QUALITY EVALUATION OF VOIP SERVICE OVER IEEE 802.11 WIRELESS LAN

Andrea Barbaresi, Massimo Colonna, Andrea Mantovani and Giovanna Zarba

Telecom Italia, via G. Reiss Romoli 274, I-10148 Torino (TO), Italy

Abstract: Wireless broadband access, in particular that based on IEEE 802.11 WLAN technology, is reaching significant penetration rates in most countries and can be considered a valid and cheap alternative to xDSL or optical fiber technology, particularly in low density areas and small villages, to offer a bundle of services including VoIP. In this paper we evaluate the performances of the VoIP service in a WLAN network by means of an event-driven simulator developed by Telecom Italia.

Keywords: VoIP, WLAN, IEEE 802.11, MOS, quality evaluation.

1. INTRODUCTION

The growing interest in Voice over IP (VoIP) service is essentially due to the fact that nowadays Internet-Protocol (IP)-based networks are widely deployed and the broadband access is offered at very low cost. Traffic voice, generally offered through the circuit-switched telephone system (PSTN networks), is moving over the packet switched Internet with the support of new protocols, SIP and H.323 defined by ITU and IETF, respectively.

Broadband access, in particular that based on IEEE 802.11 Wireless LAN (WLAN) technology, is reaching significant penetration rates in most developed countries due to heightened market competition: at the beginning WLANs have been essentially used as cable replacement for residential or office environments; in recent years, public hotspots have been deployed in airports, hotels, trading centres essentially because people, by means of very low cost wireless cards available for PCs and PDAs, begin to expect and demand anytime and anyway Internet access. Moreover, WLAN networks have been installed by some municipalities in order to offer broadband services, including VoIP (named VoWLAN), at low cost to all citizens; this kind of networks is a cheaper alternative to xDSL or optical fiber technology particularly in low density areas and small villages.

In the last years, many works detailing the throughput performance of 802.11b WLANs have been presented. Most of them, focus either on analytical modeling or on simulations for data transfer (Bianchi, 2000; Tay and Chua, 2001). Experimental analysis of the VoIP service employing a G.711 codec are shown in (Anjoum, *et al*., 2003) and in (Garg and Kappes, 2003), while simulation analysis are shown, for example, in (Hole and Tobagi, 2004) and in (Coupechoux, *et al*. 2004).

This paper analyses the performances of the VoIP service in a WLAN network with low bit rate codecs (mainly the 1.2. kbit/s AMR codec); this analysis is carried out by means of an event-driven simulator developed by Telecom Italia that simulates all the elements involved in a VoIP system including the signalling protocols, the conversation dynamics and the ITU-T quality evaluation metrics (MOS and E-model). The completeness of our simulator makes our work an enhancement of previous simulative works.

The paper is organized as follows: Section 2 presents some general information about the protocol stack in the VoIP system with an emphasis on the WLAN access technique. Section 3 shows the bi-directional traffic model for VoIP sources used in the simulations, while Section 4 presents the QoS evaluation. In Section 5 a general description of the simulated scenario is provided, as well as information about simulation hypothesis. Section 6 shows the main simulation results obtained and, finally, Section 7 summarizes the main results coming from this study.

2. VOIP OVER WLAN

2.1 VoIP protocol stack

Taken into account that packet-switched technology can deliver services more cost efficiently than today's circuit switched technology, an efficient voice encoding and decoding mechanism is vital. The purpose of a voice coder (vocoder) - also referred to as coder/decoder, or simply "codec" - is to use the analog signal coming from human speech and transform and compress it into digital data. Various voice compression schemes have been developed: we will focus on two most ITU-T low bit rate codecs, namely G.723.1 and G.729, and on the 3GPP codec for the UMTS system, namely AMR. G.723.1 is a codec that has two bit rates associated with it, 5.3 and 6.3 kbit/s, whose mode of operation can change dynamically at each frame. It encodes speech in frames of 30 ms using the MP-MLQ technique for the high rate codec and the ACELP technique for the low rate one. The G.729 codec uses the CS-ACELP coding technique and operates at 8 kbit/s with an input frame of 10 ms. The AMR codec has been chosen by 3GPP for the compression of voice signals in 3G mobile communications and it is an evolution of the GSM-EFR codec. It consists in a single integrated speech codec with eight source rates, i.e: 12.2 (GSM-EFR), 10.2, 7.95, 7.40 (IS-641), 6.70 (PDC-EFR), 5.90, 5.15 and 4.75 kbit/s and it works with speech frames of 20 ms, using MR-ACELP as a coding scheme.

