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Experimental Tuning of AIFSN and CWmin Parameters to Prioritize Voice over Data Transmission in 802.11e WLAN Networks

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Experimental Tuning of AIFSN and CWmin Parameters to Prioritize Voice over Data Transmission in 802.11e WLAN Networks

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ABSTRACT
In this paper we experimentally study the impact of two EDCA parameters, namely *AIFSN* and *CWmin*, on a mixed voice/data wireless transmission. In particular we investigate how the tuning of these parameters affects both the voice transmission quality and background data throughput. We predict end-to-end voice transmission quality from time varying transmission impairments using the latest Appendix to the ITU-T E-model. Our experimental results show that the tuning of the EDCA parameters can be used to successfully prioritize voice transmission over data in real 802.11e networks. We also demonstrate that the *AIFSN* parameter more effectively protects voice calls against background data traffic than *CWmin*. To the best of our knowledge, this is the first experimental investigation on tuning of MAC layer parameters in a real 802.11e WLAN network from the perspective of end-to-end voice transmission quality and end user satisfaction.

Categories & Subject Descriptors: C.2.1
COMPUTER-COMMUNICATION NETWORKS: Network Architecture and Design, Wireless communication; Network communications; Packet-switching networks

General Terms: Design, Experimentation, Performance

Keywords: VoIP over wireless LAN (VoWLAN), IEEE 802.11e EDCA, differentiated prioritization scheme, QoS

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 two EDCA parameters, namely *AIFSN* and *CWmin* 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, EDCA mechanism is outlined and proper de-jitter buffering at application layer is addressed. Experimental results for both EDCA parameters (*AIFSN* and *CWmin*) 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 speech transmission quality and 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.

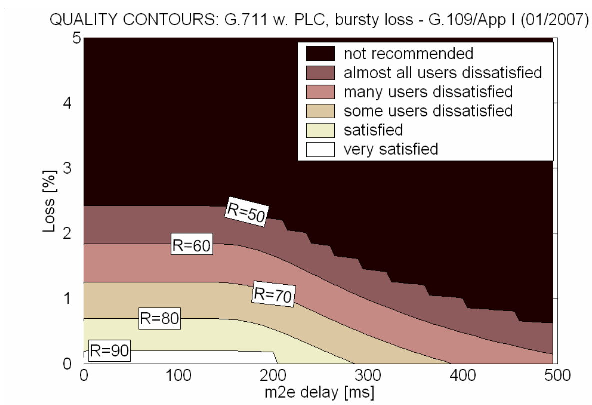

Figure 1. Quality contours for G.711 encoding scheme (bursty packet loss + PLC) [6]

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.

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 when a trade-off exists between packet delays and loss, and efforts are focused on finding the operating point where conversational quality is maximized

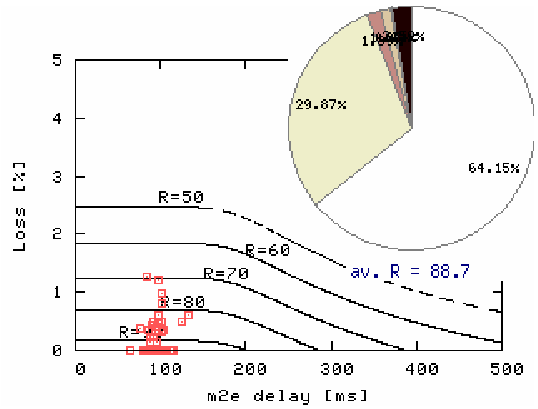

Figure 2. Predicting user satisfaction from time varying transmission impairments.

3. 802.11e WLAN EXPERIMENT

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 working implementation of IEEE 802.11e EDCA mechanism [8]. All of the nodes are also equipped with 100Mbps Ethernet cards. The PC that acts as 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 wired network). During the experiments each VoIP terminal runs one VoIP session and all sessions are bi-directional. In this way each terminal acts as the source of an uplink flow and the sink of a downlink for a VoIP session. The wired interface of the background traffic generator is used to generate background traffic which is routed via the AP to the wireless interface of the same PC (see Figure 3).

