An Empirical Study of the EDCA QoS Mechanism for Voice over WLAN

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***Abstract* — An empirical analysis of EDCA QoS mechanism defined in IEEE 802.11e standard is presented in this paper. Measurements were carried out in controlled laboratory conditions, with and without configured QoS. Mean and maximum delay, jitter and packet loss were measured in voice communications between two clients, first competing with best effort clients, and then competing with voice clients. The results show that QoS mechanism brings significant improvements in the transmission of voice, which is especially significant in the case of highly loaded network.**

***Keywords* — EDCA, IEEE 802.11e, QoS, VoWLAN.**

I. Introduction

MOTIVATED by the promising Voice over IP (VoIP) technology and the wide service availability of Wireless Local Area Networks (WLANs), the application of Voice over WLAN (VoWLAN) is expected to experience growth in the near future. Originally designed for high-rate data traffic, WLANs may experience bandwidth inefficiency when supporting delay-sensitive and low-rate voice traffic.

Recognizing the need to support real time services, in the IEEE 802.11e standard a Quality of Service (QoS) mechanism has been implemented [1]-[4]. IEEE 802.11e defines a new coordination function called Hybrid Coordination Function (HCF). HCF is a centralized coordination function that combines the aspects of Distributed Coordination Function (DCF) and Point Coordination Function (PCF) with enhanced QoS mechanisms, to provide service differentiation. The distributed, contention-based channel access mechanism of HCF is called Enhanced Distributed Channel Access (EDCA). On the other hand the centrally controlled, contention-free channel access mechanism is called HCF Controlled Channel Access (HCCA). In this paper, an empirical analysis of EDCA QoS mechanism is done.

Measurements were carried out in controlled laboratory conditions, with and without configured QoS. Mean and maximum delay, jitter and packet loss are measured for voice communications between two clients, first competing with best effort clients, and then competing with other voice clients.

The paper is organized into five sections. Theoretical basis of EDCA technique is presented in the second section. Measurement topology and method of measurement are given in section three. The fourth section contains the results of the measurements. Conclusions are presented in section five.

II. IEEE 802.11e EDCA

The EDCA provides a differentiated, distributed access to the medium using different priorities for different types of traffic. EDCA defines four Access Categories (ACs) for different types of data traffic: Background (AC_BK), Best Effort (AC_BE), Video (AC_VI) and Voice (AC_VO), where AC_BK has the lowest priority and AC_VO has the highest. Service differentiation is introduced using a different set of parameters for each AC to contend for medium. Frames from different types of data traffic are mapped into different ACs.

Every Access Point (AP) and client station maintains four transmitting queues, one for each AC, and four independent Enhanced Distributed Channel Access Functions (EDCAFs), one for each queue (Fig. 1).

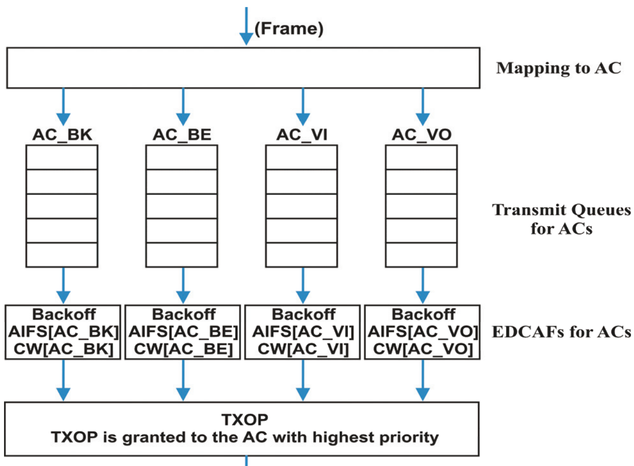

Fig. 1. EDCA queue architecture.