SIP is the main signalling protocol today for the VoIP service. It has been developed by the IETF to provide a simple, scalable and easy-to-implement protocol from an IP perspective. SIP defines packet exchange procedures for setting up, modifying and tearing down multimedia sessions. Although SIP works with most transport protocols, its optimal transport protocol is RTP. This protocol, defined in RFC 1889, operates on the layer above UDP/IP and provides delivery monitoring of its payload types through sequencing and timestamping. RTP is optionally augmented by a control protocol, RTCP, also defined in RFC 1889, that enables exchanges of control information between session participants with the goal of providing quality-related feedback. IP packets with voice frames or signalling messages are then sent to the lower layers of the protocol stack (LLC, MAC and PHY) for the transmission on the wireless medium. In our study these layers are carried out according to the IEEE 802.11 standard.

2.2 IEEE 802.11

In the IEEE 802.11 MAC Layer, the fundamental mechanism to access the medium is called Distributed Coordination Function (DCF) and it is based on the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) protocol. DCF describes two techniques that can be employed for packet transmission: the default scheme is a two-way handshaking technique called Basic Access mechanism.

According to the Basic Access mechanism (Fig. 1) a station with a packet to transmit, monitors the channel activity until an idle period equal to a DIFS (Distributed InterFrame Space) has been observed. In case the medium is sensed busy, a random backoff interval is selected. The backoff time counter is

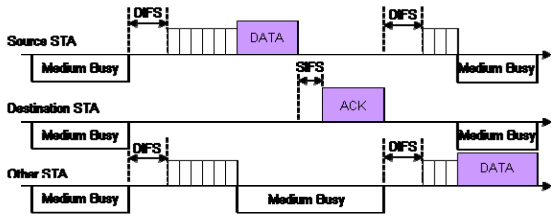

Fig. 1. Basic Access mechanism.

decremented as long as the channel is sensed idle, stopped when a transmission is detected on the channel, and reactivated when the channel is sensed idle again for more than a DIFS period. The station transmits when the backoff time reaches 0. If two or more stations start transmission simultaneously, a collision occurs. Unlike wired networks (e.g. with CSMA/CD), in a wireless environment collision detection is not possible. Hence, a positive acknowledgement (ACK) is used to notify to the station that the transmitted frame has been successfully received. The ACK transmission is initiated at a time interval equal to the SIFS (Short InterFrame Space) after the end of the reception of the previous frame. If this ACK is not received, the station retransmits the packet. The DCF adopts a binary exponential backoff technique. The backoff time is uniformly chosen in the interval [0; CW] defined as the Backoff Window (Contention Window). At the first transmission attempt, the CWmin value is considered, and it is doubled at each retransmission up to CWmax.

Concerning the Physical Layer, the most widely used standard, IEEE 802.11b, specifies a particular form of spread spectrum technology, named CCK (Complementary Code Keying), which allows a maximum data rate of 11 Mbit/s. Other data rates, used when the channel deteriorates, are 5.5, 2 and 1 Mbit/s. An alternative solution is the IEEE 802.11a standard that allows optionally a maximum data rate of 54 Mbit/s while data rates of 6, 12, and 24 Mbit/s are mandatory. The 802.11a standard uses OFDM.

