Calculating Voice Capacity in IEEE 802.11 WLANs Under Basic Access Data Traffic

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***Abstract*— In recent years, the growth of Wireless Local Area Networks (WLANs) is enormous. Supporting voice applications over WLANs is of great interest in scientific and technical community. In this paper an analytical model is developed, based on a Markov chain, for the calculation of saturation throughput and average packet delay for IEEE 802.11 WLANs. A hybrid network scenario is studied with voice stations and data stations competing for the medium. Both data and voice stations use the Basic mechanism of DCF (Distributed Coordination Function). Analytical model is validated by comparing its results with simulation outcome. The analytical model developed in this work is simple, fast, easily implemented and gives accurate results. A methodology for calculating voice capacity for the hybrid scenario is developed.**

Keywords—WLANs; IEEE 802.11; DCF; Markov chain; saturation throughput; voice codec; voice capacity

I. INTRODUCTION

Wireless Local Area Networks are growing rapidly in recent years and now offer considerable speed, reliability and low cost deployment. There is also a great interest for voice applications over wireless networks. VoIP technology provides the tools for developing voice applications and multimedia information exchange over the Internet Protocol (IP). The efficient operation of VoIP wireless technology is a field of great interest.

The main protocol utilized by WLANs is the IEEE 802.11 [1]. This protocol defines Physical layer (PHY) and Medium Access Control (MAC) specifications. According to 802.11 MAC, there are two modes of operation: Point Coordination Function (PCF) and Distributed Coordination Function (DCF). PCF is a polling – based contention free access scheme that uses an Access Point (AP) as point coordinator. DCF is a random access scheme based on Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). In this paper, our work is based on DCF.

DCF uses two techniques for transferring data packets. Basic access mechanism and Request To Send / Clear To Send (RTS/CTS) access mechanism. According to basic access mechanism (which is the default access technique) a station transmits its packet if the medium is detected free and defers if the medium is detected busy. According to RTS/CTS access mechanism two short packets are exchanged between the transmitter (sends a RTS packet) and receiver (answers with a CTS packet) in order to reserve the medium before data packet transmission.

Bianchi [2] proposed an analytical model, based on Markov chain, to compute the saturation throughput performance of the DCF. Saturation throughput is the limit reached by the system throughput as the offered load increases and represents the maximum load that the system can carry in stable conditions [2]. The key assumption of the model of [2] is that the packet collides with a constant probability at each transmission attempt, regardless of the number of retransmissions suffered. The model of [2] is extended in [3] to take into consideration packet dropping when the packet retransmission limit is reached, as specified in the IEEE 802.11 standard. Based on the analytical model of [2] and its extension of [3], an analytical model was developed by [4] for the calculation of average packet delay and jitter (standard deviation of the packet delay). In this work, considering works of [2], [3], and [4] a mathematical model is developed that calculates saturation throughput, average packet delay and voice capacity for a hybrid wireless IEEE 802.11 LAN. Mathematical model is validated comparing its results with simulation results. Simplicity, speed and ease implementation are the advantages of the developed mathematical model. Based on the saturation throughput calculation of a single station, a methodology for calculating voice capacity is developed. Voice capacity was calculated for three popular codecs: G.711, G.729 and G.723.1. Additionally the effect of data transmissions on voice capacity is examined.

This paper is organized as follows: Section II describes the DCF mechanism and the backoff procedure. Section III subsection A describes the examined network scenario and subsection B presents the characteristics of some popular voice encoders. Section IV develops the mathematical model. Specifically subsection A presents the analysis for saturation throughput, subsection B presents the analysis for average packet delay, subsection C develops the mathematical model for a hybrid network system. Subsection D presents the methodology for calculating voice capacity with two different approaches: a) using throughput analysis and b) using average packet delay analysis. In Section V mathematical model was validated by comparing analytical results with simulation results. Results of the developed mathematical model for the calculation of voice capacity are also presented. Finally section VI concludes the paper.