EDCAF is an enhanced version of DCF, and contends for the medium on the same principles of CSMA/CA and

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backoff, but based on the parameters specific to the AC. An EDCAF contends for medium according to the following parameters associated to an AC: AIFS (time period the medium is sensed idle before the transmission or backoff is started), CWmin and CWmax (size limits of Contention Window used for backoff), and TXOP Limit (maximum duration of the transmission after the medium is acquired). Values of EDCA parameters are different for different ACs. The higher priority ACs wait a small AIFS time period before they can access the medium, while the lower priority ACs have to wait a longer AIFS time. The size of Contention Window varies such that the higher priority ACs choose backoff values from a smaller Contention Window compared to the lower priority ACs. TXOP Limit is also set in a way that the higher priority ACs get the access to the medium for longer durations. Basically, the higher priority of an AC corresponds to smaller AIFS, smaller CWmin and CWmax, and larger TXOP Limit.

III. Measuring Topology and Measuring Method

Mean and maximum delay, jitter and packet loss, as basic parameters in the transfer of voice, were measured in voice communications between two clients, competing with other clients in two different cases. In Case 1 two voice clients were competing with best effort clients (network topology in Fig. 2), and in Case 2 voice clients were competing with other voice clients (network topology in Fig. 3). For the purposes of analyzing EDCA efficiency, measurements were made with and without configured QoS mechanism, and subsequently the obtained results are compared.

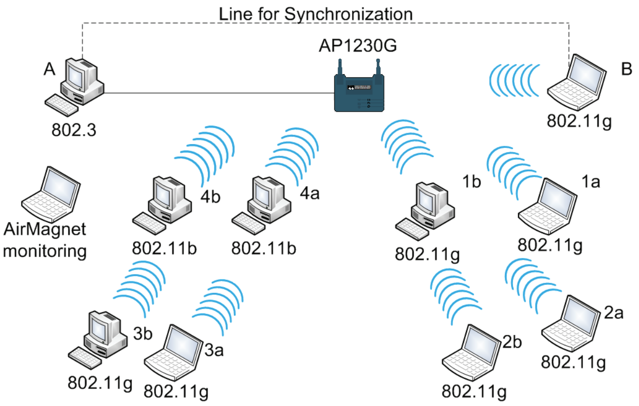

Fig. 2. Measuring topology Case 1 - voice clients competing with best effort clients.

Topologies are based on AP Cisco Aironet 1230G with 12.3(8)JA2 version of IOS, which supports IEEE 802.11g standard. QoS policy, which provides mapping of frames into the appropriate AC and defining QoS for different types of traffic, was configured for measurements with QoS mechanism.

For the purposes of measurements, an open source software for simulating and measuring VoIP traffic, Distributed Internet Traffic Generator [5] (D-ITG) was used. The program consists of several modules of which two have been used: ITGSend - an application that is used to generate the desired type of traffic (Computer A in Fig. 2 and Fig. 3 - source of VoIP packets), and ITGRecv - an application that accepts generated traffic and stores it in log files (Computer B in Fig. 2 and Fig. 3 - destination of VoIP packets).

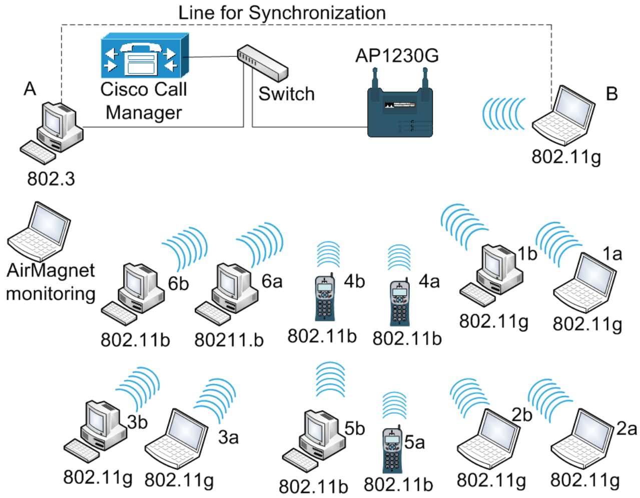

Fig. 3. Measuring topology Case 2 - voice clients competing with other voice clients.

For accurate measurements of delay and jitter, D-ITG requires accurate synchronization between the internal clocks of computers A and B. In order to achieve this, computers A and B were running Ubuntu Linux 7.10, which has an acceptable system delay. Network Time Protocol (NTP) server was installed on computer A, while computer B was synchronized with the server via a special line for synchronization (Fig. 2 and Fig. 3). Synchronization error of this method is less than 100µs.