3. VOIP TRAFFIC MODEL

In order to carry out a realistic analysis of VoIP, the conversation between two users has been modeled using the conversational speech model specified in the ITU Rec. P.59. According to this model, the conversation between two users A and B can be modeled as a four state Markov chain (Fig. 2):

- State A: A talking B silent;
- State B: A silent B talking;
- State D: Double talking;
- State M: Mutual silence.

This model is characterized by the transition probabilities between the states p1, p2, p3, considering negligible the transitions between the states A-B and D-M, due to their rare occurrence. Furthermore, the sojourn time in A, B, D, and M states is modeled by a random variable with

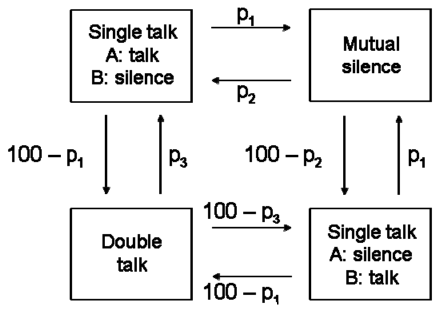

Fig. 2. State transition model for a conversation.

exponential distribution, respectively defined by parameters λ_A, λ_B, λ_D and λ_M. The typical values for all these parameters are listed in ITU-T Rec. P.59.

4. QUALITY EVALUATION: THE E-MODEL

The telephone industry employs a subjective rating system known as the MOS (Mean Opinion Score) to measure the quality of telephone connections. MOS is defined in ITU-T Rec. P.800 and is based on the opinions of many volunteers who listen to a sample of voice traffic and rate the quality of the transmission. They rate the voice samples from 1 to 5 with 5 being "excellent" and 1 being "bad": the voice samples are then awarded a MOS.

An alternative method for the voice quality evaluation is represented by the E-model defined in ITU-T Rec. G.107 as well as other associated ITU-T Recommendations: it is an analytic model of voice quality evaluation used for network planning purposes that reflects the effects of different types of impairments on the end-to-end speech transmission performance. E-model bases its own behaviour on the following consideration: "psychological factors on the psychological scale are additive", which means that each impairment factor which affects a voice call can be computed separately, even if this does not imply that such factors are uncorrelated, but only that their contribution to the estimated impairments are separable.

A basic result of the E-Model is the calculation of the R-factor, which is a simple measure of voice quality ranging from a best case of 100 to a worst case of 0. User satisfaction and the corresponding R and MOS ranges are shown in Fig. 3. The R-factor uniquely determines the MOS through the following relation:

$$\begin{aligned} &MOS=1, && R<0 \\ &MOS=1+0.035\cdot R+7\cdot 10^{-6}\cdot R\cdot(R-60)\,(100-R), && 0<R<100 \\ &MOS=4.5, && R>100 \end{aligned} \qquad (1)$$

The operational range for PSTN voice quality corresponds to MOS >= 3.6. The desirable range of operation for toll quality is MOS >= 4.

The R-factor is composed by several additive terms each one representing a specific source of voice

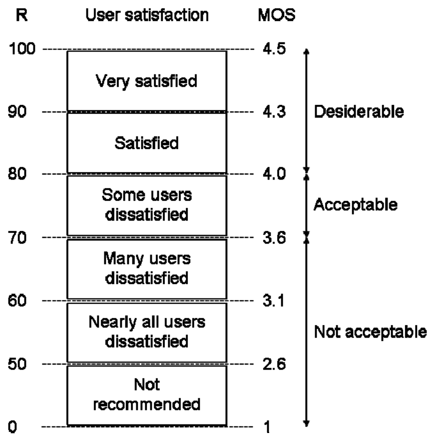

Fig. 3. Voice quality levels.

quality degradation:

$$R= R_0- I_s - I_d -(I_e\text{-eff}) + A \qquad (2)$$

In this formula,

- R_0 represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise;
- I_s, the simultaneous impairment factor, is the sum of all impairments which may occur more or less simultaneously with the voice signal;
- I_d, the delay impairment factor, represents all impairments due to delay of voice signals;
- I_e-eff, represents all the impairments caused by low bit rate codecs and by packet-losses;
- the advantage factor A allows for compensation of impairment factors when there are other advantages of access to the user.