II. DISTRIBUTED COORDINATION FUNCTION (DCF)

Each station with a packet to transmit first senses the medium to ascertain whether it is in use. If the medium is idle for a time interval greater than Distributed Inter Frame Space (DIFS), then the station transmits. If the medium is busy, then the station defers until the medium is sensed idle for DIFS. Then, station generates a random backoff interval. This is the Collision Avoidance (CA) feature of the protocol. The backoff timer is decremented when the medium is idle and frozen when the medium is busy. After a busy period of the medium, the backoff timer resumes only after the medium is idle for more time than DIFS. Finally the station transmits when the backoff timer reaches zero. If the packet is received by the destination station, the destination station waits for a Short Inter Frame Space (SIFS) time interval and responds with an ACK packet. If the ACK is not received by the transmitter within the ACK Timeout interval, or if the transmitter detects another packet transmission on the medium, the packet is considered lost and a retransmission is

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scheduled. Every station keeps a retry counter (or backoff stage), which is increased by one every time a transmission is unsuccessful. When this counter reaches the value of m, which is the retransmission limit then the packet is rejected.

The random backoff interval (i.e. the value of the backoff timer) is chosen in the $[0, W_{i-1}]$, where i is the backoff stage, $i \in [0, m]$ and W_i is the current contention window (CW). The value of (CW) depends on the number of retransmissions a packet suffers. At the first attempt to transmit the packet, CW is set equal to the minimum $CW_{min} = W$. After a retransmission CW is doubled up to a maximum value $CW_{max} = 2^{m'} \cdot CW_{min}$. If contention window reaches value of CW_{max}, then it will remain at this value until it reset after a successful transmission or a rejection of a packet. Consequently:

$$\begin{aligned} W_i &= 2^i \cdot W & 0 \le i \le m' \\ W_i &= 2^{m'} \cdot W & m' < i \le m \end{aligned} \tag{1}$$

where m represents the maximum backoff stage and m' represents the number of different window sizes. According to 802.11b standard, $m = 6$ and $m' = 5$.

III. NETWORK SYSTEM MODEL

A. Network Description – Assumptions

The considered wireless network scenario operates using Distributed Coordination Function (DCF) which is defined by IEEE 802.11 protocol. We are making the following assumptions:

1. There are no transmission errors on the channel and no hidden stations.
2. Network operates in saturation conditions, i.e. all stations always have a packet ready for transmission.
3. Network consists of voice stations and data stations. Voice stations transmit only voice packets while data stations transmit only data packets.
4. Propagation delay (δ) is very small and is neglected. This assumption is valid because all stations are very close to each other.
5. The size of a data packet is constant.
6. All voice stations are within range and transmit at same data rate using same voice codec.
7. Each VoIP call (voice session) consist of two communicating wireless stations.

B. Voice Encoders

TABLE I
Popular Voice Codecs

Voice Codec		**G.711**	**G.729**	**G.723.1**
Bit Rate		64 kbps	8 kbps	5.3/6.3 kbps
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

Voice is analog and must be converted to digital in order to transmit it into the network. Encoders convert the analog voice signal into digital. An encoder performs the following operations: sampling, quantization and encoding. The input of an encoder is an analog signal and the output is data frames. The size of the frame depends on the type of the encoder (codec) and the sampling interval. The international standard for encoding telephone audio is G.711. Encoder G.711 creates voice packets of 160 bytes every 20 ms (sampling interval), leading to a bit rate of 64 Kbps. Encoder G.729 creates voice packets of 20 bytes every 20 ms (bit rate 8 Kbps) while G.723.1 creates voice packets of 24 bytes every 30 ms (bit rate 6.3 Kbps). Table I presents the basic characteristics of codecs used in this work.

IV. MATHEMATICAL MODEL

The mathematical model presented in this paper, is based on: a) Bianchi's [2] work for the performance of the IEEE 802.11 DCF, b) H. Wu's [3] work, which extends Bianchi's model to include packet dropping, c) P. Raptis [4] work for the computation of average packet delay, d) B. D. Amanatiadis [5] analytical model for voice capacity.

