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The Capability of the EDCA Mechanism to Support Voice Traffic in a Mixed Voice/Data Transmission over 802.11e WLANS: an Experimental Investigation

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The capability of the EDCA mechanism to support voice traffic in a mixed voice/data transmission over 802.11e WLANs - an experimental investigation

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Abstract

In this paper we experimentally evaluate the capability of the EDCA mechanism to support voice traffic in a mixed voice/data transmission over 802.11e WLANs. In particular we investigate how real-time voice transmission can be supported by tuning four EDCA parameters, namely AIFSN, CWmin , CWmax, and TXOP and how this impacts on background data transmission. The experimental set-up involves fifteen VoIP terminals sending bi-directional traffic between wired and wireless subnets and another station injecting various types of heavy background loads to the wireless subnet. End-to-end voice transmission quality is predicted from time-varying transmission impairments with the use of the latest Appendix to the ITU-T E-model. Our experimental results show that the AIFSN parameter more effectively protects voice calls against background data traffic than CWmin. We also demonstrate that tuning of the TXOP parameter does not improve the quality of voice transmission. To the best of our knowledge, this is the first experimental investigation regarding tuning of MAC layer EDCA parameters in a real 802.11e WLAN network from the perspective of end-to-end voice transmission quality and end user satisfaction.

1. Introduction

Real-time voice transmission over wireless LAN (VoWLAN) imposes stringent requirements on transmission impairments such as end-to-end delays, jitter, and packet loss. The responsibility of meeting these requirements is shared between the various communication layers. Actions at the application layer include efficient encoding and packetization schemes, packet loss concealment (PLC) techniques, adaptive de-jitter buffering, echo cancellation, etc. On the network side, the new IEEE 802.11e protocol supports voice traffic by differentiating channel access probability among different traffic categories. In particular, the new, extended channel access mechanism (EDCA) allows for adjustment of a number of channel access parameters at the L2/MAC layer to prioritize VoIP packets over other traffic types. Application-layer adaptation mechanisms and MAC-layer parameters tuning can greatly mitigate the effect of transmission impairments and thus improve speech transmission quality. However, these mechanisms are often complex and difficult to tune properly. We claim that if a part of the VoIP transmission path is being tuned, the impact of local tuning actions on the whole end-to-end (mouth-to-ear) transmission has to be taken into account. For this reason we have developed a method for evaluating end-to-end VoIP transmission quality from time varying transmission impairments. This method has shown to be particularly effective in evaluating various playout buffer algorithms [1, 2], assessing VoIP performance in Voice over WLAN systems [3, 4, 5], and was recently standardized by the ITU-T [6].

In this paper we use this method to experimentally evaluate the capability of the EDCA mechanism to support voice traffic in a mixed voice/data transmission over 802.11e WLAN. We investigate how real-time voice can be supported by tuning three EDCA parameters, namely *AIFSN*, *CWmin, TXOP* and how this impacts background data transmission.

This paper is structured as follows. Section 2 briefly introduces the new method for predicting VoIP transmission quality from transmission impairments. In Section 3, the 802.11e WLAN experimental setup is described, the EDCA mechanism is outlined and proper de-jitter buffering at the application layer is addressed. Experimental results for three EDCA parameters (*AIFSN*, *CWmin, TXOP*) are presented and discussed in Section 4. Finally, the paper is concluded in Section 5.

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2. Predicting voice transmission quality from time-varying transmission impairments

The latest appendix to the ITU-T E-model [6] introduces so-called quality contours (or contours of user satisfaction) that can be used to predict voice transmission quality from time-varying transmission impairments. The quality contours determine transmission quality (indicated by the R-factor) for all possible combinations of packet loss and moth-to-ear delay. High values of R in a range of R>90 should be interpreted as excellent quality; while lower values indicate a lower quality. Values below 50 are clearly unacceptable. Based on the R rating, ITU-T Rec. G.109 [7] also introduced categories of speech transmission quality and categories of user satisfaction. Table I defines these categories in terms of R.