R_0 and I_s are not a function of the underlying packet network and consequently they can be assigned their default values listed in ITU-T Rec. G.107; I_d can be evaluated with the following expression (Atzori, *et al.*, 1994):

$$I_d =0.024\cdot d+0.11\cdot(d-177.3)\cdot u(d-177.3) \qquad (3)$$

where *d* is the end-to-end delay (measured in ms) and u(…) is the Heaviside step function; I_e-eff is finally calculated using the expression (G.107, 2005):

$$I_e\text{-eff} = I_e +(95- I_e)\cdot p/(p+Bpl) \qquad (4)$$

where p is the packet loss rate (expressed in % of received frames) and I_e and Bpl are codec-dependent parameters whose provisional values for the codecs previously described are listed in ITU-T Rec. G.113.

5. SIMULATION METHODOLOGY

Several dynamic simulations have been carried out in order to evaluate the QoS for the VoIP service. The network architecture used in all the simulations is depicted in Fig. 4. It is an 'infrastructure' network formed by a single Access Point (AP) and a

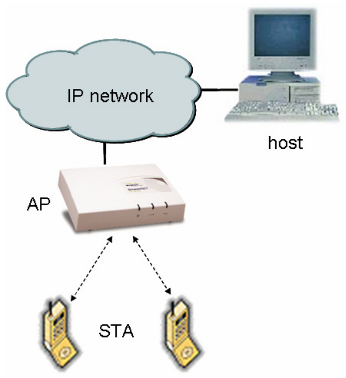

Fig. 4. Simulated architecture

multiplicity of Stations (STA) connected to it. Each VoIP call is established between a STA and the host inside the IP fixed network. The IP network does not introduce delays and losses on the VoIP packets. Hence, the performance evaluation is limited to the WLAN component of the network. Moreover, since we are interested on the specific behaviour of the MAC layer, we also assume that the wireless channel is ideal: path loss and fading phenomena are both absent and the only source of errors on received packets is the collisions caused by the contemporary transmission of two or more stations (including the AP).

In order to perform the simulations, some specific modules for VoIP service have been used:

- VoIP traffic generator: it produces the VoIP traffic for the peer entities involved in the voice call;
- VoIP client: it uses RTP protocol to send VoIP frames over the network and it includes a buffer (dejittering buffer) to compensate jitter effects when receiving VoIP frames from the peer entity;
- VoIP sessions control module: it uses the SIP protocol to establish and release VoIP calls.

The simulation work has been focused on the evaluation of the QoS experienced by VoIP users with respect to various parameters; the final objective is to find out the maximum capacity that can be achieved by the MAC layer, in terms of the maximum number of active STAs (i.e. STAs that have a call in progress at the same time), with an acceptable quality (MOS $\geq$ 3.6).

Simulations have been conducted with the following additional hypothesis:

- Mean VoIP call duration = 120 s;
- Mean VoIP traffic per user = 250 mErlang;
- 802.11 Multple Access technique: Basic Access.

6. SIMULATION RESULTS

This section shows the main results of the simulations, with a focus on the following application-layer parameters:

- MOS: average value of the Mean Opinion Score;
- Transmission delay: end-to-end delay of voice frames between the STA and the host;
- Packet loss: percentage of packets that are discarded at the receiving side if they arrive to early and the dejittering buffer is full or to late after their nominal playout time;
- Number of playout blocks: number of times per session in which the dejittering buffer becomes empty.

The previous parameters are obtained averaging the different values calculated in each call.

6.1 Performance with an AMR codec

This first simulation aims at characterizing the performances of the WLAN network for the four 802.11b transmission rates. This simulation has been carried out considering the AMR codec with the fixed rate of 12.2 kbit/s, a frame length of 20 ms and a dejittering buffer of 60 ms at the receiver (that means that the buffer can contain up to three AMR frames).