The probability τ that a station transmits in a randomly chosen slot time is [3]:

$$\tau = \sum_{i=0}^{m} b_{i,0} = \frac{1 - p^{m+1}}{1 - p} b_{0,0} \tag{2}$$

where $b_{0,0}$ is given by (3)

$$b_{0,0} = \begin{cases} \dfrac{2(1-2p)(1-p)}{W(1-(2p)^{m+1})(1-p) + (1-2p)(1-p^{m+1})} & m \le m' \\ \dfrac{2(1-2p)(1-p)}{W(1-(2p)^{m'+1})(1-p) + (1-2p)(1-p^{m+1}) + W2^{m'}p^{m'+1}(1-2p)(1-p^{m-m'})} & m > m' \end{cases} \tag{3}$$

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

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

Equations (2) and (4) represent a non-linear system with two unknown τ and p, which can be solved using numerical methods and has a unique solution as proved by [2].

A. Saturation Throughput

Let S be the normalized system throughput, defined as the fraction of time the channel is utilized to successfully transmit payload bits. Let P_{tr} be the probability that there is at least one transmission in the considered slot time. Since n stations contend on the channel and each transmits with probability τ :

$$P_{tr} = 1 - (1 - \tau)^n \tag{5}$$

Let now P_s be the probability that a transmission occurring on the channel is successful. This probability is given by the probability that exactly one station transmits on the channel, conditioned on the fact that at least one station transmits:

$$P_s = \frac{n\tau(1-\tau)^{n-1}}{P_{tr}} = \frac{n\tau(1-\tau)^{n-1}}{1-(1-\tau)^n} \tag{6}$$

If $E[P]$ is the average packet payload size, then the average amount of payload information successfully transmitted in a slot time is $P_{tr}P_sE[P]$, because a successful transmission occurs in a slot time with probability $P_{tr}P_s$.

The average length of a slot time $E[Slot]$ is obtained considering that a slot time is empty with probability 1-P_{tr}, contains a successful transmission with probability $P_{tr}P_s$ and contains a collision with probability $P_{tr}(1 - P_s)$:

$$E[Slot] = (1 - P_{tr})\sigma + P_{tr}P_sT_s + P_{tr}(1 - P_s)T_c \tag{7}$$

where σ is the duration of an empty slot time, T_s is the average time the channel is sensed busy due to a successful transmission and T_c is the average time the channel is sensed busy due to a collision. (We will calculate T_s, T_c and $E[P]$ in section C later in this chapter).

Now S can be calculated as follows:

$$S = \frac{P_{tr}P_sE[P]}{(1 - P_{tr})\sigma + P_{tr}P_sT_s + P_{tr}(1 - P_s)T_c} \tag{8}$$

B. Average Packet Delay

The delay D for a successfully transmitted packet is defined as 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 reaches the retry limit, the delay time for the packet will not be calculated in the average delay.

The analytical model developed by Raptis [4] is used to calculate the average packet delay. According to [4] analysis, average packet delay $E[D]$ is given by:

$$E[D] = \sum_{j=0}^{m}\left(T_s + jT_c + E[Slot]\sum_{i=0}^{j}\frac{W_i - 1}{2}\right)\frac{p^j(1-p)}{1 - p^{m+1}} \tag{9}$$

where m is the retry limit.

C. Hybrid System

In this section, T_s , T_c and $E[P]$ will be calculated for a hybrid system with data and voice stations transmitting simultaneously. Particularly N_{data} stations are considered, transmitting only data packets and N_{voice} stations are considered transmitting only voice packets, where:

$$n = N_{data} + N_{voice} \tag{10}$$

In the considered model, voice packets (with size l_{voice}) are transmitted using the Basic mechanism of the DCF and data packets (with size l_{data}) are transmitted using also the Basic mechanism of the DCF.

In this hybrid model, data packets have different size from voice packets. Generally, the size of a data packet is larger than the size of a voice packet. In this work, data packets of 8184 bits are used, in order to have comparable results with the results presented in the references. The size of a voice packet depends on codec used (i.e. for G.711 codec with samples at every 20 ms the size of the voice packet is 1280 bits).

A collision involves two or more packets. The probability of three or more packets to be involved in a collision is very small and can be neglected. It is therefore considered that a collision always involves two packets. To calculate collision duration, it should be considered if a collision involves at least one data packet. In this case and considering that data packets are longer than voice packets, the time duration of the collision is calculated by the duration of the longer data packet l_{data}.

