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Perceptual Quality of Internet Telephony over IEEE 802.11e Supporting Enhanced DCF and Contention Free Bursting

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Abstract

As the IEEE 802.11 based internet access is becoming ubiquitos available it is excepted that WLAN will be applied for telephony. In this paper we follow the objective to access whether WLAN allows calls at toll quality. Toll quality is the minimal quality of PSTN-based telephons calls. With simulations we evaluate the MAC protocol modes distribution coordination function (DCF), enhanced DCF (EDCF) and contention free bursting (CFB). We have verifed our simulation model by cross-checking it with the results of other researchers. We present quantitative results of the perceptual quality and enhance the precision of performance evaluation on Voice over Wifi systems.

Contents

Chapter 1

Introduction

The IEEE 802.11 standard supports two MAC mechanisms, the Distributed Coordination Function (DCF) and the Point Coordination Function (PCF). These mechanisms are considered to be insufficient for achieving a reasonable quality [15] in scenarios with high background load. Thus, QoS enhancements are vividly studied and evaluated. Currently, the QoS enhanced MAC protocol IEEE 802.11e is under design and in the standardization process [26]. IEEE 802.11e introduces two additional MAC modes: the Enhanced Distributed Coordination Function (EDCF) and the Hybrid Coordination Function (HCF).

In this paper, we present an open-source, verified simulation model of IEEE 802.11e's EDCF mode for the network simulator (ns-2). We verified our model by comparing it with previous published results [18]. Our 802.11e EDCF model includes contention free bursting (CFB=TXOP bursting), which allows the transmission of a train of small packets without intermediate contention.

We apply our simulation model to evaluate the quality of telephone calls using EDCF model. To measure the quality we develop a performance tool model that combines the ITU E-model, the ITU PESQ algorithm, and various implementation of playout schedulers. Our performance tool outperforms previous approach because it considered also the impact of playout rescheduling in addition to transmission delay and speech quality. Thus, we are able to evaluate in high precision to what extend various MAC protocol modes can be applied for telephony.

Our simulation scenario is a Basic Service Set (BSS), which consists of a base station and multiple wireless nodes. Bidirectional streams of Voice over IP (VoIP) packets model telephone calls. Our simulations solves the following issues:

First, we are interested in the effect of best-effort background traffic on voice transmissions. Parallel to a voice call, we transmit UDP or TCP flows with the maximum possible rate. TCP harms the quality of the voice streams less than UDP, because it slows down its sending rate. In the DCF mode, UDP displaces any voice stream that has the same direction. EDCF and EDCF+CFB enable parallel data and voice traffic.

Secondly, we measure the throughput of TCP and UDP with DCF and EDCF without any voice transmission to get a statement about the efficiency of EDCF. With EDCF the throughput of the background traffic is lower because of the lower priority.

Last, we consider how many simultaneous telephone calls are supported at which quality level. We vary the MAC mechanisms (DCF, EDCF and EDCF+CFB) and measure the perceptual telephone quality with ITU's E-Model [3] and PESQ [4] for uplink and downlink. On the uplink direction, the basic DCF can transmit the highest number of telephone calls, followed by EDCF and EDCF+CFB. On the downlink instead, DCF performs worst after EDCF and EDCF+CFB.

This paper is structured as follows: First, we explain the technical background of our work, namely the MAC mechanisms and the perceptual evaluation of speech- and telephone quality. Then, we discuss the related work on voice over WLAN. The fourth chapter describes the simulation model, its verification, and the simulation scenarios. Next, we present our simulation results, which are finally concluded.

Chapter 2

Technical Background

2.1 IEEE 802.11 MAC

Two medium access functions are defined in 802.11: the Point Coordination Function (PCF) and the Distributed Coordination Function (DCF). While the DCF is responsible for asynchronous data services, the PCF offers time-bounded services. The PCF is used in the contention-free period (CFP), while the DCF handles the contention period (CP). One CFP and one CP are combined to a superframe. Superframes are separated by periodic management frames, the so-called Beacon frames. 802.11 uses the three different inter-packet gaps SIFS, PIFS and DIFS, denoted as interframe spaces, to control the medium access, i.e. to give stations in specific cases a higher or lower priority (see figure 2.1).