Fig. 5 shows the downlink average MOS values for the four transmission rates of the standard and the uplink values for the 11 Mbit/s rate, while Table 1 shows the average transmission delay and the average discarded packets for the 11 Mbit/s transmission rate in both downlink and uplink.

To understand the behaviour of the network, let us focus on the MOS values for the 11 Mbit/s transmission rate (Fig. 5). When the number of active STAs is low, less than 6, all the offered voice traffic is immediately transmitted by the equipment without waiting in the MAC layer buffers; this is confirmed by the value of the transmission delay that is about 61 ms (Table 1) and it is constituted by the sum of the dejittering buffer delay (60 ms), the MAC protocol delay (about 0.84 ms) and the transmission delay on the air interface (about 0.06 ms). The collisions and the subsequent retransmissions, that are however always present, give a minimal contribute. At the same time, the percentage of discarded packets is null. In this case MOS is mainly determined by I_e (that is equal to 5 for the AMR codec) and for this reason it remains constant to the maximum value of 4.47. When the offered traffic increases, i.e. when the number of active STAs reaches the 7 units, the network approaches its throughput limit and contemporarily the number of collisions becomes more relevant: packets have to wait more in the MAC buffers causing an increase of the transmission delay above all and minimally of the discarded packets; the final result is a MOS decrease. When the number of active STAs equals or exceeds the 8 units, the offered traffic is higher then the maximum throughput and the voice packets have to wait much longer in the MAC buffer before their transmission: this is evidenced by the rapid increase of the delay curve. At the same time, also the discarded packets increase but they are still quite low (less than 1%) and they have a

Table 1 Average transmission delay and discarded packets for the 11 Mbit/s transmission rate

	Downlink		Uplink	
Number of active STAs	Delay [ms]	Discard. packets [%]	Delay [ms]	Discard. packets [%]
1	61.0	0	61.0	0
6	61.6	0	61.3	0
7	73.5	0.03	61.5	0
8	190	0.35	61.8	0
9	360	0.75	61.9	0

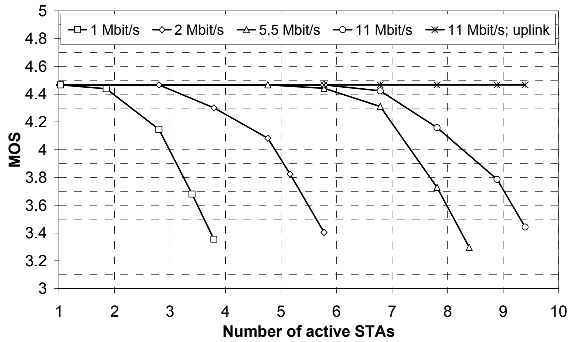

Fig. 5. Downlink average MOS for the IEEE 802.11b transmission rates and uplink MOS for the 11 Mbit/s rate.

limited effect on the MOS degradation. The maximum number of active STAs in order to have an acceptable voice quality (MOS ≥ 3.6) is 9.

The uplink MOS, on the contrary, remains constant to the value of 4.47 due to the low delay (with 9 active STAs it is equal to 62 ms) and to the null value of the discarded packets. This difference with the downlink is due to the IEEE 802.11b multiple access technique that has been designed to be fair between all the equipments. In our scenario, instead, the AP has to transmit the half of the voice traffic and so it follows to be damaged.

When the transmission rate reduces, the network performance get worse and the maximum number of active STAs supported by the network decreases to 7 (5.5 Mbit/s), 5 (2 Mbit/s) and 3 (1Mbit/s) units. This performance reduction is not directly proportional to the rate reduction due to the increasing efficiency of the MAC protocol due to the fixed duration of some MAC overhead elements (DIFS, SIFS, PHY layer preamble, ACK).