In Case 1 measurements were carried out by measuring voice parameters, first without competing best effort clients, and then the network is loaded with pairs of clients. One of these clients was source of TCP best effort traffic and the other one was destination. The maximum number of competing clients was 8, and some of them were using IEEE 802.11b WLAN card (Fig. 2). In that case, a mixed mode WLAN was active. For generating TCP best effort traffic, D-ITG was used. Competing clients were configured with a requested data rate equal to 10% of maximum supported data rate of WLAN.

In Case 2, voice parameters were measured without competing voice clients first, and then the network was loaded with pairs of voice clients, with established voice connections. The maximum number of competing clients was 12, and some of them were using IEEE 802.11b WLAN card (Fig. 3). In that case, a mixed mode WLAN was active.

Computers with installed Cisco IP Communicator v2.1 and Cisco 7920 Wireless Phones are used as competing voice clients (Fig. 3). Voice communication between voice clients was maintained by Cisco Call Manager v4.1. Clients were using G.729.2 codec sending VoIP packets at 20ms interval, with deactivated Real-time Transport

Protocol (RTP) header compression and Voice Activity Detection (VAD).

As it is well known, communication parameters of WLAN client, depend on its distance from the AP. Distances of clients from the AP in measuring topology were relatively small. Because of that, measurements were performed by forcing WLAN to operate at maximum data rates 1, 2, 5.5, 11, 24, 36 and 54Mb/s. In this way, the analysis included cases of real distances of clients from the AP.

Each individual measurement was performed 5 times in Case 1 and 3 times in Case 2. As a result, the mean values of all individual measurements are used, except for maximum delay, where we used the maximum value from all individual measurements. The duration of each individual measurement was 30 seconds, and before each measurement synchronization was performed.

Measurements were performed on the first channel - 2412MHz, which was found to be free. During the measurements WLAN was configured with Open Key authentication system. All other parameters of AP were set to default values. For additional control and verification of AP and client functioning software, Airmagnet Laptop Analyzer [6] was used. As EDCA parameters for each AC, default values for AP Cisco Aironet 1230G were configured (Table 1).

TABLE 1: VALUES OF EDCA PARAMETERS.

Access Category		AC_BK	AC_BE	AC_VI	AC_VO
CWmin	AP	5	5	4	2
	Client	5	4	4	2
CWmax	AP	10	6	5	4
	Client	10	10	5	3
AIFS	AP	7	3	1	1
	Client	7	3	2	2
TXOP Limit	AP	0	0	3008	1504
	Client	0	0	3008	1504

IV. RESULTS

A. Measurement results of Case 1 - voice clients competing with best effort clients

Measurement results of Case 1 for the mean and maximum delays for all data rates and both cases of configured and no configured QoS mechanism are shown in Fig. 4. Mean jitter values, for these cases are shown in Fig. 5. The percentages of lost VoIP packets for all data rates and both cases for QoS mechanism are presented in Table 2.

Measurement results show that the use of QoS mechanisms brings significantly lower values of the mean and maximum delay (Fig. 4), compared to the case without configured QoS. Mean delays with the applied QoS mechanism were under 10ms, even at a maximum number of competing best effort clients. On the contrary, it was not the case without QoS, especially for data rates 1 and 5.5Mb/s with a highly loaded network (6 or 8 competing best effort clients), where the mean delays were comparable to or higher than the maximum allowed value of 150ms.

Maximum delays with the applied QoS mechanism were reduced to values below 100ms in all cases. That was not the case without QoS when the maximum delays were greater than 150ms, for all data rates with a highly loaded network (6 or 8 competing best effort clients).

By applying QoS mechanism the average jitter value was reduced several times (Fig. 5). For data rates above 1Mb/s the average jitter is reduced to values less than 1.5ms, even with a highly loaded network. For a data rate of 1Mb/s the average jitter is reduced for approximately 50%.

In Table 2 it is shown that the implementation of QoS mechanism reduces percentages of lost VoIP packets to 0 in all cases, except for a data rate of 1Mb/s. For a data rate of 1Mb/s the percentages of lost VoIP packets were approximately the same in both cases of configured and no configured QoS mechanism.