Table 1. Definition of categories of user satisfaction [7]

R	Speech transmission quality	User satisfaction
90-93.2	Best	very satisfied
80-90	High	satisfied
70-80	Medium	some users dissatisfied
60-70	Low	many users dissatisfied
50-60	Poor	nearly all users dissatisfied
0-50		not recommended

Figure 1 shows an example of quality contours indicating speech transmission quality and user satisfaction for the G.711 encoding scheme (bursty packet loss) with Packet Loss Concealment (PLC) implemented.

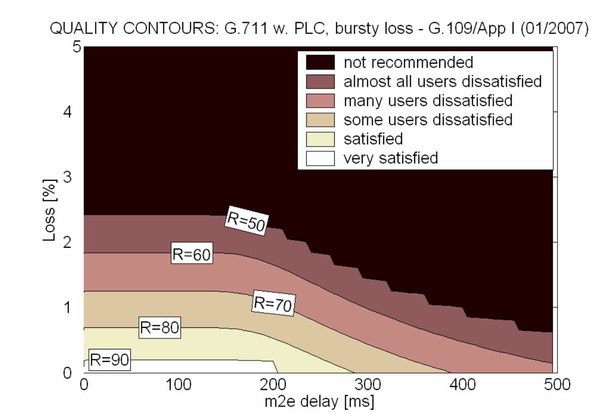

Figure 1. Quality contours for conversational speech (G.711 w. PLC and bursty loss)

The procedure of predicting speech transmission quality from transmission impairments is as follows: 1) playout delays (i.e. mouth-to-ear delays) and packet loss are calculated over non overlapping time windows of 10 seconds at the output of the de-jitter buffer; 2) quality contours are chosen for a specific encoding scheme; 3) playout delays and packet losses are mapped onto chosen quality contours; 4) overall user satisfaction regarding speech transmission quality (in the form of pie chart or average R) is derived from the distribution of playout delays and packet losses on quality contours as shown on Figure 2.

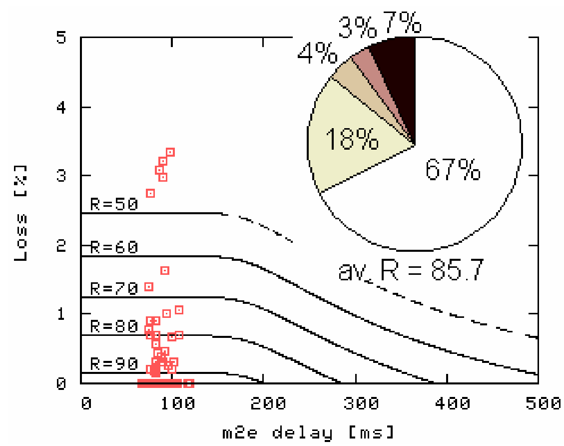

Figure 2. Predicting user satisfaction from time varying transmission impairments user satisfaction

With quality contours, the impact of delay and packet loss on conversational speech quality can be studied in two ways: either as the combined effect of loss and delay on overall quality, or as individual contributions of packet loss to speech degradation and playout delay to interactivity degradation. This is especially useful in the process of parameter tuning where a trade-off exists between packet delays and loss, and efforts are focused on finding the operating point where conversational quality is optimized.

