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Rastin Pries, Andreas Mäder, Dirk Staehle

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University of Würzburg
Department of Computer Science
Am Hubland, D-97074 Würzburg, Germany
pries@informatik.uni-wuerzburg.de

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Rastin Pries, Andreas Mäder, Dirk Staehle
University of Würzburg
Department of Computer Science
Am Hubland, D-97074 Würzburg, Germany
pries@informatik.uni-wuerzburg.de

Abstract

The increasing number of Voice over IP (VoIP) phones in wired and wireless technologies leads us to the performance discussion of VoIP traffic in IEEE 802.11 WLAN environments. Originally designed to transport best effort data traffic, WLANs may experience some problems with delay-sensitive and low-rate voice traffic. This paper investigates the Voice over WLAN (VoWLAN) capacity for different WLAN standards. Our simulations show that the IEEE 802.11e Quality-of-Service (QoS) extension does not increase the system performance in terms of maximum number of supported VoWLAN users. Therefore, we try to improve the capacity by using a header compression mechanism which reduces the upper layer headers to around 10 percent of its original size.

1 Introduction

Wireless LANs were designed to transport best effort data traffic, in places where a wired LAN cannot be deployed. In such scenarios, the network is mainly used for applications like email or web browsing. However, the dramatic growth of VoIP users in wired networks leads to performance discussions of VoWLAN. E.g., voice applications like Skype have been downloaded over 216 million times. Nevertheless, Skype is mostly used for personal calls, where performance problems should not appear due to the limited number of users. Therefore, we look on the performance of the ITU-T G.711 [1] and the ITU-T G.729 [2] voice codecs which are widely used in companies.

In order to evaluate the ability of wireless LAN to support such a QoS demanding application, we preformed complex simulation studies. We investigate the capacity in IEEE 802.11b [3] and IEEE 802.11g [4] environments with and without the QoS extension IEEE 802.11e [5]. Furthermore, we take a look on the capacity gain when using header compression for the Real-Time Protocol (RTP), User Datagram Protocol (UDP), and Internet Protocol (IP).

This paper is organized as follows. In the second section, we give an overview of the related work. This is followed by Section 3 where we introduce the wireless LAN medium access protocols with focus on the just completed QoS standard. In Section 4 robust header compression is explained. Our simulation scenarios are shown in Section 5, and Section 6 illustrates the results of our simulations. Finally, we will conclude this paper in Section 7 and give a brief outlook.

2 Related Work

Some papers have been published which concentrate on the voice over IP capacity in simple IEEE 802.11 networks [6–13]. The first four papers focus on the VoIP performance in 802.11b networks using the normal Distributed Coordination Function (DCF). They use simulation, experimental setups, and analysis to evaluate the number of supported voice users in such networks. These papers show that only around six VoWLAN users can be supported in 802.11b networks when using the ITU-T G.711 voice codec. Nevertheless, they do not take a look at the QoS standard IEEE 802.11e.

The paper from Veeraraghavan [10] calculates the maximum number of voice calls in a polling-based wireless LAN environment. However, centrally controlled MAC protocols have a higher complexity and no access point vendor has yet implemented the used Point Coordination Function (PCF).

The three papers [11–13] examine the performance for different wireless LAN standards. The last one takes the new IEEE 802.11e standard [5] into account where it is possible to prioritize voice traffic. The results show that up to 15 VoIP users can be supported in an IEEE 802.11b environment by using the QoS extension of the wireless LAN standard. We will take a look if we see the same increase from six users without QoS extension [9] to 15 users with the QoS extension [13].

As far as we know, there is not yet any paper published where the VoIP capacity for several wireless LAN standards together with Robust Header Compression (ROHC) [14] is simulated. Therefore, we will show the VoWLAN capacity for network configurations with and without header compression.

3 Wireless LAN MAC protocol overview

The Distributed Coordination Function (DCF) is the primary access mode using the CSMA-CA protocol for sharing the wireless medium as shown in Fig. 1. Stations which want to transmit a packet have to compete with each other for access and all stations have equal rights. However, wireless LAN stations are not able to detect a collision on the medium.