B. Measurement results of Case 2 - voice clients competing with other voice clients

Measurement results of Case 2 for the mean and maximum delays for all data rates and both cases of configured and no configured QoS mechanism are shown in Fig. 6, while the mean jitter values, for these cases are shown in Fig. 7. The percentages of lost VoIP packets for all data rates and both cases for QoS mechanism are presented in Table 3.

For data rates 1 and 2Mb/s, the maximum number of competing voice clients was 6 and 10, respectively. In both cases, after adding one additional pair of voice clients, communication with Call Manager was interrupted. It is important to emphasize that configured QoS policy was not concerned with signaling between voice clients and Call Manager.

Measurement results show that the use of QoS mechanisms in most cases brings significantly lower values of the mean and maximum delay (Fig. 6). Mean delays with the applied QoS mechanism were under 3.5ms, even at a maximum number of competing voice clients. On the contrary, it was not the case without QoS, especially for data rates 1, 2 and 5.5Mb/s, where the mean delays were comparable to or even higher than the maximum allowed value of 150ms.

Maximum delays with the applied QoS mechanism are reduced to values below 100ms, except for a data rate of 5.5Mb/s and 12 competing voice clients. That was not the case without QoS when the maximum delays were greater than 150ms, for data rates 1, 2 and 5.5Mb/s and a highly loaded network.

In Fig. 7 it is shown that the implementation of QoS mechanism reduced the average jitter values for several times. For data rates 1 and 2Mb/s jitter is reduced to less than 2ms, and for greater data rates less than 1ms.

Use of QoS mechanisms brings significantly lower percentages of lost VoIP packets, especially for data rates 1, 2 and 5.5Mb/s and a maximum number of competing voice clients (Table 3). For greater data rates this percentage is reduced to 0.