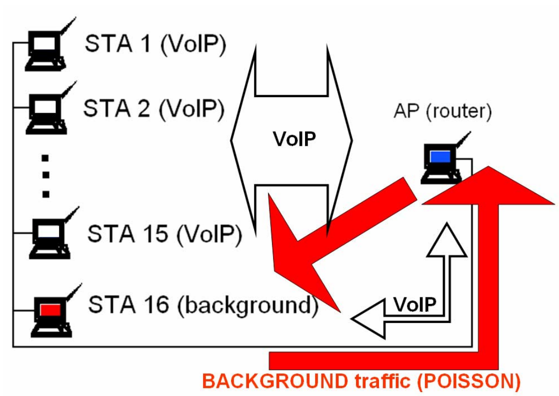

Figure 3. Experimental 802.11e test bed.

All generated traffic involved a wired and a wireless interface so that no traffic was generated between wireless interfaces. The wireless stations were located within 5 meters range from the AP to ensure that the wireless link quality is good. This test bed is illustrated in Fig. 5. 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 (goodput) of the background traffic we used TRPR package [12]. The size and sending rate of the IP packets comprising the load is specified in Table 2.

Table 2. The size and sending rate of the packets comprising the background load [7].

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 done by the 802.11 MAC-layer so TCP do net 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 receiving terminals to be processed later (off-line) by a program that simulated the behavior of the de-jittering buffer and finally by the quality assessment algorithm described in Section 2.

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. The newer standard called 802.11e offers two quality enhancement mechanisms: contention-based channel access mechanism called Enhanced Distribution Coordinate Access (EDCA) and contention-free channel access mechanism called Hybrid Controlled Channel Access (HCCA). When the Enhanced Distributed Channel Access (EDCA) mechanism is used, packets are categorized in different traffic categories (TCs), and later mapped to four prioritized output queues called access categories (ACs). Each AC uses its own set of channel access parameters that control access to the wireless medium. Those parameters are: Arbitration Inter-Frame Space (*AIFS*), minimum and maximum Contention Window (*CWmin* and *CWmax*), and the maximum length of a single transmission (*TXOP*).

Configuring these parameters for each AC separately enables service differentiation between TCs as follows: A station with packet to send waits until the medium is idle and for an additional period of time defined by the *AIFSN* parameter. *AIFS* period for voice AC should be smaller than *AIFS* for background AC. This way time-sensitive voice traffic will content sooner for access to the wireless medium winning transmission opportunities over less-sensitive background traffic. After the *AIFS* period, the stations with a packet to send generate random numbers between the *CWmin* and *CWmax* for each contending access category. Since the smallest number indicates the winner, the value of *CWmin* and *CWmax* should be lower for voice AC than for background AC. In general the combination of *AIFS, CWmin* and *CWmax* should be configured so that high-priority voice packets win transmission opportunities over background traffic. However, to avoid situations in which the low-priority traffic is completely blocked, the sum of *AIFSN* plus *CWmax* for high-priority voice should be greater than *AIFSN* for low-priority traffic. 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.

In our experiments the voice packets were mapped into the voice AC (AC_VO) queue while the data traffic was mapped into the background (AC_BK) queue based on their TOS values in IP packets' 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. All 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 $CWmin_{[AC\_BK]}$ from 7 to 1023. All the other AC_BK parameters were: *AIFSN*=2, *CWmax*=1023, *TXOP*=0 and they were kept fixed for the duration of the second experiment. The parameters under consideration for both AC_BK and AC_BK are listed in Table 3..

Table 3. EDCA parmeters 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	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 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. In our experiments we used the 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's algorithm by adding one parameter, namely *playout_offset* that represents additional pre-buffering delay. In our solution the playout time *pi* 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:

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

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

Parameter β (discrete values: 0, 0.5…5) controls the delay/packet loss ratio while parameter α (continuous values: 0…0.998002) controls the agility of the estimation process. 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. $\alpha = 0.998002$, $\beta = 2$, *playout_offset* = 40ms) that maximized rating factor *R* for all possible *AIFSN* and *CWmin* settings.

4. Experimental Results

4.1 Tuning the *AIFSN* 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, 4Mbps), 4 packetization schemes for background (256Bytes, 512Bytes, 1024Bytes and 1500Bytes packets) and 14 settings of the $AIFSN_{[AC\_BK]}$ parameter: 2, 3 …14, and 15.