The number of voice stations pairs (CM_{voice}) can be calculated as follows:

$$CM_{voice} = \binom{N_{voice}}{2} = \frac{N_{voice}!}{2!\,(N_{voice} - 2)!} \tag{11}$$

Also the number of all stations pairs (CM) can be calculated as follows:

$$CM = \binom{n}{2} = \frac{n!}{2!\,(n-2)!} \tag{12}$$

From (11) and (12), the probability P_{voice} that a collision involves two voice stations can be calculated by:

$$P_{voice} = \frac{CM_{voice}}{CM} \tag{13}$$

The probability that a collision involves one data station and one voice station, or two data stations, is:

$$P_{data} = 1 - P_{voice} \tag{14}$$

If a packet is successfully transmitted, the probability PS_{voice} that this transmission is a voice transmission can be calculated by:

$$PS_{voice} = N_{voice}/n \tag{15}$$

If a packet is successfully transmitted, the probability PS_{data} that this transmission is a data transmission can be calculated by:

$$PS_{data} = 1 - PS_{voice} \tag{16}$$

Voice packet overhead due to protocols RTP, UDP and IP, is calculated by:

$$RTPUDPIP = RTP + UDP + IP \tag{17}$$

where RTP represents the packet overhead due to the Real-time Transport Protocol, UDP represents the packet overhead due to User Datagram Protocol and IP represents the packet overhead due to Internet Protocol.

Let T_{cdata} be time duration of a collision that involves at least one data station and T_{cvoice} is time duration of a collision between two voice stations. As previously mentioned, if a collision involves a data station and a voice station, then the time duration of the collision is T_{cdata}. T_{cvoice} and T_{cdata} are calculated by:

$$T_{cdata} = DIFS + H + \left(\frac{l_{data}}{C}\right) + SIFS + ACK \tag{18}$$

$$T_{cvoice} = DIFS + H + RTPUDPIP + \left(\frac{l_{voice}}{C}\right) + SIFS + ACK \tag{19}$$

where C is the bit rate and H is packet header (given by table II).

Let T_{sdata} is the time duration that the medium is sensed busy during a successful data packet transmission and T_{svoice} is the time duration that the medium is sensed busy during a successful voice packet transmission. T_{sdata} and T_{svoice} can be calculated as follows:

$$T_{sdata} = DIFS + H + \left(\frac{l_{data}}{C}\right) + SIFS + ACK \tag{20}$$

$$T_{svoice} = DIFS + H + RTPUDPIP + \left(\frac{l_{voice}}{C}\right) + SIFS + ACK \tag{21}$$

Now using equations (11) through (21) T_s and T_c can be calculated:

$$T_s = PS_{data} \cdot T_{sdata} + PS_{voice} \cdot T_{svoice} \quad (22)$$

$$T_c = P_{data} \cdot T_{cdata} + P_{voice} \cdot T_{cvoice} \quad (23)$$

The average length of a successfully transmitted packet (l) is:

$$l = PS_{voice} \cdot l_{voice} + PS_{data} \cdot l_{data} \quad (24)$$

D. Voice Capacity

Voice capacity is the maximum number of voice sessions that can be served simultaneously by the network at specific quality limitations [7], [8].

Using equations (8), (22), (23) saturation throughput S_{voice} of all voice stations can be calculated as follows:

$$S_{voice} = \frac{PS_{voice} \cdot P_{tr} \cdot P_s \cdot l_{voice}}{(1-P_{tr})\sigma + P_{tr}P_sT_s + P_{tr}(1-P_s)T_c} \quad (25)$$

In order to calculate the saturation throughput S_{single} of a single voice station, S_{voice} is divided by the number of voice stations N_{voice}:

$$S_{single} = S_{voice}/N_{voice} \quad (26)$$

This work assumes that voice capacity can be calculated at saturation conditions and that the throughput of a voice station must be equal or higher of the limit S_{limit} :

$$S_{single} \geq S_{limit} \quad (27)$$

If S_{single} exceeds this limit, then the quality of the transmitted voice by the station is acceptable.