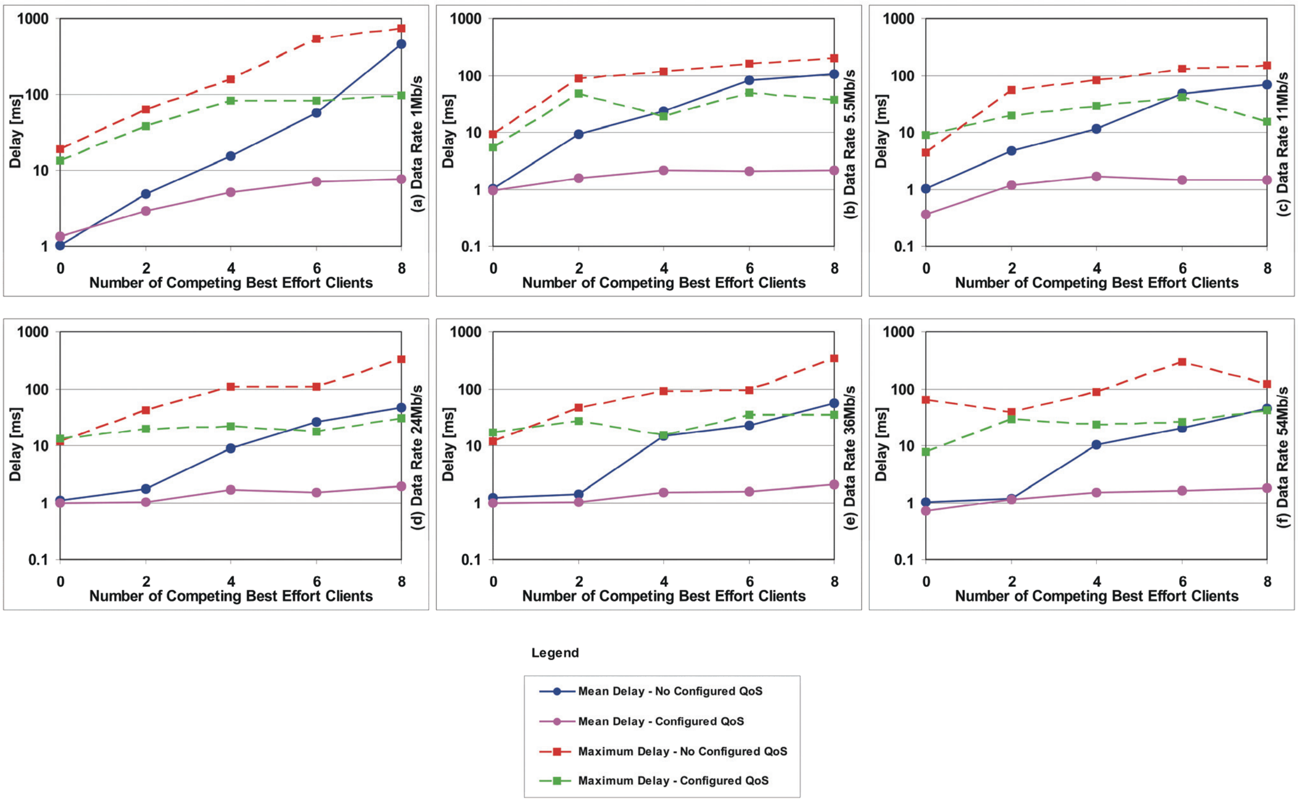

Fig. 4. Mean and maximum delay depending on number of competing best effort clients and data rate for measuring topology Case 1.

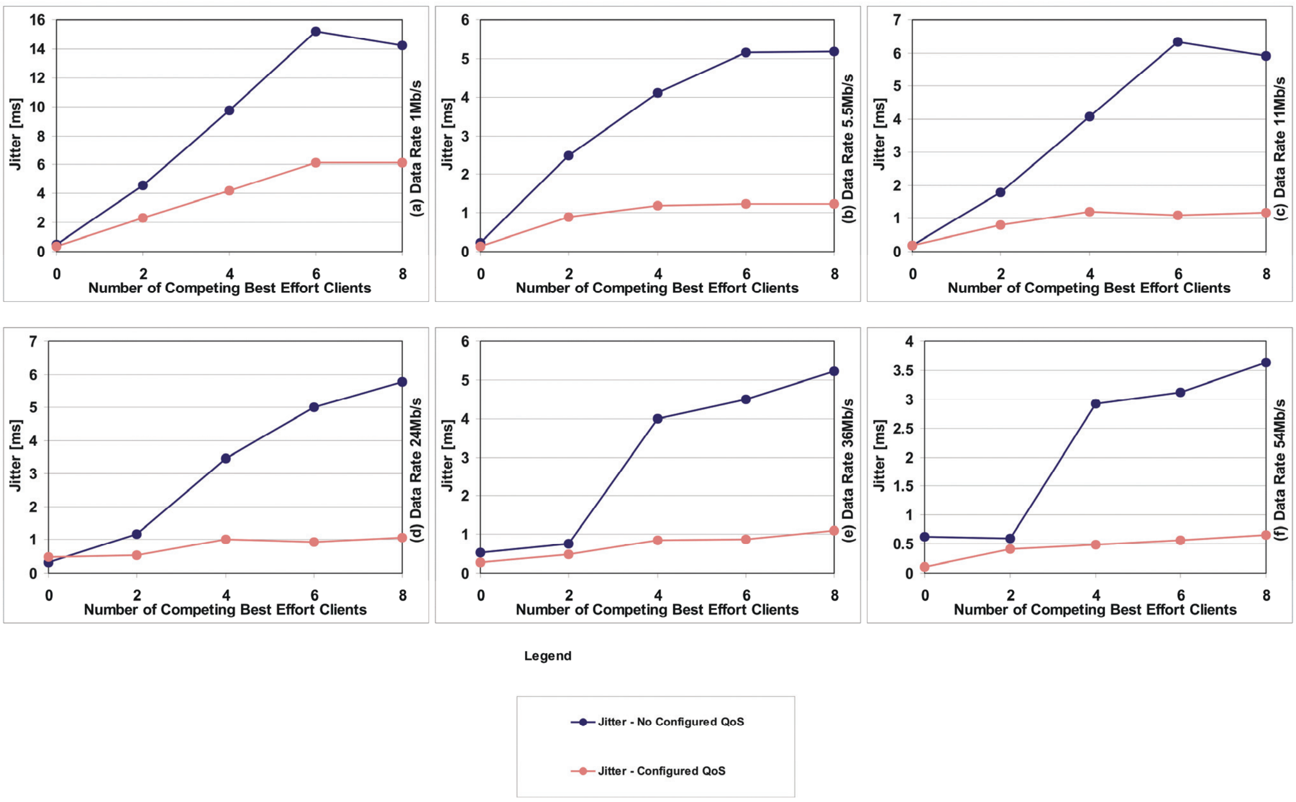

Fig. 5. Mean jitter depending on number of competing best effort clients and data rates for measuring topology Case 1.