2.1.1 DCF

In the DCF, sending stations stand in contention to each other. Due to CSMA/CA (carrier sense multiple access with collision avoidance), a station has to sense the channel before being allowed to send a frame. The station may send if the medium has been idle for at least DIFS. In the case of a busy medium the station starts a random backoff procedure, the binary exponential backoff. It determines a random *backoff time=slotTime*Random*, where *Random* is a pseudorandom integer value out of the uniformly distributed contention window [0, CW] with CW_{min}≤CW≤255. Initially CW=CW_{min} is set to 7 in 802.11. If the medium is idle again at least for DIFS, the station decrements the backoff time until the medium gets busy. The station is allowed to send immediately if the random backoff time is equal to zero.

The random backoff procedure has to be started after every transmission. In the case of a successful acknowledged transmission the procedure will be started after the received ACK (denoted as postbackoff). Otherwise the procedure will be started after the expiration of the ACK timeout interval. In this case, a collision is assumed. A collision occurs if two (or more) stations have detected the medium as idle for DIFS, both are allowed to send and both start their transmissions immediately. To avoid repeating collisions, increasing $CW_{min}s$ have to be chosen. For the first up to the fourth retransmission the CW_{min}-value is set to $CW_{min,new}$=2*$CW_{min,old}$ + 1 (⇒ 15, 31, 63, 127 in 802.11). This algorithm is denoted as binary exponential backoff. For five or more retransmissions the CW_{min}-value has to be set to CW_{max}=255.

2.1.2 PCF

The PCF can only be used in an infrastructure-based network because it requires an access point (AP). Usually the Point Coordinator (PC) is installed on this AP. The PC manages the access to the medium in the CFP by polling stations sequentially.

The PCF comes up with a higher complexity than the DCF. It is mostly implemented in currently installed infrastructured BSS but it is not used very often due to the lack of an optimized scheduling / polling.

2.2 IEEE 802.11e MAC

For achieving QoS, 802.11e uses multiple priority queues for the prioritized and separate handling of different traffic categories (TCs). In addition, 802.11e introduces the Enhanced Distributed Coordination Function (EDCF) and the Hybrid Coordination Function (HCF). The EDCF manages the medium access in the CP while the HCF is responsible for the CFP and the CP. Both functions are described below.

We based our studies of 802.11e on the paper [18], the November 2001 draft [25] and the 2003 draft version 5 [26]. [26] introduces several changes in the nomenclature (TC in EDCF ⇒ access category (AC), EDCF ⇒ enhanced distributed channel access (EDCA), HCF ⇒ HCF controlled channel access (HCCA)).

2.2.1 EDCF

The EDCF enhances the 802.11 DCF by introducing an own backoff instance with a separate backoff parameter set for each priority queue. Each TC on a station contends for a transmission opportunity (TXOP). A TXOP is defined in [18, 26] as *"an interval of time when a station has the right to initiate transmissions, defined by a starting time and the maximum duration"*. Parameters for prioritization of TCs are the arbitration interframe space (AIFS), the minimum size of the CW ($CW_{min}[TC]$) and the TXOPLimit[TC]. The AIFS describes (similar to 802.11 DCF) the duration of time up to which the medium must be idle before a station may access the channel or decrement the backoff of the corresponding TC. To ensure a prioritized access with respect to legacy 802.11 stations, the EDCF should use smaller CW_{min}-values for high-priority data flows. The TXOPLimit[TC] specifies the maximum duration of one TXOP.

The determination scheme of the backoff time for each TC is in the current draft [26] equal to the legacy DCF binary exponential backoff procedure. In earlier versions, the CW was enlarged after an unsuccessful transmission to

$$newCW[TC] = [(oldCW[TC] + 1) * PF] - 1. \tag{2.1}$$

The persistence factor (PF) ϵ [1,16] was another prioritization parameter [25], which is not used anymore in [26] (PF=2 leads to a binary exponential backoff).