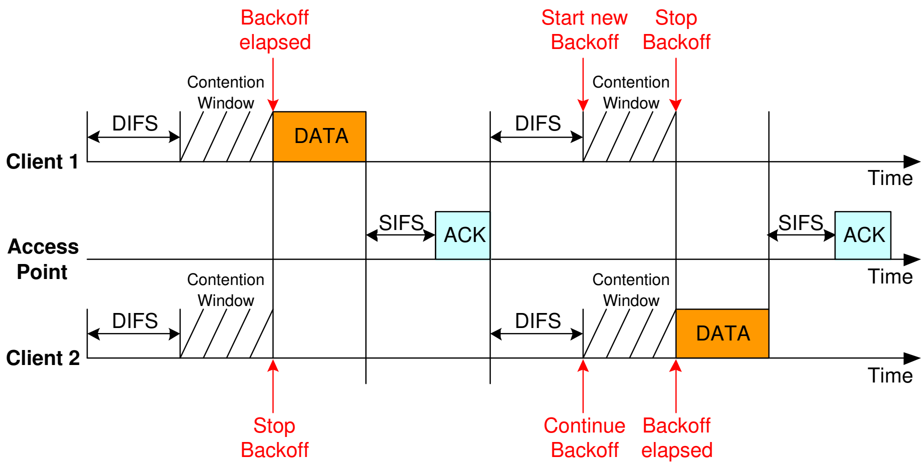

Figure 1: Medium Access Example for DCF Clients

Therefore, an acknowledgment scheme has to be performed. If no acknowledgment is received by the sending station it will retransmit the packet. In order to reduce the collision probability on the wireless medium, the stations sense the medium for a period of time called Distributed Inter-Frame Space (DIFS) and perform a backoff before transmitting a packet.

The DCF is extended in the IEEE 802.11e [5] by the Enhanced Distributed Channel Access (EDCA). In contrast to the DCF, EDCA is based on different priorities. The contention window and backoff times are adjusted to change the probability of gaining medium access to favor higher priority classes. It supports eight different priorities from 0 to 7 as defined by the IEEE 802.1D standard [15], shown in Table 1.

These priorities are mapped to four Access Categories (ACs) as shown in Fig. 2. The ACs are sorted from AC0 to AC3 with AC3 having the highest priority for medium access. The service differentiation according to these ACs is achieved by varying the amount of time a station senses the channel to be idle before starting the contention window (carrier sensing interval), the length

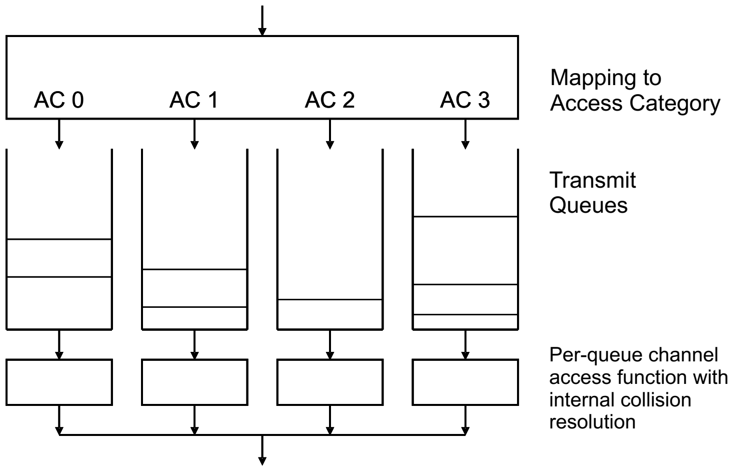

Figure 2: HCF Access Categories

of the contention window to be used, and the duration a station may transmit after it acquires the right to transmit (TXOPLimit).

For each Access Category (AC) an enhanced variant of the DCF called Channel Access Function (CAF) contends for the medium using a set of EDCA parameters from the EDCA Parameter Set element. Each CAF represents a virtual DCF station with own parameters. The EDCA parameter set used by each CAF is defined by the Arbitration Inter-Frame Space (AIFS) Number, CWmin, CWmax, and TXOPLimit.