Another effect able to affect the overall QoS level of VoIP service is the increase of the time needed to setup and to release the call because of the increasing downlink transmission delay. The mean value of the session setup delay, for the 11 Mbit/s transmission rate, increases from 0.3 s (with 1 active STA) to 1.3 s (with 9 active STAs), while similarly the mean value of the session release delay increases from 0.06 s (with 1 active STA) to 1.0 s (with 9 active STAs). Fig. 6 shows the call setup and call release CDFs for

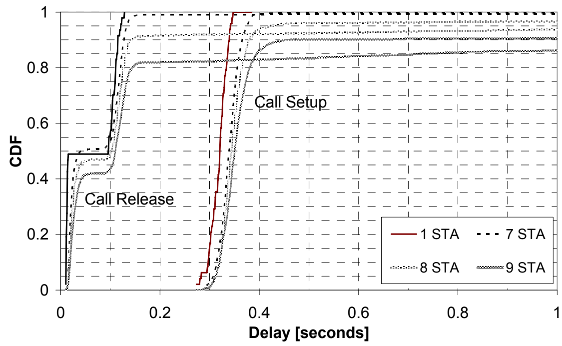

Fig. 6. CDF for the call setup and call release delays with the 11 Mbit/s transmission rate.

the 11 Mbit/s transmission rate. When the number of active STAs is high, i.e. 8 or more, the call release procedure suffer higher delays more frequently than the call setup procedure even if the latter involves a greater number of SIP messages. Nevertheless, even with these high delays, all the calls are correctly setup and released and there are no calls that are dropped during their progress due to the excessive delay of voice packets.

6.2 Effect of buffer length and VAD

In the following, we analyse the impact on performance due to dejittering buffer length variations and VAD (Voice Activity Detection).

Fig. 7 shows the downlink MOS values versus the maximum number of active STAs for different lengths of the dejittering buffer (20 ms, 40 ms, 60 ms and 120 ms). Each variation of this length causes an identical variation of the transmission delay: the MOS changes accordingly but the maximum number of active STAs supported by the network remains unchanged (at least for the buffer lengths used in our simulations). The optimum length for this buffer can be determined looking at the number of playout blocks per session: they have to be minimized because even if they do not influence the voice quality expressed through the E-model, they impact on the overall quality perceived by the user because they produce "silence spikes" during the conversation. Fig. 8 shows the number of playout blocks versus the buffer length in three different cases; a low number of playout blocks can only be obtained with a buffer length at least equal to 40 ms.

The quality/bandwidth ratio can be enhanced using the technique called Voice Activity Detection (VAD). With VAD, during inactivity periods, the coding scheme does not process speech fragments, but it generates a Silence Descriptor (SID) that contains a set of characteristics that describe background ambient noise: this SID is sent to the receiver, which decodes it and plays a "comfort noise". Moreover, SID is coded with a lower bit rate than speech frames. The generation of SID frames is not continuous and the algorithm taken into account

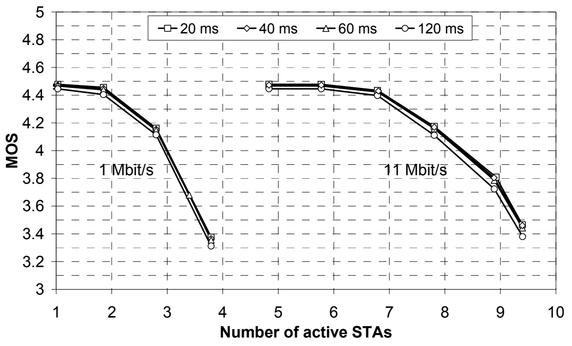

Fig. 7. Downlink average MOS for different dejittering buffer lengths with the transmission rates of 1 Mbit/s and 11 Mbit/s.