Different TCs on one station with their own set of parameters (AIFS, CW, PF) and their own backoff instance are shown in figure 2.2. Up to eight TC queues could be set up per station in the older draft [25]. Due to the mapping to 4 ACs, [26] uses only four queues. The structure of several queues may lead to a collision if two TCs are allowed to send at the same

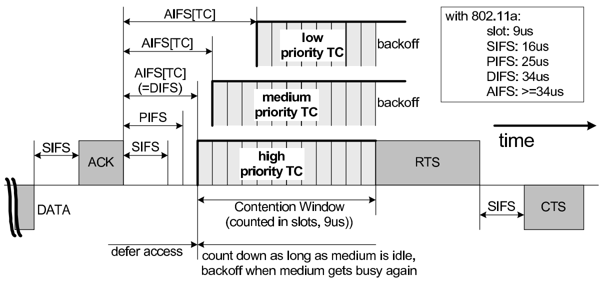

Figure 2.1: IFS relationships in 802.11e (source [18])

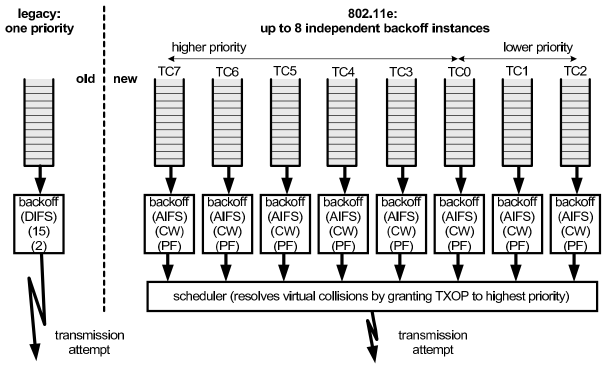

Figure 2.2: IEEE802.11e Mac structure (source [18])

time. This collision is solved by a virtual scheduler which grants access to the TC with the highest priority and starts a backoff for the lower TC (after increasing $CW_{min}[TC]$).

2.2.2 HCF

The HCF controls both the CFP and CP. It uses a polling scheme to control the medium access.

To grant and to administrate polled-TXOP requests a kind of scheduling management, which introduces a lot of complexity, is needed at the AP (denoted as Hybrid Coordinator (HC)). The HCF is still a moving target in the IEEE draft. It is very difficult to model and to verify without an exact definition.

Generally, the HCF as well as the PCF in 802.11 requires an infrastructured BSS. For ad hoc networks an additional MAC mode beside HCF/PCF is required (e.g EDCF). On the other hand, if EDCF works well in both ad hoc and infrastructured networks, HCF is not needed. In this paper, we will show that the vast majority of VoIP traffic scenarios do not require HCF.

2.3 Contention Free Bursting

Tourrilhes [22] proposed the idea of Contention Free Bursting (CFB) to improve the performance for small packets (of timebounded services) in Wireless LANs. CFB decreases the overhead and delay and increases the throughput. CFB sends multiple small packets as a burst without intermediate contention as soon as the station gains access to the medium (see figure 2.3).

It is possible to send packets to different destinations in one burst frame. Between an ACK and the following packet only a time interval of SIFS is required. Therefore the station keeps control over the medium for the whole burst. Sending multiple small packets in a burst avoids contention for each single packet and increases the efficiency. However, the medium access time might be increased because packet bursts occupy the medium for a longer period.

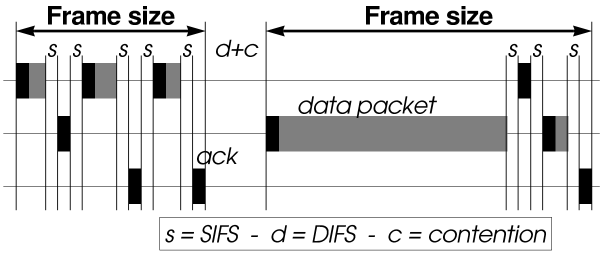

Figure 2.3: principle of PFG (source: [22])

2.4 VoIP Quality

Interactive voice transmissions have stringent requirement on packet loss, delay and jitter. The main quality criteria is the perceptual quality because telephone calls are used usually between humans. The network centric metric loss and delay are less precise.

To predict the human quality rating with precise measurable parameters, psycho-acoustic quality models have been developed. In this paper we apply models that evaluate the perceptual telephone and speech quality, namely ITU's E-Model and PESQ. Those quality models show a high correlation with subjective tests and take into account speech quality and delay to estimate the performance of VoIP transmissions.