In DCF mode, a station uses a carrier sensing interval of DIFS to decide if the medium is idle. In EDCA mode, different time intervals are used. These AIFSs are usually longer time periods as the DIFS. The length of the AIFS can be calculated as shown in Equation 1.

$$AIFS_{length} = AIFS \cdot slot_{time} + sifs_{time} \tag{1}$$

With these different interframe spaces, a certain prioritization can be reached. If two stations want to transmit at the same time, the station with the shorter IFS will get access. Therefore, lower priorities use larger IFSs.

In EDCA mode, the backoff procedure of the DCF is changed. The basic mechanism defines that a number of backoff slots is taken from the interval $[0, CW]$. The number is chosen uniformly

Table 1: User Priority to Access Category Mapping

User Priority	802.1D Designation	AC	Designation (Informative)
1	Background (BK)	0	Best Effort
2	-	0	Best Effort
0	Best Effort (BE)	0	Best Effort
3	Excellent Effort (EE)	1	Video Probe
4	Controlled Load (CL)	2	Video
5	Video (VI)	2	Video
6	Voice (VO)	3	Voice
7	Network Control (NC)	3	Voice

Table 2: Access categories and their settings

AC	CWmin	CWmax	AIFS
0	aCWmin	aCWmax	7
1	aCWmin	aCWmax	3
2	$\frac{aCWmin+1}{2} - 1$	aCWmin	2
3	$\frac{aCWmin+1}{4} - 1$	$\frac{aCWmin+1}{2} - 1$	2

Table 3: Access categories and their values using IEEE 802.11g

AC	CWmin	CWmax	AIFS	TXOPLimits
0	15	1023	72 μs	One MSDU
1	15	1023	37 μs	One MSDU
2	7	15	28 μs	3008 μs
3	3	7	28 μs	1504 μs

distributed. Initially, the CW value is set to CWmin. Whenever a packet loss occurs, the CW value is increased by $CW' = (CW + 1) \cdot 2 - 1$ until the maximum value CWmax is reached.

For DCF mode in an IEEE 802.11g network, the default values are $CWmin = 15, CWmax = 1023$. EDCA uses these values to define different priorities. The standard settings for every IEEE 802.11 standard can be seen in Table 2. If we take the standard settings of CWmin and CWmax from the IEEE 802.11g network, we will receive the parameter set for EDCA as shown in Table 3. Here, the highest priority class is assigned a CWmin of 3 and a CWmax of 7 while the lowest priority class is assigned the values 15 and 1023. This will lead to different mean contention window sizes. Clearly, a station with a lower mean contention window will get access to the medium much more often. Thus, an additional prioritization can be reached. A more detailed description of the EDCA mechanism can be found in [5], [16], or [17].

Two more medium access mechanisms are proposed by the IEEE 802.11 standard. The Point Coordination Function (PCF) as a simple polling mechanism and the Hybrid coordination function Controlled Channel Access (HCCA) with a prioritized polling mechanism proposed in the IEEE 802.11e standard. Both mechanisms are not considered for our simulations.

4 Robust Header Compression

As wireless networks have high bit error rates and high latencies, it is important to use the available resources as efficiently as possible. When using VoIP, the RTP/UDP/IP header is about one third of the complete packet size. For example, if a G.711 voice codec is used with a 10 ms frame size, a voice packet payload is 640 bit, while the RTP/UDP/IP header overhead is 320 bit.

Though, the large packet header overhead can significantly affect the capacity of VoWLAN. These headers are very important if we look at the end-to-end connection, comprised of multiple hops, but over just one link the headers are not useful. It is possible to compress the headers, sometimes to more than 90% which saves bandwidth and the expensive resources can be used efficiently.

Let us now take a look at the efficiency of these techniques. We consider a voice over IP call in todays wireless LANs. Every packet has an RTP, UDP, and IP header. The IP version 4 header is 20 bytes, the UDP header 8 bytes, and RTP has a 12 bytes header, which leads to a complete header size of 40 bytes. With Robust Header Compression [14] this header size can be compressed to an average size of 4 bytes, see Figure 3.