3. 802.11e WLAN experiments

3.1. Experimental testbed

The 802.11e wireless/wired test bed consists of 15 desktop PCs acting as wireless VoIP terminals, one desktop PC acting as a background traffic generator, and one desktop PC acting as an access point (AP). All machines in the test bed use 802.11 PCMCIA wireless cards based on Atheros chipsets controlled by

MadWiFi wireless drivers and Linux OS (kernel 2.6.9). The MadWiFi drivers (Release 0.9.1 and above) provide a working implementation of the IEEE 802.11e EDCA mechanism [8]. All of the PCs nodes are also equipped with 100Mbps Ethernet cards. The PC that acts as the access point routes traffic between the wired network and the wireless clients, and vice versa (each PC has two interfaces: one on the wireless and one on the wired subnet). During the experiments each VoIP terminal runs one VoIP session and all sessions are bi-directional. In this way each terminal acts as both the source of an uplink flow and the sink of a downlink for a VoIP session. The wired interface is used to generate background traffic which is routed via the AP to the wireless interface of the same PC.

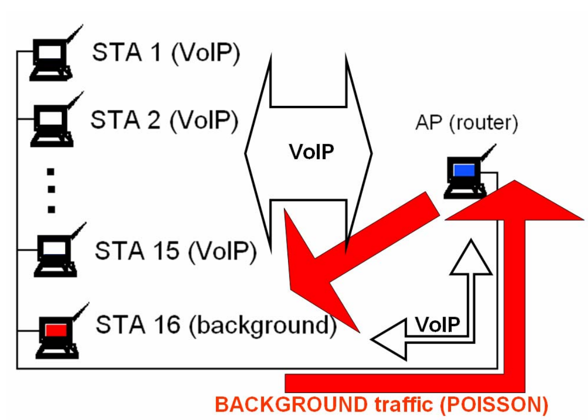

Figure 3. Experimental 802.11b testbed

All generated traffic involved both wired and wireless interfaces so that no traffic was generated between wireless interfaces. The wireless stations were located within 5 meters range of the AP to ensure that the wireless link quality is good. This test bed is illustrated in Figure 3. Voice traffic was generated using RTPtools [9] which generated G.711 encoded voice packets (80bytes audio frames created every 10ms) with fixed IP packet overhead of 12bytes for RTP, 8bytes for UDP, and 20bytes for IP layer. During the experiments bi-directional transmission of packets was realized in the form of alternating active and passive periods modeled as a four state Markov chain (talker A active, talker B active, both active, both silent). The duration of states and the transitions between them followed the ITU-T recommendation P.59. [10]. This resulted in an ON-OFF modulated CBR traffic stream being generated. Background traffic in the form of Poisson distributed UDP packet flow was generated using MGEN traffic generator [11]. For the experiments we used 1, 2, and 4Mbps background traffic. To measure effective throughput (i.e. goodput) of the background traffic we used the TRPR package [12]. The size and sending rate of the IP packets comprising the load is specified in Table 2.

Table 2. Size and sending rate of the packets comprising the background load

IP packet size [Bytes]	1Mbps load [pps]	2Mbps load [pps]	4Mbps load [pps]
256	488	977	1954
512	244	488	977
1024	122	244	488
1500	83	167	336

The reasoning behind choosing UDP and not TCP as a transport protocol for carrying background traffic is threefold: 1) UDP background traffic gives more accurate estimate of the actual load in the network (no retransmissions at transport layer); 2) results obtained with UDP constitute an upper bound for the throughput possible with TCP; 3) retransmissions of lost or corrupted packets is performed by the 802.11 MAC-layer so TCP do not get affected by the packet loss [13].

During experiments all the measured VoIP data (packet arrival times, timestamps, sequence numbers, and marker bits) was collected at all the receiving terminals to be processed later (off-line) by a program that simulated the behavior of the de-jittering buffer. Finally, the quality assessment algorithm described in Section II was used to predict the R-rating for the simulated speech.