S_{limit} can be calculated as follows:

$$S_{limit} = l_{voice}/T_{pck} \quad (28)$$

where T_{pck} is the packetization time i.e. the time required to create a voice packet .

Alternatively, voice capacity can be calculated using average packet delay [4]. Specifically, voice capacity is calculated at the point where the average packet delay $E[D]$ is less or equal of the limit T_{pck} :

$$E[D] \leq T_{pck} \quad (29)$$

V. RESULTS

The values reported in the following figures have been obtained using the system parameters of the DSSS (Direct Spread Sequence Spectrum) physical layer, used in IEEE 802.11b standard. Parameter values are shown on Table II.

To validate the proposed mathematical model, analytical results are compared with simulation results. In all subsequent diagrams, analytical model values are presented by lines and simulation values by marks. In all diagrams analytical model results coincide with simulation results. This fact proves that the proposed analytical model is reliable. In any case the difference between analytical model results and simulation results is lower than 0.5%.

Figure 1 plots the throughput of a single voice station against the number of voice sessions, with or without data stations present, for voice codec G.711. The voice capacity is calculated at the point where the saturation throughput for the specific packetization interval crosses the horizontal line, of value S_{limit}. The value of S_{limit} is calculated from (28) and equals to 0.00554. Calculated voice capacity is presented in Table III.

Table IV shows voice capacity of proposed algorithm (PA: proposed algorithm) for the most popular codecs, assuming that there are no data stations present. Results for voice capacity of [9] are also presented on table IV for comparison. Table IV shows that the results of PA are similar as those of [9], even though PA is simpler and easily implemented. This fact indicates effectiveness of analytical model used.

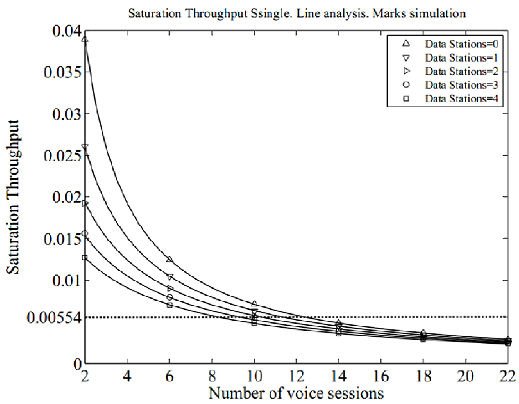

Fig1. Voice Capacity using G.711 codec, packetization interval 20 ms

TABLE II

Parameters DSSS, 802.11b, G.711, G.729, G.723.1

Parameter	Value
Data packet	8184 bits
Slot	20 μs
SIFS	10 μs
DIFS=2•slot+SIFS	50 μs
PHY (physical) layer header	192 bits at 1Mbps
ACK	112 bits at 11 Mbps + PHY at 1 Mbps
MAC layer header	224 bits
H (packet header)	224 bits MAC data at 11 Mbps + PHY at 1 Mbps
RTP	12 • 8 = 96 bits
UDP	8 • 8 = 64 bits
IP	20 • 8 = 160 bits
CW_{min}	32
CW_{max}	1024
Number of CW sizes (m′)	5
Retry Limit	6
Voice codec	G.711
Interval	20 ms
Packet payload	1280 bits
Voice codec	G.729
Interval	20 ms
Packet payload	160 bits
Voice codec	G.723.1
Interval	30 ms
Packet payload	192 bits

TABLE III

Voice Capacity using G.711 codec, packetization interval 20 ms

Data stations	Voice sessions
0	12
1	11
2	10
3	9
4	8

TABLE IV
Comparison of voice capacities for various codecs

	G. 711		G.729		G.723.1	
Packetization Interval (ms)	**PA**	**[9]**	**PA**	**[9]**	**PA**	**[9]**
10	7	6	7	7		
20	12	11	14	13		
30	16	15	20	19	20	19
40	19	18	25	23		
50	22	20	30	28		
60	24	22	35	32	35	33