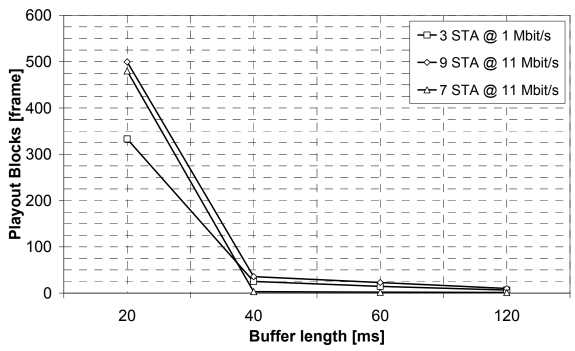

Fig. 8. Downlink average playout blocks versus the dejittering buffer length.

is the following: when inactivity is detected, the codec generates 7 frames similar to the speech ones, 1 SID frame, 2 NOTX frames (NOTX frames are frames in which nothing is transmitted), 1 SID frame and then a periodic sequence of 7 NOTX frames and 1 SID frame up to the end of the silence period. Table 2 compares the maximum number of active STAs with and without VAD for the four 802.11b transmission rates. Improvements are minimal because there are however SID frames that consume bandwidth and they are limited to the 2 Mbit/s and the 5.5 Mbit/s transmission rates.

6.3 IEEE 802.11a PHY layer

Fig. 9 shows the downlink average MOS for several IEEE 802.11a transmission rates in the same conditions of the previous analysis. The maximum number of active STAs ranges from 24 (6 Mbit/s) to 53 (54 Mbit/s). Comparing these curves with those related to the 802.11b PHY layer (Fig. 5), we notice that the 6 Mbit/s transmission rate has better performance than the 11 Mbit/s one. This improvement is due to the higher efficiency of the 802.11a PHY layer. Moreover, also with this PHY layer, the usage of VAD has a marginal effect (Table 2).

6.4 Comparison with the G.723 and G.729 codecs

In this section, we will compare the AMR codec with two other ITU-T low bit rate codecs; in particular we will analyze the G.723.1 one with the fixed rate of 5.6 kbit/s and the G.729 one with the rate of 8 kbit/s. Also for these codecs the frame length and the dejittering buffer length have been supposed respectively of 20 ms and 60 ms. With a low offered traffic, the downlink MOS, shown in Fig. 10, is exclusively determined by the intrinsic quality of the coded, i.e. by its I_e value. Instead, when the offered traffic increases, the transmission delay prevails on I_e and this reduces the MOS differences.

<u>Table 2 Maximum number of active STAs with and without VAD</u>

Transmission rate	No VAD	VAD
1 Mbit/s	3	3
2 Mbit/s	5	6
5.5 Mbit/s	7	8
11 Mbit/s	9	9
6 Mbit/s (802.11a)	24	27
54 Mbit/s (802.11a)	53	55

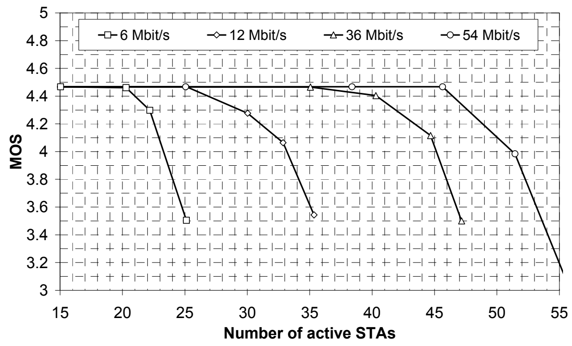

Fig. 9. Downlink average MOS for several IEEE 802.11a transmission rates.

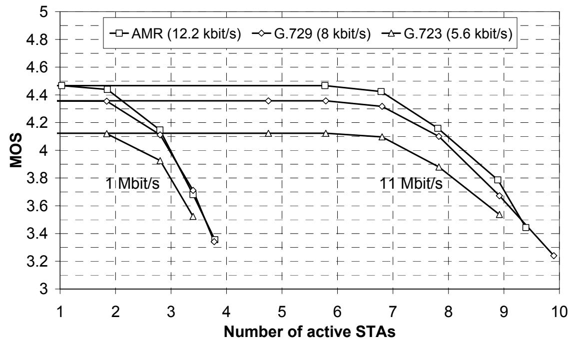

Fig. 10. Downlink average MOS for three low bit rate codecs at 1 Mbit/s and 11 Mbit/s.