3.2. MAC-layer parameters tuning

The original 802.11 standard does not support any type of service differentiation needed by real-time applications such as VoIP. To address this problem, the newer IEEE 802.11e standard offers two modes of MAC operation: contention-based channel access called Enhanced Distribution Coordinate Access (EDCA) and contention-free channel access called Hybrid Controlled Channel Access (HCCA). In our experiments we have focused on the performance of the EDCA mode that differentiates the channel access probability among different traffic categories. When this operational mode is used, packets are classified according to different traffic categories (TCs) at the network layer, and are mapped to four prioritized output queues (voice, video, best effort, background) at the MAC layer, called access categories (ACs). Each

AC uses a set of parameters that controls the access probability to the wireless medium:

- *AIFSN* controls the idle time (i.e. the arbitration interframe space, AIFS) after which a transmission may occur;
- *CWmin* and *CWmax* define the range of the contention window (CW) values from which the back-off time is randomly selected;
- *TXOP* controls the time interval for which a station holds the channel allowing for multiple packet transmission on a single channel access opportunity.

Configuring the EDCA parameters for each AC separately introduces access probability differentiation between TCs. Since a station with a packet to send must wait until the medium is idle and then wait for an additional period of time AIFS, the *AIFSN* parameter for the voice AC_VO ($AIFSN_{[AC\_VO]}$) should be smaller than the *AIFSN* parameter for the background AC_BK ($AIFSN_{[AC\_BK]}$). In this way time-sensitive voice traffic will contend sooner for accessing the wireless medium and thus will win on average more transmission opportunities over the less time-sensitive background traffic. After the *AIFS* period, the stations with a packet to send select random numbers between the *CWmin* and *CWmax* for each contending access category. Since the smallest number indicates "the winner", the values of *CWmin* and *CWmax* should be lower for the voice queue than for the background queue. In general, the combination of *AIFS, CWmin* and *CWmax* should be configured so that high-priority voice packets win more transmission opportunities over background traffic. However, to avoid situations in which the low-priority traffic is completely blocked, the sum of *AIFS* plus *CWmax* for high-priority voice should be greater than *AIFS* plus *CWmin* for low-priority traffic. In our experiments the voice packets were mapped into the voice queue (AC_VO) while the data traffic was mapped into the background queue (AC_BK) based on their TOS values specified in their IP headers.

During the first experiment we prioritized voice over background traffic by increasing the number of time slots comprising the background *AIFS* period ($AIFSN_{[AC\_BK]}$) from 2 to 15. The other AC_BK parameters were: *CWmin*=7, *CWmax*=1023, *TXOP*=0 and they were kept fixed for the duration of the first experiment. During the second experiment we prioritized voice over data traffic by increasing the $CWmin_{[AC\_BK]}$ parameter from 7 to 1023. The other AC_BK parameters were: *AIFSN*=2, *CWmax*=1023, *TXOP*=0 and they were kept fixed for the duration of the second experiment. Finally, during the third experiment we prioritized voice over data traffic by increasing the $TXOP_{[AC\_VO]}$ parameter from 0 to 8192 μs while keeping the $TXOP_{[AC\_BK]}$ disabled. The other AC_BK and AC_VO parameters were: *AIFSN*=2, *CWmin=7, CWmax*=1023 and they were kept fixed for the duration of the third experiment.

The parameters under consideration for both AC_BK and AC_BK are listed in Table 3

Table 3. EDCA parameters settings during the experiments

EDCA parameter	AC_VO class (STAs and AP)	AC_BK class (STAs and AP)
CWmin	7	7,15,31,63,127,511,1023
CWmax	1023	1023
AIFSN	2	2,3,4, …13,14,15
TXOP	0, 512, 1024, 2016, 4000, 8192 μs	0

3.3. Application-layer parameters tuning

Impairments introduced by de-jitter buffering at the receiver can be more substantial than the transmission impairments introduced by the network. This can be often observed in a WLAN environment where the delay variation is high due to contention-based access mechanisms causing congestion at the AP. Good de-jittering schemes can mitigate the effects of high jitter by minimizing buffering delays and minimizing number of discarded packets due to their late arrival. Consequently, we claim that proper tuning of the de-jitter mechanism is essential for ensuring acceptable quality speech