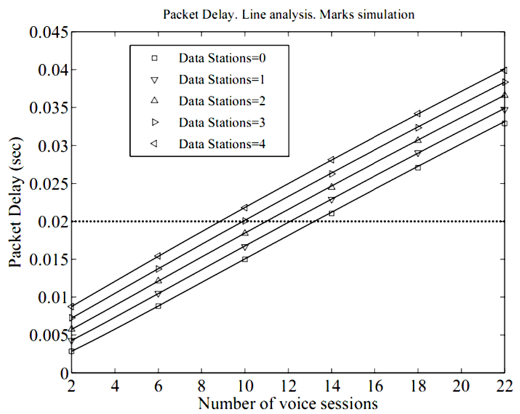

Fig2. Voice Capacity – Packet Delay, packetization interval 20 ms

TABLE V
Voice Capacity (from Packet Delay) using G.711 codec, packetization interval 20 ms

Data stations	Voice sessions
0	13
1	12
2	10
3	9
4	8

Figure 2 plots the average packet delay against the number of voice sessions, with or without data stations present, for voice codec G.711 (packetization interval 20 ms). Voice capacity is calculated at the point where the line of the average packet delay crosses the horizontal line of value T_{pck}. The value of T_{pck} equals to 20 ms (for G.711 codec).Table V shows voice capacity results. Results are similar with those calculated using saturation throughput of a single station S_{single}. Consequently, since similar results obtained from two different calculation methods, then our proposed model is reliable.

Figure 3 plots the throughput of a single voice station against the number of voice sessions, with or without data stations present, for voice codec G.729. Voice capacity is calculated at the point where the saturation throughput for the specific packetization interval crosses the horizontal of value S_{limit}, which is equal to 0.00069358. Voice capacity is presented in Table VI.

TABLE VI
Voice Capacity using G.729 codec, packetization interval 20 ms

Data stations	Voice sessions
0	14
1	13
2	12
3	10
4	9

Figure 4 plots the throughput of a single voice station against the number of voice sessions, with or without data stations present, for voice codec G.723.1. Voice capacity is calculated at the point where the saturation throughput for the specific packetization interval crosses the horizontal of value S_{limit}, which is equal to 0.0005548. Voice capacity is presented in Table VII.

Figure 5 plots the throughput of a single voice station against the number of voice sessions, with or without data stations present, for voice codec G.723.1 (packetization interval is 60 ms). Voice capacity is calculated at the point where the saturation throughput for the specific packetization interval crosses the horizontal of value S_{limit}, which is equal to 0.0005548. Voice capacity is presented in Table VIII.