Figure 3: basic technology of header compression

The ROHC scheme make use of the predictable behavior of the header sequence. More details about header compression can be found in [18] or [19].

5 Simulation Setup

To simulate the capacity of VoIP users in wireless LANs, we used the well-known OPNET Modeler 11.5 [20]. The tool already offers a large variety of voice codecs which are easy to modify. We focus on the ITU-T G.711 [1] and G.729 [2] voice codecs because they are the most widley used codecs in business environments. The parameters can be seen in Table 4.

Table 4: Voice over IP traffic models

Parameter	G.711	G.729
Frame size	10 ms - 40 ms	10 ms
Coding rate	64 kbps	8 kbps
ROHC (yes/no)	both	both
Lookahead size	0 ms	5 ms
Comp./decomp. delay	0 ms	0 ms

We set the compression and decompression delay to 0 because we just want to look on the delay, jitter, and packet loss on the wireless interface only. According to the ITU-T definition [21], the tolerable packet loss rate is 1 to 3 percent, and the one way tranmsission delay is preferably shorter than 150 ms but should not exceed 400 ms. Therefore, we set the maximum end to end delay on the wireless interface to 100 ms and if we add the additional compression delay and backbone delay to these limits, the total delay should not exceed 400 ms.

Our simulation setup can be seen in Figure 4. The VoIP clients are uniformly distributed in the wireless LAN cell and they are all configured with the same data rates. Furthermore, to neglect the auto-fallback of wireless LAN, we reduce the cell size to 50 m.

In order to evaluate the maximum VoWLAN capacity for the different settings, we performed at least 10 simulation runs for each parameter, using different seeds. The maximum number of voice clients is reached, if in one of the simulation runs, either the packet delay between a client and the access point exceeds 100 ms or the packet loss is above 1 percent.

6 Simulation Results

In our first scenario, we want to evaluate the maximum number of voice calls in an IEEE 802.11b environment using the normal Distributed Coordination Function. Therefore, we use the G.711

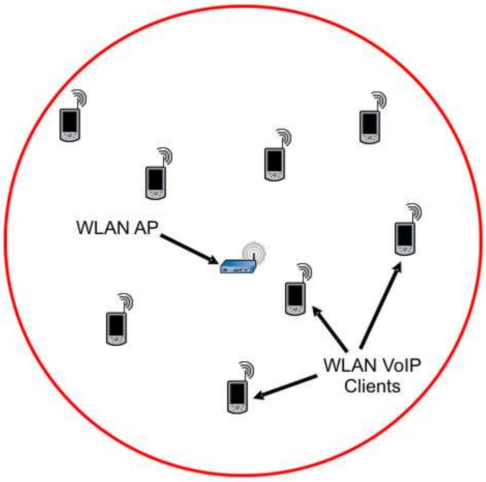

Figure 4: Typical simulation scenario

voice codec with a data rate of 64kbps. We increase the frame size from 10 ms up to 40 ms. Figure 5 shows the results.