To predict the subjective speech quality (which could only be derived by cost-intensive experiments with many people) with an objective metric, we applied the PESQ algorithm [4]. PESQ compares an original speech sample with its degraded, transmitted version to estimate the Mean Opinion Score (MOS). The MOS value ranges from 1 (bad) to 5 (excellent speech quality).

The quality of a telephone call is not described by speech quality entirely. Instead, other factors like delay play an important role, too. We estimated the quality of a telephony call with the ITU E-Model [3]. The E-Model's primary output is the R-Factor, which ranges from 0=bad over 70=toll-quality to 100=excellent. To calculate the R-Factor, the E-Model takes into account the speech quality, the mouth-to-ear delay and other factors that influence the quality (e.g. echoes and loudness).

2.5 Related Work

One primary design goal of the IEEE 802.11 wireless LAN standard has been to define a way to connect wireless computers in local area networks. The main traffic in a LAN consists of file services or Internet traffic. But it has always been of high interest, how well the data centric WLAN technology can transmit interactive voice.

The DCF and PCF modes provide inadequate performance [24] and various performance improvements have been proposed and evaluated (overview in [17]). In the following we concentrate on papers that cover WLAN as well as QoS to enhance voice flows.

Veeraraghavan et al. [23] have analyzed how many voice flows can be transmitted simultaneously in an IEEE 802.11 network if the PCF polling mode is applied. D. Chen et al. [5] studied the capacity of IEEE 802.11b's PCF mode to transmit variable bit rate (VBR) VoIP calls. The capacity is up to 17 respective 10 voice calls in the considered modulation of 11 Mbps respective 2 Mbps.

In [15], Köpsel et al. simulated whether the DCF and PCF MAC mechanism can transmit real-time traffic. In the DCF mode stringent delay requirements are fulfilled only in low load scenarios. In a high load scenario or in a scenario with a high number of nodes, DCF fails to provide a low delay and low jitter. Therefore, the authors suggests to switch in those cases from the DCF to the PCF mode. In [15], the audio flows are transmitted over a 2 $MBit/s$ wireless channel. In case of an audio stream with 64 $kbit/s$ coding rate and $20ms$ packetization, the capacity is 12 stations in the DCF mode and 15 in the PCF mode. As a minimal quality level, the authors have chosen a maximum transmission delay of 250ms and maximal 5% packet loss. The usage of PCF, however, decreases the overall throughput because of unsuccessful polling attempts.

In a follow-up publication [16], Köpsel studies the benefit of higher data rates. An increase of the data rate (up to 54 MBit/s) leads only to limited quality improvements. This effect can be explained because of the packet overhead of the IEEE 802.11 PHY and MAC protocol, containing large protocol headers at a low rate, immediate acknowledgements (ACK) and large spaces between the packet transmissions (interframe spaces). Instead, to improve delay and jitter the authors suggest to use a transmit queue that supports two priorities. The high priority is reserved for interactive voice flows whereas the low priority is considered for best effort traffic. If the priority queue is present, the author does not see an immediate need for an extended DCF mode.

In [8], S. Garg et al. experimentally studied the capacity of IEEE 802.11b to determine the maximum number of VoIP calls. The maximum number depends on the packetization of VoIP (reciprocal of the packet frequency), the geographic distribution of the wireless client, and the distance between wireless client and base station. The authors measured quality of the VoIP calls by using packet delay, packet jitter and loss rate. Using G.711 and 10ms packet sizes six simultaneous calls were possible. Starting the seventh, the wired-to-wireless streams failed to meet the obligation regarding the packet loss. The authors concluded that lowering the packet frequency is the most efficient solution to increase the number of VoIP calls in an WLAN cell.

P. Garg et al. simulated the ability of IEEE 802.11e's EDCF and HCF coordination function to support a better QoS and higher channel efficiency [7]. They transmitted various flow types (audio, video and ftp) over a basic service set and measure delay distribution and bandwidth. The simulation is an extended version of Atheros Communication's 802.11e model for ns-2. Their findings lead to the conclusion that both coordination functions are highly sensitive to the chosen parameters. However, both can reach the desired QoS requirements but HCF has a higher bandwidth efficiency than EDCF.

