCNC 2005), pp. 267-271, 3-6 Jan. 2005.

[7] Garg S. and Kappes M., “Can I add a VoIP call?,” in *Proc. of International Conference on Communications (ICC) 2003,* vol. 2, pp. 779-783, 11-15 May 2003.

[8] Medepalli K., Gopalakrishnan P., Famolari D. and Kodama T., “Voice capacity of IEEE 802.11b, 802.11a and 802.11g Wireless LANs”, *in Proc. IEEE Global Telecommunications Conf.* (GLOBECOM ’04), vol. 3, pp. 1549-1553, 29 Nov.-3 Dec. 2004.

[9] Wang W., Liew S. C. and Li V.O.K., “Solutions to performance problems in VoIP over a 802.11 wireless LAN”, *IEEE Trans. on Vehicular Technology*, vol. 54, pp. 366-384, no. 1, January 2005.

[10] Zahedi A. and Pahlavan K., “Capacity of a wireless LAN with Voice and Data services”, *IEEE Trans. On Comm.* vol. 48, no. 7, pp. 1160-1170, July 2000.

[11] Bianchi G.. “Performance Analysis of the IEEE 802.11 Distributed Coordination Function”, *IEEE Journal on Selected Area in Comm.* vol. 18, no. 3, pp. 535-547, 2000.

[12] Wu H., Peng Y., Long K., Cheng S., Ma J., “Performance of Reliable Transport Protocol over IEEE 802.11 Wireless LAN: Analysis And Enhancement”, *in Proc. 21st Annual Joint Conference of the IEEE Computer and Communications Societies* INFOCOM'2002, vol. 2, pp. 599-607, 23-27 June 2002.

[13] Chatzimisios P., Boucouvalas A. C., Vitsas V., “IEEE 802.11 Packet Delay – A finite retry limit analysis” in *Proc. IEEE Global Telecommunications Conf.* (GLOBECOM ’03), San Francisco, CA, USA, 2003, vol. 2, pp. 950–954

[14] R. Caputo, *CISCO Packetized Voice & Data Integration*, McGraw-Hill , 2000.