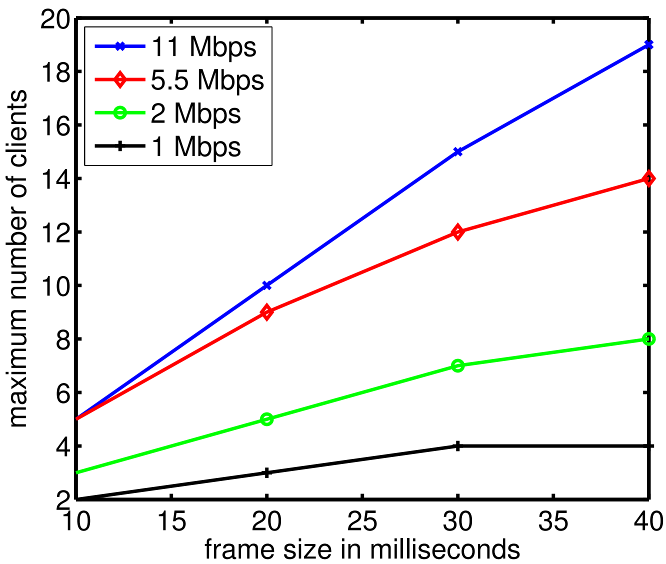

Figure 5: DCF using IEEE 802.11b data rates

Taking a look at the curve for 11 Mbps, we see that the maximum number of supported voice calls varies between 5 for 10 ms frame size and 19 for a frame size of 40 ms. From these differences it is obvious that the overhead of the DIFS and the backoff has a large influence on the system performance. Therefore, we change the medium access mechanism from DCF to EDCA, where the voice calls have the highest priority with the shortest AIFS and contention windows. The results are shown in Fig. 6.

At the first glance, it is surprising that the number of supported voice calls does not increase. Taking a closer look on the limiting factors for voice calls reveals the problem in an EDCA network. In contrast to the previous results where the end to end delay of the data traffic is the limiting factor, in an EDCA network, the dropped data packets due to an exceeded retry counter limits the number of users. Here, we have very short contention windows. For the first transmission, the

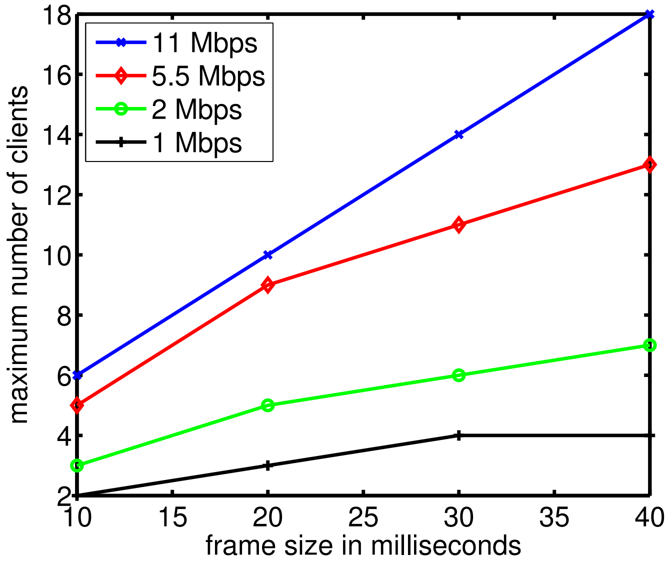

Figure 6: EDCA using IEEE 802.11b data rates

backoff is chosen between 0 and 7. If we assume that every station is in a saturated mode, the collision probability is

$$P_{collision} = 1 - \left(\frac{CWmin}{CWmin + 1} \right)^n \tag{2}$$

Here, n is the number of stations, including the access point. If we are in a saturated mode using the IEEE 802.11b with an initial backoff between 0 and 7 and 5 clients and one access point are in the system, the collision probability already exceeds 50 percent.

We have seen that shorter backoff intervals do not lead to a higher capacity of the network. Now, let us take a look at the capacity if we use robust header compression. As described in Section 4, it is possible to reduce the header size to 2-4 bytes. Due to the fact that we sometimes have to transmit the complete RTP/UDP/IP Header, we get to an average header size of 4 bytes. Table 5 illustrates the gain when using robust header compression.

Table 5: Header compression gain in IEEE 802.11b networks

Mbps	maximum gain	average gain
1	1.50	1.27
2	1.33	1.13
5.5	1.07	1.02
11	1.10	1.03

For a 1 Mbps connection, we have seen an average gain of 27 %. This is quite a high percentage, but the number of supported voice clients is still very small. The maximum gain was achieved when using a 10 ms frame size. Here, the number of supported clients has increased from 2 to 3.