In our experiments we used Ramjee's algorithm [14] which is often used as a reference playout buffer controller. The algorithm uses the same playout delay throughout a given talkspurt but permits different playout delays for different talkspurts. We modified the original Ramjee algorithm by adding one parameter, namely *playout_offset* that represents additional pre-buffering delay. In our solution the playout time p_i at which the the *i-th* packet, assumed to be the first packet in a talkspurt (played at the destination) is calculated as follow:

$$p_i = t_i + \hat{d}_i + \beta \cdot \hat{v}_i + playout\_offset \qquad (1)$$

where $\hat{d}_i$ and $\hat{v}_i$ are the estimates of delay i-th packet delay n_i and its variance respectively and are calculated as follows:

$$\hat{d}_i = \alpha \cdot \hat{d}_{i-1} + (1-\alpha) \cdot n_i \tag{2}$$

$$\hat{v}_i = \alpha \cdot \hat{v}_{i-1} + (1-\alpha) \cdot | \hat{d}_i - n_i | \tag{3}$$

Parameter β controls the delay/packet loss ratio while parameter α controls the ability of the algorithm to follow the changes in the delay. By experimenting with different values of α, β, and *playout_offset* in a real wireless environment we were able to chose the values (i.e. α = 0.998002, β = 2, *playout_offset* = 40ms) that maximized rating factor R for all possible *AIFSN* and *CWmin* settings.

4. Experimental results

4.1. Tuning the AIFSN[AC_BK] parameter

Firstly, we experimentally investigated the impact of the *AIFSN* parameter on the access probability differentiation between AC_VO and AC_BK in a mixed voice/data wireless transmission. Experiments covered 3 background traffic loads (1, 2, and 4Mbps), 4 packetization schemes for background (256Bytes, 512Byte, 1024Byte and 1500Byte packets) and 14 settings of the $AIFSN_{[AC\_BK]}$ parameter: 2, 3 …14, and 15.

Figure 4 shows the average voice transmission quality (at wireless and the wired interfaces) in terms of the R-rating factor calculated for all 15 VoIP terminals and the effective throughput (i.e. goodput) as a function of $AIFSN_{[AC\_BK]}$ for three background traffic loads of a) 1Mbps, b) 2Mbps, and c) 4 Mbps.

It can be seen that voice transmission at the wireless subnet can be effectively prioritized over data by tuning the $AIFSN_{[AC\_BK]}$. Increasing $AIFSN_{[AC\_BK]}$ essentially promotes the AC_VO queue at the expense of the AC_BK queue in terms of probability access. The bigger the difference in *AIFSN* values, the easier it is for the AC_VO queue to win transmission opportunities from the AC_BK queue. As a result, transmission impairments (delay, jitter and packet loss) are reduced and the overall transmission quality is improved. For example, when the *AIFSN* difference between AC_VO and AC_BK was 6 ($AIFSN_{[AC\_BK]}$=8 and $AIFSN_{[AC\_VO]}$=2), all VoIP stations could experience at least “toll” voice transmission quality (indicated by $R \geq 70$) for all examined background traffic loads and packetization schemes. Conversely a substantial reduction in the background traffic goodput was observed. In some cases (i.e. the 256 Bytes background packets load) the goodput of the background traffic was almost halved. Increasing the *AIFSN* difference between AC_BK and AC_VO further penalizes background traffic by making it more difficult to win transmission opportunities.