Choi et al. [6] compared IEEE 802.11 DCF with IEEE 802.11e's EDCF and CFB according to throughput, dropped data rate and delay in an IEEE 802.11b PHY. In their scenarios, they used a combination of unidirectional voice, video and data traffic. They noticed a large decrease in dropped packets and delay as well as a more constant throughput for voice and video transmission in the EDCF simulations. CFB in addition to EDCF increased the throughput of voice and video only marginally but decreased the data dropping rate for both significantly. Because of the interaction of voice and video, the delay of video transmissions was reduced by CFB due to less contention overhead while it was increased for the voice flows.

Chapter 3

Simulation and Evaluation Environment

3.1 IEEE 802.11 EDCF Simulation Model

We used the discrete event simulator ns-2.26 [2] for our work where an 802.11 DCF model is already included. The ns-802.11 model does not provide the PCF as well as any MAC-management mechanisms like Association/Reassociation, Authentication/Deauthentication. In addition, no superframe structure with Beacon frames and no power-saving methods are supported. We expanded the ns-802.11 model by EDCF and CFB.

EDCF uses four priority queues with own backoff instances. The priority parameters of each instance are CW_{min}, CW_{max}, AIFS, TXOPLimit and PF. Further details about the implementation of the simulation model can be found in its open-source distribution [1].

3.2 Verification

During the implementation of our model we found a couple of errors in ns-2 and removed them. To ensure the correctness of our simulation model we compared it with the work of Mangold [18]. Mangold has implemented an EDCF simulation model in WARP and conducted some performance evaluations. If our simulation model achieves similar results to Mangold's work we assume it as verified and as “correct”.

Mangold [18] utilized the IEEE 802.11a-PHY with a data rate of 24Mbps, therefore we had to adopt the PHY parameters of our model, too. At first we considered and compared the maximum achievable throughput in ns-2.26 with the simulations in [18]. The scenario is a BSS consisting of a QoS AP (QAP) and only one wireless station. On this station one flow is sent to the QAP. [18] performed separate simulations for each TC with an increased generation rate respective a larger MSDU. The throughputs in Mbps are listed in table 3.1. We reached similar results with only minor differences.

We took the following scenario (figure 3.1) out of [18] to verify our model. The number of the wireless stations increases from 1 to 15. All stations are in the range of each other and the stations are not moving. Each station according to [18] provides three flows: a high-priority isochronous flow with 128kbps and 80byte MSDUs, a medium- and a low-priority Poisson

Table 3.1: *Our*/Mangold's results [Mbps] ([18])

Data frame size	80 bytes	200 bytes	2304 bytes
High	*3.52*/3.5	-	*19.98*/19.81
Medium	-	*6.32*/6.22	*19.32*/19.16
Low	-	*5.29*/5.21	*18.37*/18.22

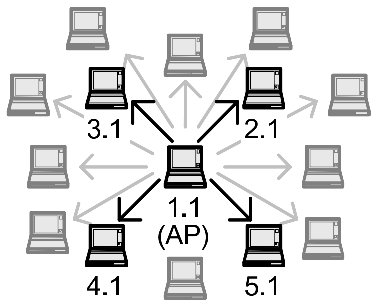

Figure 3.1: AP with variable number of stations (source: [18])

flow with each 160kbps and 200byte MSDUs.

We decided to use isochronous flows for all three TCs due to interface queue lengths of 50 packets as well as a medium interarrival time of 10ms for the 200byte packets. Therefore Poisson arrivals (*arrival rate=variance=100packets/s*) will be averaged out in overload scenarios, which we are interested in for testing our EDCF solution. The advantage of this simplification is an earlier termination of the simulations.

In [18], the backoff parameters were chosen as shown in table 3.2. The results of [18] are shown in figure 3.2 while our results are displayed in figure 3.3. In our simulations, the low and the medium priority flow can carry its traffic only up to a number of 8 respective 11 stations (in [18]: 9 respective 12 stations). Afterwards the curves drop faster and much more down in our simulations.