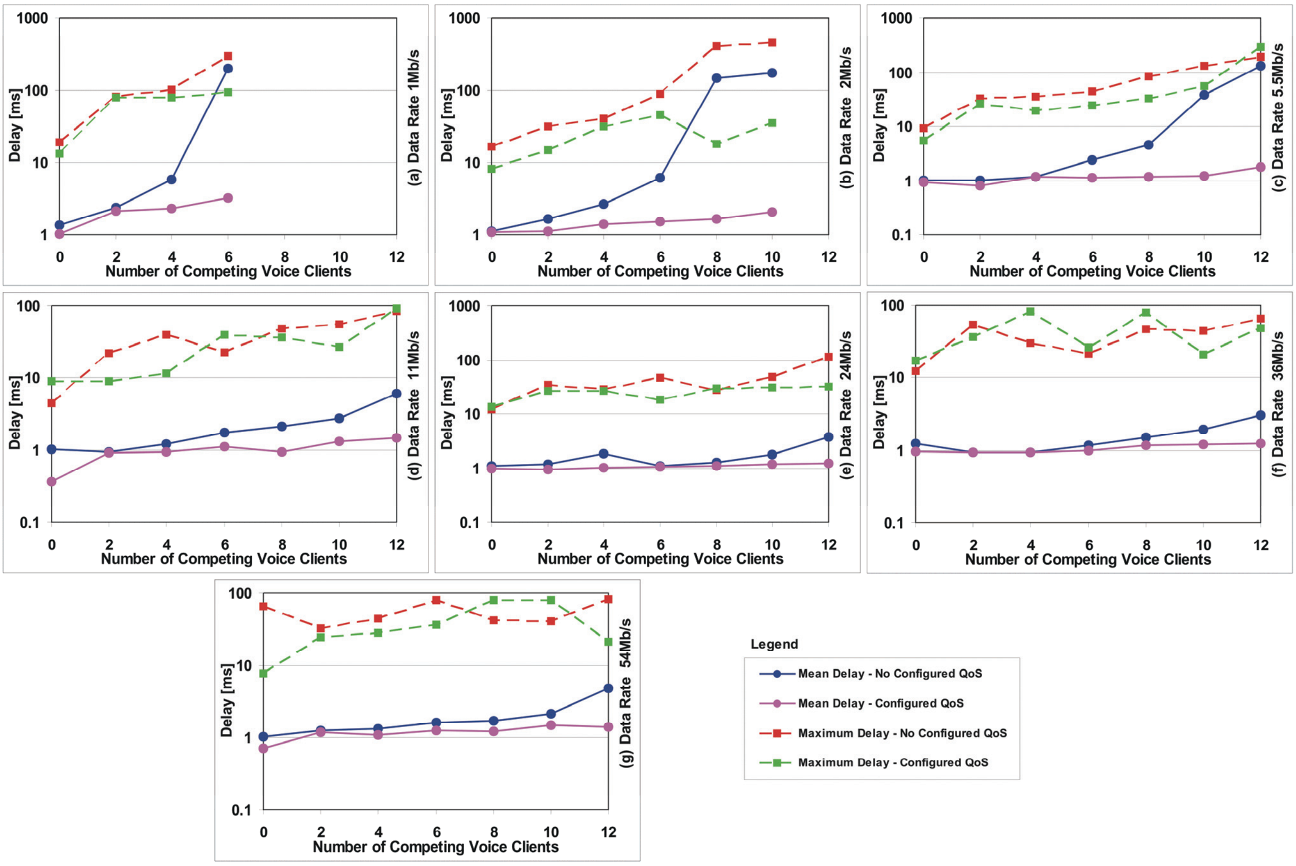

Fig. 6. Mean and maximum delay depending on number of competing voice clients and data rate for measuring topology Case 2.