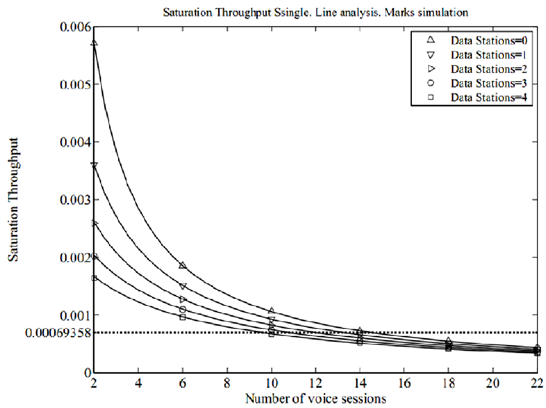

Fig3. Voice Capacity using G.729 codec, packetization interval 20 ms

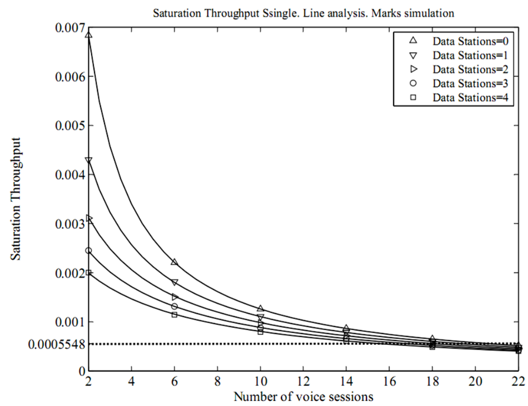

Fig4. Voice Capacity using G.723.1 codec, packetization interval 30 ms

TABLE VII
Voice Capacity using G.723.1 codec, packetization interval 30 ms

Data stations	Voice sessions
0	20
1	19
2	18
3	16
4	15

Considering calculated voice capacity for various codecs in tables III, VI, VII, VIII, we can conclude that if a data station enters the network then a voice session is lost. For example, table III shows voice capacity using G.711 codec, for packetization interval 20ms. If there are no data stations in the network, then voice capacity has calculated to 12 voice sessions. If one data station is present, then voice capacity is reduced to 11 voice sessions, if two data stations are present, then voice capacity is reduced to 10 voice sessions, etc.

Vourkas [6] made similar calculations, assuming that data packets were transmitted using RTS/CTS mechanism of the

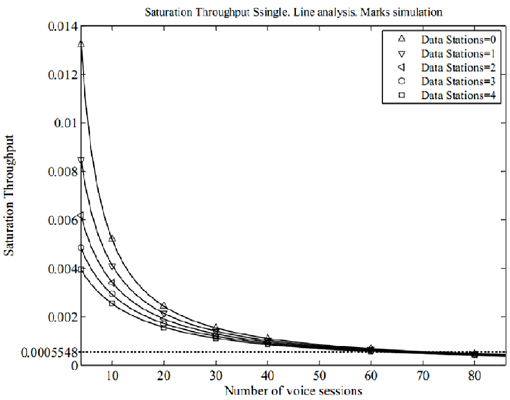

Fig5. Voice Capacity using G.723.1 codec, packetization interval 60 ms

TABLE VIII
Voice Capacity using G.723.1 codec, packetization interval 60 ms

Data stations	Voice sessions
0	35
1	34
2	33
3	32
4	31

TABLE IX
Voice Capacity calculated by [6], using G.711 codec and packetization interval 20 ms

Data stations	Voice sessions
0	12
1	10
2	9
3	8
4	7

DCF. Voice capacity calculated by [6] using G.711 codec and packetization interval 20ms, is presented in Table IX.
Comparing results presented in tables III and IX, it is observed that if data stations utilize basic access method, the voice capacity of the network is increased for a small number of data stations. It seems that Basic mechanism of the DCF (for data packets), uses better the medium and gives slightly better throughput and slightly lower average packet delay. This is explained as follows:
When a few voice stations are present in the network (left part of curves in Figures), collisions are limited. Thus the advantage of RTS/CTS access method is not important and Basic access method achieves higher results. RTS/CTS access method efficiency is increased when the number of collisions is high. This happens because collision duration is decreased, because only small RTS frames are involved in the collision instead of the data frames.
RTS/CTS access method is not efficient when the number of collisions is small and the number of successful transmitted packets is large. In this case RTS/CTS overhead reduces efficiency.
As the number of voice stations is increasing, the number of collisions is increasing as well as more stations compete for the medium. As data stations are significantly fewer than voice stations, the probability that a collision involves two voice stations or one voice station and one data station, is greater than the probability a collision to involve two data stations.
In the RTS/CTS scenario [6], when a collision occurs and the collision involves a data station, then the small RTS packet collides. If a collision involves a voice packet and a RTS or CTS packet, then the duration of the collision is calculated considering the greater voice packet. Thus the advantage of RTS/CTS access method is not taken, because the probability a collision to involve two data stations is very small (if many voice stations are present).

VI. CONCLUSIONS

In this paper a mathematical model was developed that calculates saturation throughput and average packet delay for a hybrid wireless IEEE 802.11 LAN, with voice and data stations transmitting simultaneously. Our network scenario assumes that data packets and voice packets are transmitted using the basic access method of the DCF. The mathematical model was validated by comparing analytical with simulation results. Based on saturation throughput, a methodology for calculating voice capacity was developed. Voice capacity was calculated for three popular codecs: G711, G.729 and G.723.1.Results indicate that voice capacity is reduced by one or two sessions for each data station entering the network. Basic access method gives better results on voice capacity. Comparing results of our work with results of [6], voice capacity of our scenario is one or two sessions greater. That can be explained because RTS/CTS overhead reduce efficiency.

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