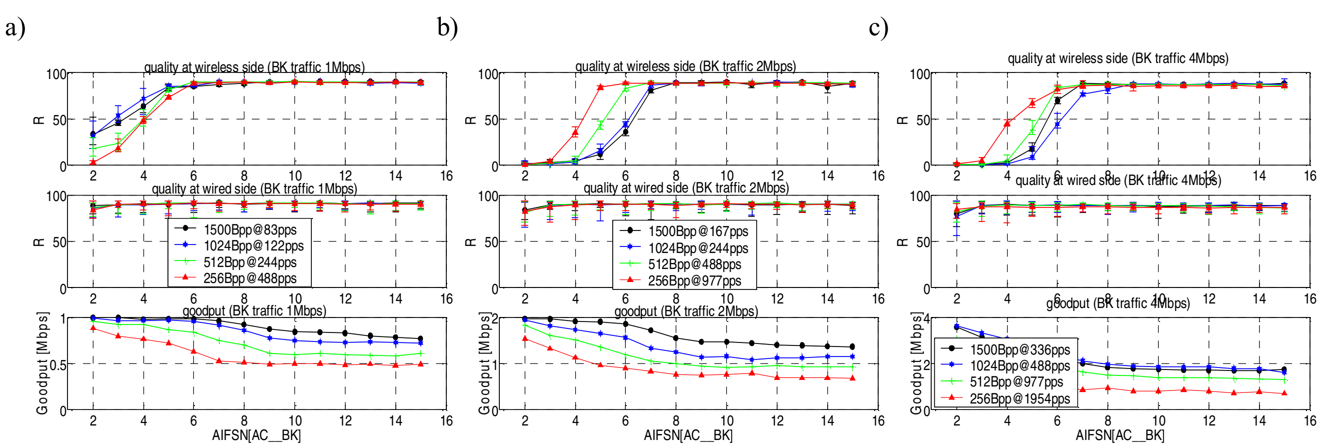

Figure 4. Quality of voice transmission vs *AIFSN[AC_BK]* (wired and wireless side) and effective throughput of the a) 1Mbps, b) 2Mbps, and c) 4Mbps background traffic.

4.2. Tuning the CWmin[AC_BK] parameter

A second set of experiments was conducted to experimentally investigate the impact of the *CWmin* parameter on a mixed voice/data wireless transmission. Similar to the first set of experiments we considered 3 background traffic loads and 4 packetization schemes.

situations (see the 1500Byte packet curve on Figure 7), when changes in the *CWmin* parameter have limited effects on throughput differentiation [15]. A substantial reduction in the background traffic throughput can be observed when higher background traffic loads of 4Mbps are injected to the network (see Figure 5c).

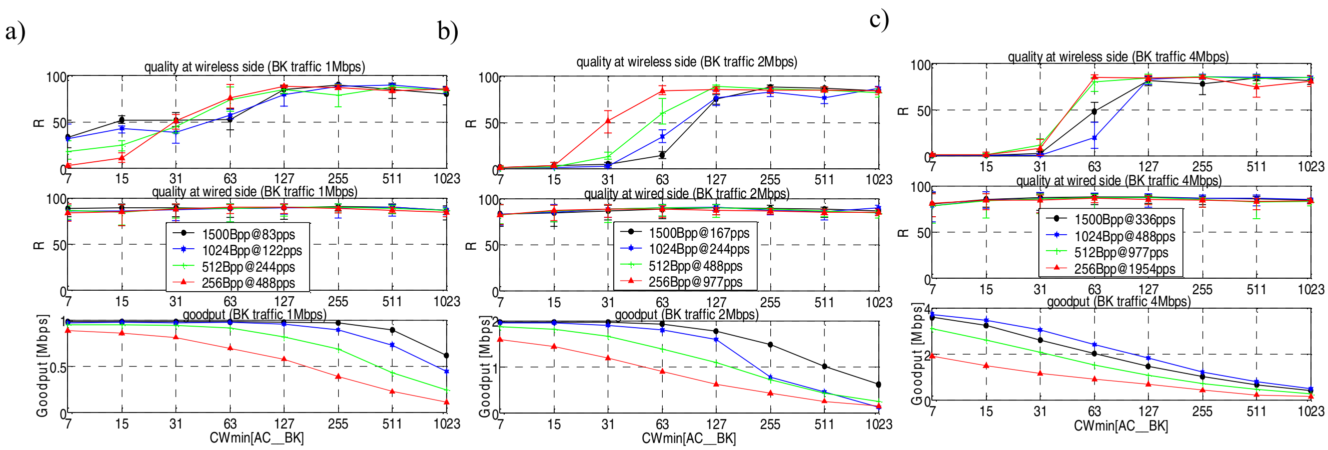

Figure 5. Quality of voice transmission vs *CWmin[AC_BK]* (wired and wireless side) and effective throughput of the a) 1Mbps, b) 2Mbps, and c) 4Mbps background traffic.