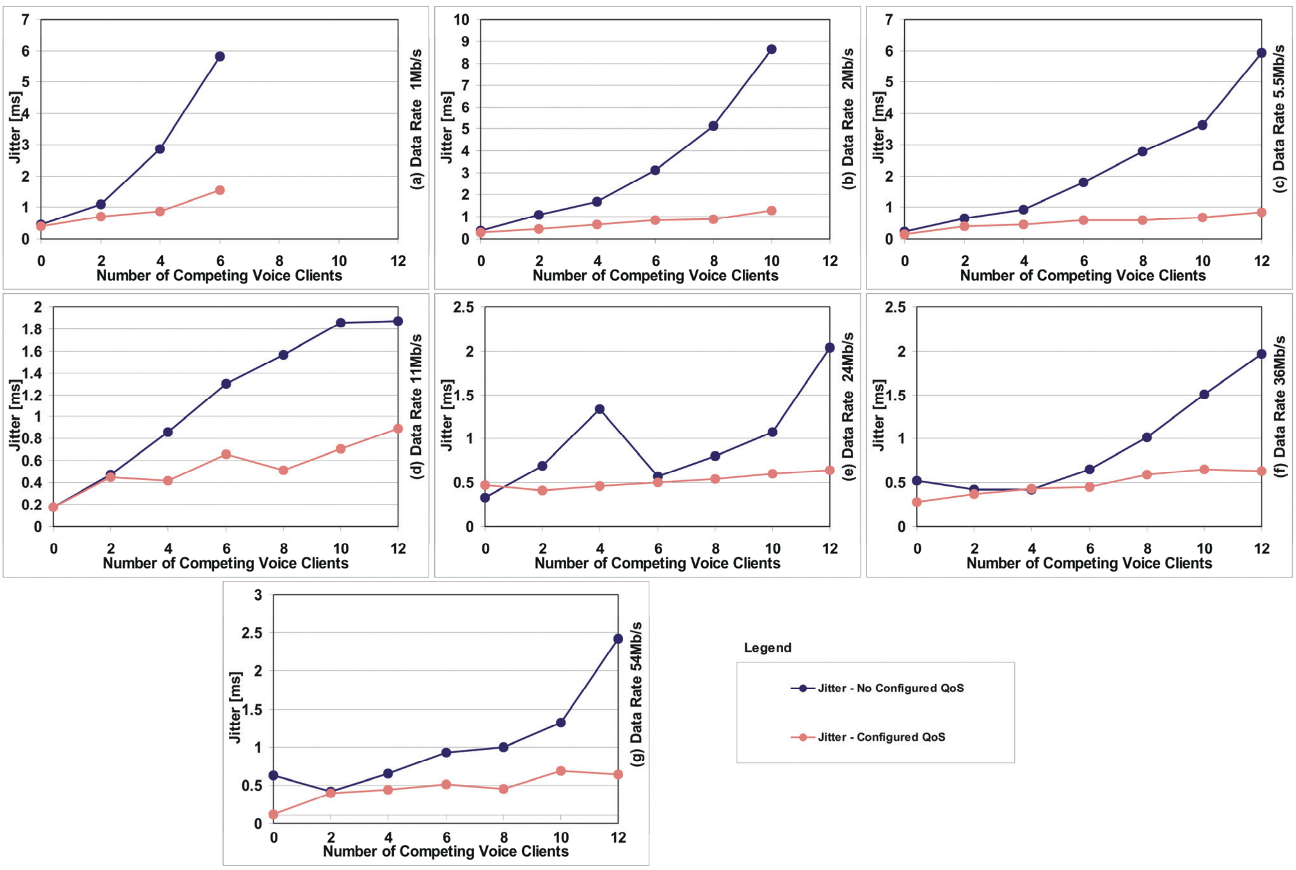

Fig. 7. Mean jitter depending on number of competing voice clients and data rates for measuring topology Case 1.

TABLE 2: PERCENTAGE OF LOST VOIP PACKETS IN CASE 1 - VOICE CLIENTS COMPETING WITH BEST EFFORT CLIENTS.