In our first simulations, we have seen that the gain of using the EDCA mechanism nor robust header compression has really improved the wireless LAN performance. Now, we take a closer look on the performance of IEEE 802.11g networks. Here, the contention window sizes change due to a shorter slot time as shown in Table 3. Furthermore, we have higher data rates of up to

54 Mbps. Like before, we start with a simple simulation scenario using the DCF. Fig. 7 shows the results.

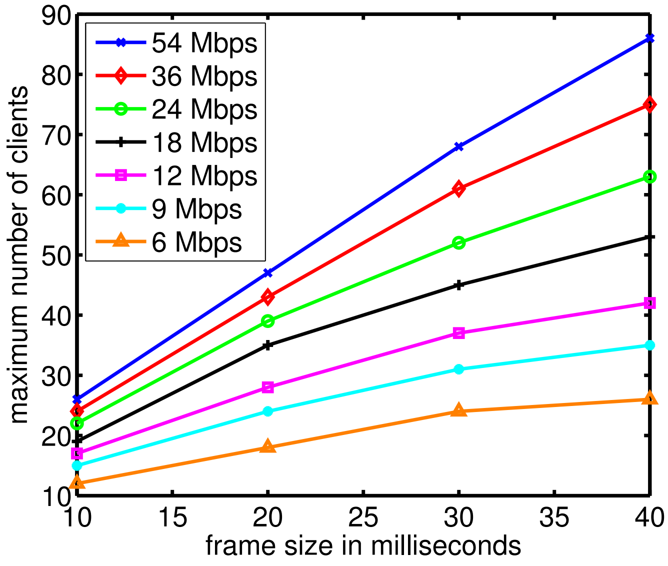

Figure 7: DCF using IEEE 802.11g data rates

If we take a look at the curve for 6 Mbps, we will see a better performance compared to the results of an 11 Mbps connection. This performance increase depends on the backoff interval and the shorter slot length. On one hand, the CWmin is reduced from 31 to 15 and on the other hand, which is much more important, the slot time with which the contention window is multiplied is also reduced from 20 μs down to 9 μs.

The second simulation runs for the IEEE 802.11g standard are performed with the EDCA mechanism. Here, the minimum and maximum contention windows are reduce to 3 and 7 respectively. The results, as shown in Fig. 8, were expected.

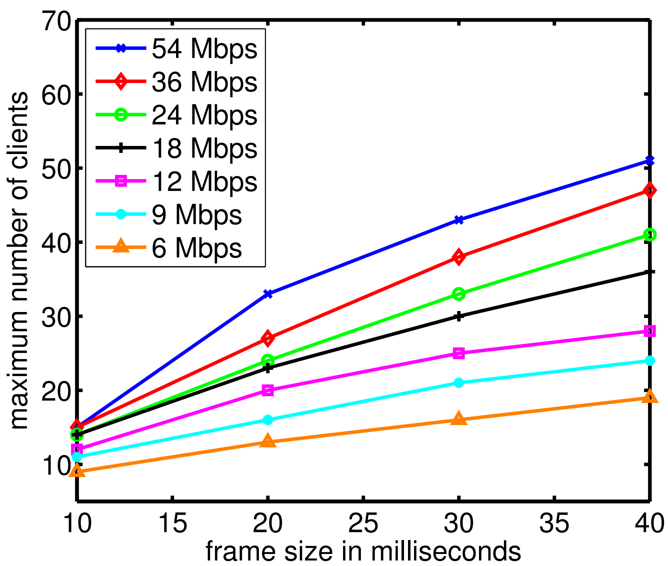

Figure 8: EDCA using IEEE 802.11g data rates

The number of supported voice calls is, compared to the results for the DCF, much lower. If we take for example the curve for 54 Mbps, the number of supported clients with a frame size of 40 ms decreases from 86 down to 51.

Therefore, we have to be able to set the contention windows individually at the access points depending on the number of VoIP users in our wireless network. We can even go further and say that the access points should adapt the contention windows depending on the different queue sizes and broadcast these settings to the clients.

Finally, we want to take a look at the advantages of header compression in IEEE 802.11g networks, this time for DCF and EDCA. Fig. 9 and 10 show the results. The solid lines illustrate the results when using header compression and, for comparison, the dashed lines show the maximum number of supported voice clients when no header compression is used.