Figures 4, 5 and 6 shows the average voice transmission quality (at wireless and wired interface) calculated for all 15 VoIP terminals and effective throughput (i.e. goodput) as a function of $AIFSN_{[AC\_BK]}$ for three background traffic loads (1Mbps, 2Mbps, and 4 Mbps respectively). 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 AC_BK. 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_BK and AC_VO 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

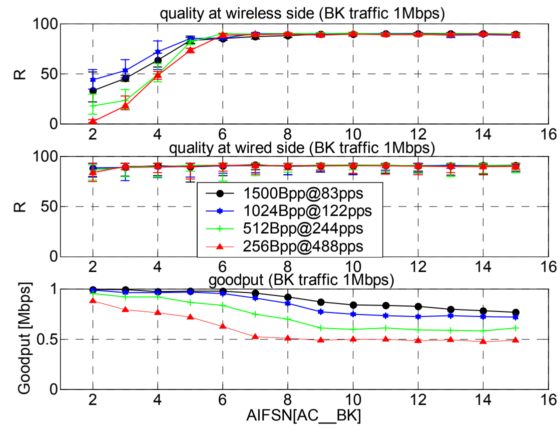

Figure 4. Quality of voice transmission and effective throughput of 1Mbps background traffic vs $AIFSN_{[AC\_BK]}$.

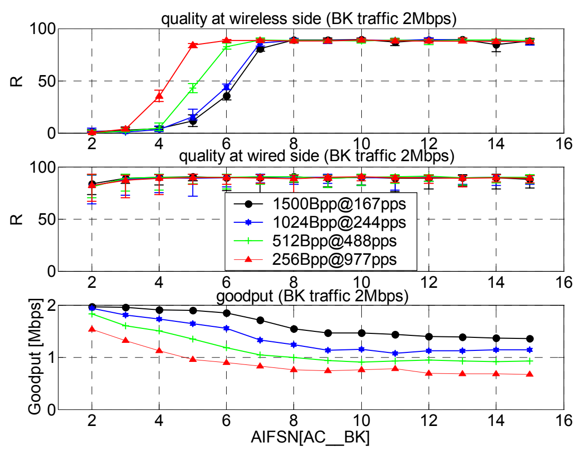

Figure 5. Quality of voice transmission and effective throughput of 2Mbps background traffic vs $AIFSN_{[AC\_BK]}$.

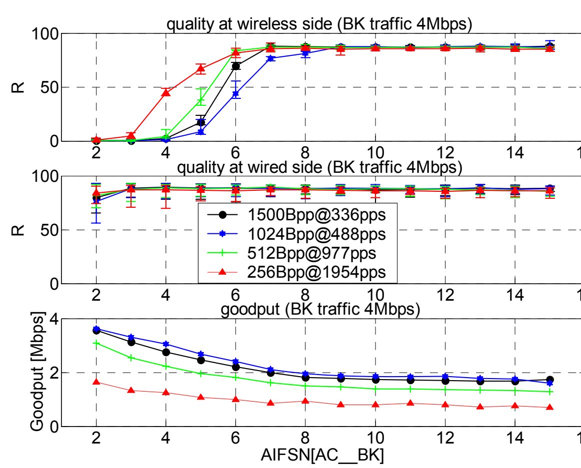

Figure 6. Quality of voice transmission and effective throughput of 4Mbps background traffic vs $AIFSN_{[AC\_BK]}$.

At the same time (when the *AIFSN* difference between AC_BK and AC_VO was 6) a substantial reduction in the background traffic goodput was observed. In some cases (256Bytes packets comprising the background load) the goodput of the background traffic was almost halved. Increasing the *AIFSN* difference between AC_BK and AC_VO penalizes background traffic by making it more difficult to win transmission opportunities.

4.2 Tuning the *CWmin* parameter

A second set of experiments was conducted to experimentally investigate the impact of the *CWmin* 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 8 settings of the $CWmin_{[AC\_BK]}$ parameter: 7, 15, 31, 63, 127, 255, 511, and 1023. Figures 7, 8 and 9 show 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 (1Mbps, 2Mbps, and 4 Mbps respectively). This time the channel access probability differentiation was provided by using different values of *CWmin for* AC_VO and for AC_BK. Stations with lower value of *CWmin* experienced smaller average time needed to win transmission opportunity (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* for AC_BK, the higher probability of winning the contention by the AC_VO what resulted in improved voice transmission quality. Consequently, it can be seen from Figures 7, 8, and 9 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 situations (see 1500Bytes 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 9).