This time we examined 8 settings of the $CWmin_{[AC\_BK]}$ parameter: 7, 15, 31, 63, 127, 255, 511, and 1023.

Figure 5 shows the average voice transmission quality (at wireless and wired interface) calculated for 15 VoIP terminals and the goodput of the background traffic as a function of $CWmin_{[AC\_BK]}$ for three background traffic loads a) 1Mbps, b) 2Mbps, and c) 4 Mbps.

This time the channel access probability differentiation was introduced by using different values of *CWmin* for the AC_VO and for AC_BK queues. Stations with lower values of *CWmin* experienced a smaller average waiting time required to win transmission opportunity (i.e. shorter back-off time), and thus could experience improved performance in comparison to the stations with higher *CWmin* values. In other words, the higher the *CWmin* value for AC_BK queue, the higher the probability of winning a transmission opportunity ahead of the AC_BK queue resulting in improved voice transmission quality.
Consequently, it can be seen from Figures 8, 9, and 10 that as $CWmin_{[AC\_BK]}$ increases, the average voice transmission quality at the wireless subnet increases as well. However, tuning the $CWmin_{[AC\_BK]}$ parameter is not as effective as tuning the $AIFSN_{[AC\_BK]}$. This can be observed especially in low network congestion

4.3. Tuning the TXOP[AC_VO] parameter

A third set of experiments was conducted to investigate the impact of the *TXOP* parameter on a mixed voice/data wireless transmission. Similarly to the first set of experiments we took into account 3 background traffic loads and 4 packetization schemes. However, this time we examined 6 settings of the $TXOP_{[AC\_VO]}$ parameter: 0, 512, 1024, 2016, 400, 8192 µs. Figure 6 shows the average voice transmission quality (at wireless and wired interface) calculated for 15 VoIP terminals and the goodput of the background traffic as a function of $TXOP_{[AC\_VO]}$ for three background traffic loads a) 1Mbps, b) 2Mbps, and c) 4 Mbps.

As can be seen from the Figure 5, the TXOP parameter has a limited influence on the quality of voice transmission. In fact, the capability of the TXOP parameter tuning to support voice transmission is limited to situations when the background traffic is low (see Figure 6a). In the situations with higher background loads (e.g. 2 and 4Mbps), the quality of voice transmission was poor ($R<50$). The TXOP parameter defines the maximum length of a single transmission and plays important role when large amount of data is to be sent (when data to be sent is too large to transfer within the TXOP limit, the station splits it into multiple transmissions.) Since voice packets are short, setting the TXOP parameter can be neglected.

background goodput performance. Increasing the $CWmin_{[AC\_BK]}$ parameter produces a greater reduction in the goodput compared to the $AIFSN_{[AC\_BK]}$.