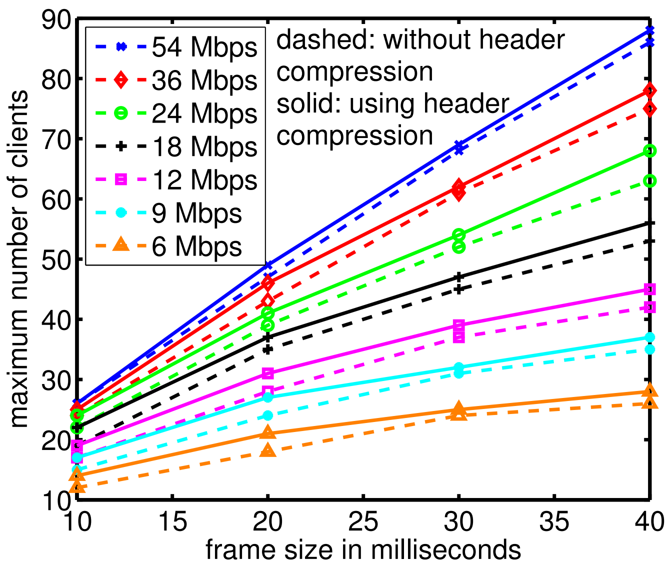

Figure 9: DCF using IEEE 802.11g data rates with and without header compression

The results again show only a small performance increase with robust header compression. The maximum gain is again achieved for the lowest data rate, here for 6 Mbps with 22 %. If we calculate the average gain over all data rates, we will get 6.25 %.

We have seen that the performance does not increase much if we use robust header compression for a voice codec with a high data rate of 64 kbps. In order to show the gain for a low rate voice codec, we changed the voice codec to G.729 with the parameters described in Table 4. The G.729 voice codec has a data rate of 8 kbps. With a frame size of 10 ms, the header overhead is four times larger than the payload. Therefore, we expected a large performance gain when using header compression for this codec. However, the results show that the gain is similar to the G.711 codec. The results are shown in Table 6.

Table 6: Header compression gain with G.729

Access mechanism	maximum gain	average gain
DCF	1.27	1.14
EDCA	1.22	1.11

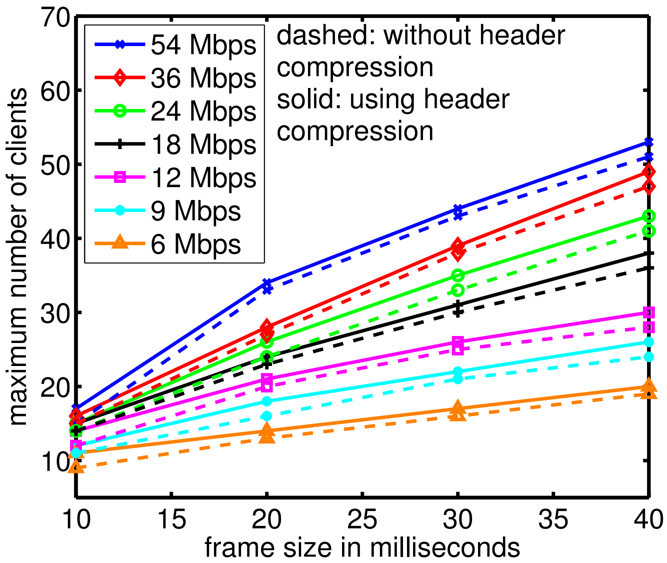

Figure 10: EDCA using IEEE 802.11g data rates with and without header compression

We think that this small gain does not cope the extra effort for implementing a header compression mechanism into the clients.

One way to support more voice clients in a wireless LAN network might be the multiplexing of voice streams on the wireless LAN downlink. Sze [22] has proposed such a mechanism and claim to have a gain of around 100 percent.

7 Conclusion

In this paper, we have seen the number of supported VoIP clients with and without header compression. The results have shown that we need to automatically adapt the parameters for the contention windows when using the QoS EDCA standard. If we have a larger number of VoIP users in our network, the access point should increase the contention window size in order to minimize the collision probability. Furthermore, we have seen that the gain of using header compression for voice over IP traffic does not cope the extra overhead.

In future work we want to implement an automatic adaptation of the contention window sizes according to the load in a wireless LAN cell and we will take a look on some traffic mix scenarios where we place FTP clients in our networks.

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