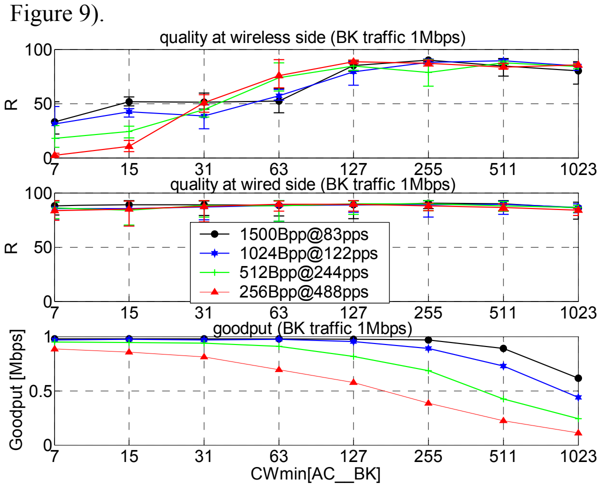

Figure 7. Quality of voice transmission and effective throughput of 1Mbps background traffic vs $CWmin_{[AC\_BK]}$.

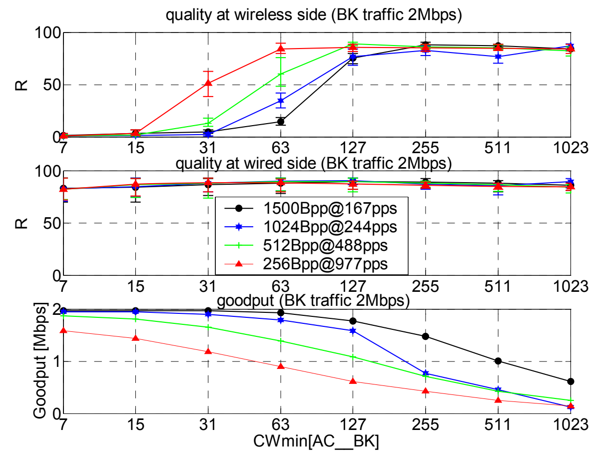

Figure 8. Quality of voice transmission and effective throughput of 2Mbps background traffic vs $CWmin_{[AC\_BK]}$.

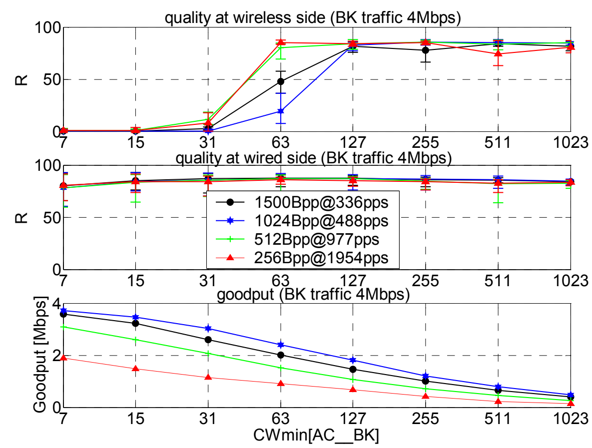

Figure 9. Quality of voice transmission and effective throughput of 4Mbps background traffic vs $CWmin_{[AC\_BK]}$.

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 two quality enhancement mechanisms: the usage of different arbitration interframe spaces (controlled by the *AIFSN* parameter) and the usage of different minimum contention windows (controlled by the *CWmin* parameter).

Our results show that the proper tuning of either *AIFSN* or *CWmin* parameters can improve voice transmission quality at the wireless subnet reducing 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 contention window differentiation with 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 for 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). A substantial reduction in the background traffic throughput was also observed as a result of increasing either the $AIFSN_{[AC\_BK]}$ *or* $CWmin_{[AC\_BK]}$ parameters. However, increasing the $CWmin_{[AC\_BK]}$ resulted with unnecessary higher reduction of the background goodput than increasing the $AIFSN_{[AC\_BK]}$. Our experimental results confirm earlier analytical and simulation-based findings that the *AIFSN* parameter more effectively protects voice calls against data than the *CWmin* [15][16][17][18]. The *AIFSN* differentiation is a superior mechanism to *CWmin* differentiation because of the very 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 mechanism for the high-priority stations.

To our knowledge, all experimental work regarding voice transmission quality 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 voice transmission quality and user satisfaction

6. ACKNOWLEDGMENTS
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