Number of Competing Best Effort Clients	0	2	4	6	8
1Mb/s - No Conf. QoS	0.00	0.00	0.00	0.07	0.20
1Mb/s - Conf. QoS	0.00	0.00	0.13	0.07	0.13
5.5Mb/s - No Conf. QoS	0.00	0.07	0.13	0.07	0.07
5.5Mb/s - Conf. QoS	0.00	0.00	0.00	0.00	0.00
11Mb/s - No Conf. QoS	0.00	0.00	0.07	0.33	0.20
11Mb/s - Conf. QoS	0.00	0.00	0.00	0.00	0.00
24Mb/s - No Conf. QoS	0.00	0.00	0.20	0.27	10.93
24Mb/s - Conf. QoS	0.00	0.00	0.00	0.00	0.00
36Mb/s - No Conf. QoS	0.00	0.00	0.40	0.40	0.47
36Mb/s - Conf. QoS	0.00	0.00	0.00	0.00	0.00
54Mb/s - No Conf. QoS	0.00	0.00	0.27	0.40	0.47
54Mb/s - Conf. QoS	0.00	0.00	0.00	0.00	0.00

TABLE 3: PERCENTAGE OF LOST VOIP PACKETS IN CASE 2 - VOICE CLIENTS COMPETING WITH OTHER VOICE CLIENTS.

Number of Competing Voice Clients	0	2	4	6	8	10	12
1Mb/s - No Conf. QoS	0.00	0.00	0.07	15.13	-	-	-
1Mb/s - Conf. QoS	0.00	0.07	0.00	0.00	-	-	-
2Mb/s - No Conf. QoS	0.00	0.00	0.00	0.07	3.00	47.83	-
2Mb/s - Conf. QoS	0.00	0.00	0.00	0.00	0.07	0.14	-
5.5Mb/s - No Conf. QoS	0.00	0.00	0.00	0.00	0.07	0.20	24.93
5.5Mb/s - Conf. QoS	0.00	0.00	0.00	0.00	0.00	0.07	0.07
11Mb/s - No Conf. QoS	0.00	0.00	0.00	0.07	0.00	0.13	0.13
11Mb/s - Conf. QoS	0.00	0.00	0.00	0.00	0.00	0.00	0.00
24Mb/s - No Conf. QoS	0.00	0.00	0.00	0.07	0.00	0.07	0.07
24Mb/s - Conf. QoS	0.00	0.00	0.00	0.00	0.00	0.00	0.00
36Mb/s - No Conf. QoS	0.00	0.00	0.00	0.00	0.07	0.13	0.13
36Mb/s - Conf. QoS	0.00	0.00	0.00	0.00	0.00	0.00	0.00
54Mb/s - No Conf. QoS	0.00	0.00	0.00	0.00	0.00	0.13	0.20
54Mb/s - Conf. QoS	0.00	0.00	0.00	0.00	0.00	0.00	0.00

V. CONCLUSION

Experimental analysis has shown that the EDCA QoS mechanism brings significant improvements in the transmission of voice, which is especially significant in the case of a highly loaded network. Measurement results have shown that the use of QoS mechanisms brings significantly lower values of the mean and maximum delay, jitter and lost packets, compared to the case without QoS mechanism.

Further research should be directed toward optimization of EDCAF parameters (CWmin, CWmax, AIFS, TXOP) analyzing their impact on the quality of voice services, and determining their impact on the maximum achievable throughput among all clients in network.

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REFERENCES

[1] IEEE Standard 802.11 – 2007.

[2] D. Gu and J. Zhang; "QoS enhancement in IEEE 802.11 wireless local area networks," *Communications Magazine, IEEE* , vol.41, no.6, pp. 120- 124, June 2003.

[3] P. Wang; H. Jiang; and W. Zhuang; "IEEE 802.11e enhancement for voice service," *Wireless Communications, IEEE* , vol.13, no.1, pp. 30- 35, Feb. 2006.

[4] O. Aboul-Magd, *Wireless Local Area Networks Quality of Service.* New York: IEEE Press, 2007, pp. 97–145.

[5] http://www.grid.unina.it/software/ITG/

[6] www.airmagnet.com