7. CONCLUSIONS

In this paper we have analysed the performance of the VoIP service when it is provided to WLAN users. By means of an event-driven simulator that simulates all the elements involved in a VoIP system, we have derived for the AMR codec the maximum number of active STAs supported by an IEEE 802.11b and 802.11a network: it ranges from 3 to 9 in an IEEE

802.11b network and from 24 to 53 in a 802.11a one. We have also derived the minimum value for the dejittering buffer length (it is 40 ms) and we have evaluated the effect of the VAD option. Finally, the AMR performances have been compared with those of two popular ITU-T low bit rate codecs (G.723.1 and G.729).

ACKNOWLEDGEMENTS

This work has been partially performed within the framework of the IST-AROMA project (Casadevall, *et al.*, 2006) (www.aroma-ist.upc.edu), which is partially founded by the European Community.

REFERENCES

G. Bianchi, “Performace analysis of the IEEE 802.11 Distributed Coordination Function”, IEEE *Journal on Selected Areas in Communications,* **Vol. 18**, **No. 3**, March 2000.

Y. C. Tay, K. C. Chua, “A capacity analysis for the IEEE 802.11 MAC protocol”, *Wireless Networks*, **Vol. 7**, **Issue 2**, March 2001.

F. Anjum *et al.*, Voice Performance in WLAN Networks - An Experimental Study”, *Proceedings of IEEE GLOBECOM’03*, **Vol. 6**, December 2003.

S. Garg, M. Kappes, “An experimental study of throughput for UDP and VoIP traffic in IEEE 802.11b networks”, *Proceedings of IEEE WCNC’03*, March 2003.

D. P. Hole, F. A. Tobagi, “Capacity of an IEEE 802.11 wireless LAN supporting VoIP”, *Proceedings of IEEE ICC 2004*, 2004.

M Coupechoux *et al.*, “Voice over IEEE 802.11b capacity”, Proc. of the 16th ITC Specialist Seminar on Performance Evaluation of Wireless and Mobile Networks, September 2004.

L. Atzori, M. L. Lobina, “Speech playout buffering based on a simplified version of the ITU-T E-model”, *IEEE Signal Processing Letters*, **Vol. 11**, **No. 3**, March 2004.

P. F. Casadevall *et al.*, “Overview of the AROMA Project”, IST Mobile & Wireless Communications Summit, 2006.

BIOGRAPHY

Andrea Mantovani received the Laurea degree in Electronic Engineering in 2003 from Politecnico di Torino, Italy. After a first job experience as electronic designer, in 2005 he started his activities in Telecom Italia, Turin, Italy. His main research interests concern wireless communications and, in particular, on the analysis of real time packet switched services over 3G mobile networks and WLANs.

Andrea Barbaresi received a Laura degree cum laude in Electronic Engineering from the University of Ancona, Italy, in 1999. He has been working in the R&D center of Telecom Italia since 2000, in radio-access and cellular planning group. His activities are in the field of mobile communication systems, especially radio resource and QoS management, system-level analyses, optimization of 3G and B3G systems performances. Since 2002, he has been participating in several IST European research projects as Telecom Italia representative.

Massimo Colonna received the Laurea degree in Electronic Engineering from Politecnico di Torino, Italy, in 1997. In the same year he joined CSELT, the R&D Center of Telecom Italia, Turin, Italy, in the wireless access networks group. His activity was initially related to the fixed wireless access systems (BFWA, LMDS) and then, from 2000, to the wireless LANs. Actually he is involved in research projects dealing with radio resource management in 3G/WLAN integrated networks and with statistical localization within mobile networks.

Giovanna Zarba received the Laurea degree in Electronic Engineering from Politecnico di Torino, Italy, in 1992 and she joined CSELT, the R&D Center of Telecom Italia in the same year. Firstly, her activity was related to the design and development of microwave components for satellite applications. From 2004 she is active in the radio-access and cellular planning department and she has been involved in research projects related to 3G and WLAN networks.