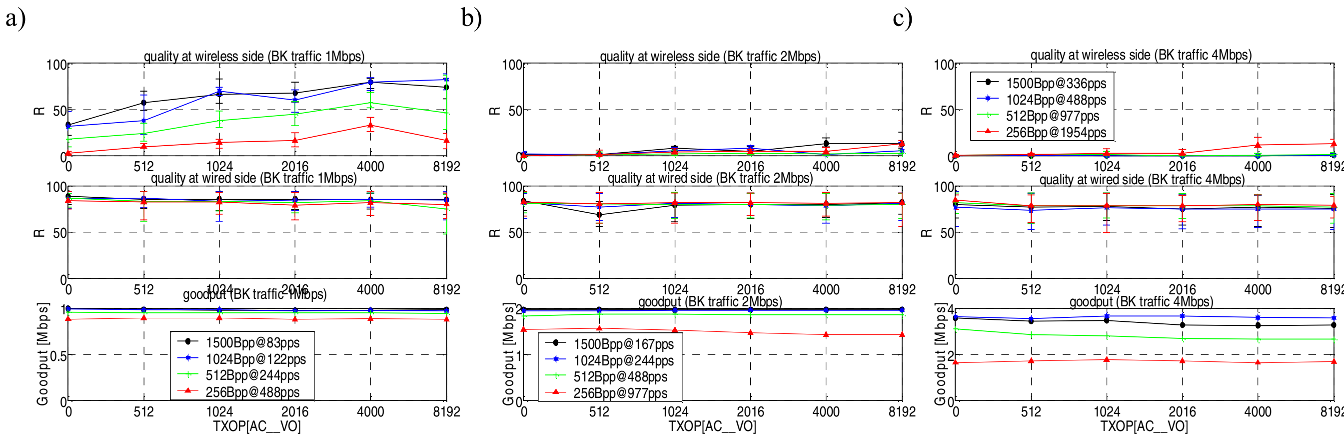

Figure 6. Quality of voice transmission vs *TXOP[AC_VO]* (wired and wireless side) and effective throughput of the a) 1Mbps, b) 2Mbps, and c) 4Mbps background traffic.

5. Conclusions

In this paper we have experimentally evaluated the capability of the new 802.11e MAC protocol to support voice calls in a mixed voice/data transmission over WLANs. In our experiments we have focused on the contention-based mode of MAC operation called Enhanced Distributed Channel Access (EDCA) and more specifically on the three quality enhancement parameters: the *AIFSN*, *CWmin* and *TXOP*.

Our results show that the proper tuning of either *AIFSN* or *CWmin* parameters can improve voice transmission quality at the wireless subnet while reducing the goodput of the background data traffic. We have also demonstrated that the quality differentiation with the *AIFSN* parameter provides superior and more robust operation than access differentiation through the *CWmin* parameter. For example, when the *AIFSN* difference between AC_BK and AC_VO was 6 ($AIFSN_{[AC\_BK]}$=8 and $AIFSN_{[AC\_VO]}$=2), all VoIP terminals could experience at least "toll" voice transmission quality (indicated by $R \geq 70$) in the presence of the heavy background traffic injected to the network. The same results ($R \geq 70$) could be obtained only for some VoIP terminals when the difference between *CWmin* for AC_BK and AC_VO was 120 ($CWmin_{[AC\_BK]}$=127 and $CWmin_{[AC\_VO]}$=7). The impact of the $AIFSN_{[AC\_BK]}$ and $CWmin_{[AC\_BK]}$ parameters is different on the

Our experimental results confirm earlier analytical and simulation-based findings that the *AIFSN* parameter more effectively protects voice calls against background data traffic than the *CWmin* [15][16][17][18]. The *AIFSN* differentiation is a superior mechanism to *CWmin* differentiation because of the existence of discrete instants of times (protected slots represented by the *AIFSN* difference) where a lower number of stations may compete and access the channel. This increases the effectiveness of the overall random access mechanism for the high-priority stations. The TXOP parameter has limited influence on the quality of voice transmission. This parameter plays an important role when large data comprising large packets sizes is to be sent. Since voice packets are short, setting the TXOP parameter can be neglected.

To our knowledge, all experimental work regarding voice transmission in real 802.11e WLAN networks was focused only on MAC layer delays introduced by the EDCA mechanism [19]. This paper is the first experimental demonstration of voice prioritization over background data transmission from the perspective of end-to-end speech transmission quality and user satisfaction.

6. Acknowledgments

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