Our high priority flow can carry its traffic only up to 13 stations (in terms of throughput

Table 3.2: Mangold's [18] backoff-parameter set

	High	Medium	Low
AIFS	2	4	7
CWmin	7	10	15
CWmax	7	31	255
PF	2	2	2

Table 3.3: priority parameter set (source: [26])

	TC[0]	TC[2]	802.11b
AIFS	2	3	(2 = DIFS)
CWmin	7	31	31
CWmax	15	1023	1023
TXOPLimit	3 ms	0	-

per station) then it decreases slightly. Our results differ to Mangold in that respect. However, we explain this effect due to the different retransmission and packet drop behavior of both simulation models.

In our simulation model packets are dropped after the seventh collision. Retransmissions due to a collision occur often if many stations compete for the medium. Also, the overall throughput reaches an upper limit if the number of stations increases. Even after extensive checking of our simulation code, we see no indications to doubt the results of our simulation model and its implementation.

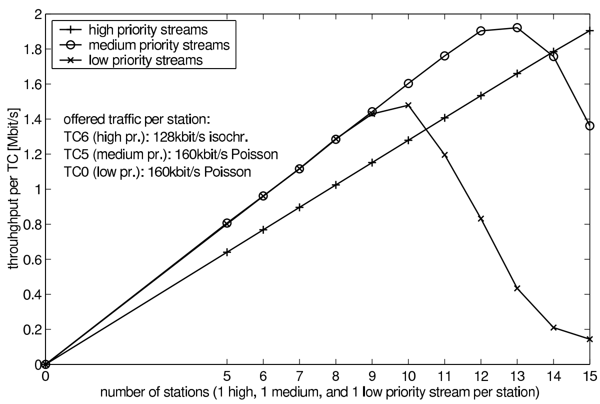

Figure 3.2: Mangold's results for increasing number stations vs. throughput (source: [18])

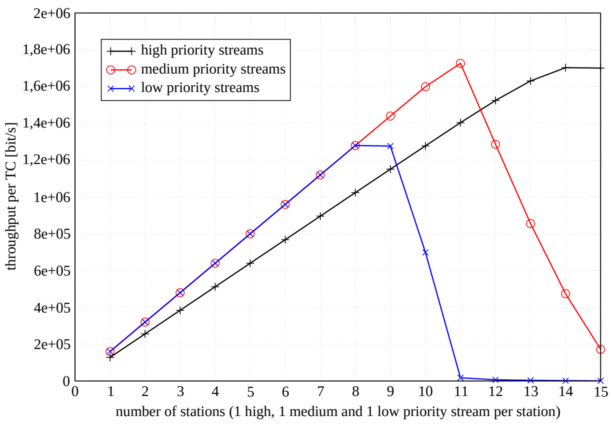

Figure 3.3: Our results for an increasing number stations vs. throughput

3.3 Simulation Scenarios

For the following simulations, we chose an 802.11b physical layer with a basic rate of 1Mbps and a data rate of 11Mbps. The 802.11e priority parameters were taken out of [26] and are shown in table 3.3 on the left side. For a better comparison, the 802.11b backoff parameters are listed on the right hand side. We basically consider two scenarios. Both consist of a base station connected to a wired network. The first scenario includes two stations in the wireless network as well as in the wired counterpart (figure 3.4).

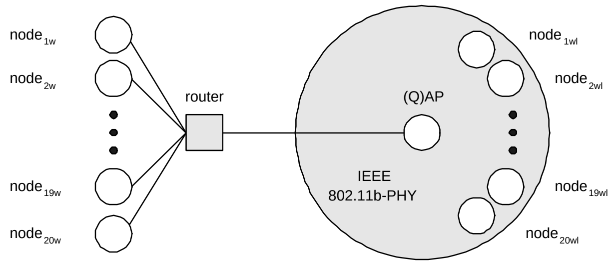

Figure 3.4: scenario

In the first scenario we consider the behavior of VoIP traffic in combination with best-effort traffic. In separate simulations we ran a CBR and an ftp stream together with VOIP traffic through the network. The CBR source as well as the FTP agent produce 1500 MSDUs with a rate of 11Mbps. Both best-effort flows were handled with the parameters of TC[2] and were sent from the wired into the wireless network. To analyze and to compare the QoS improvement of 802.11e EDCF for VoIP traffic, we simulated the first scenario with three different MACS: 802.11 DCF, 802.11e EDCF and 802.11e EDCF+CFB. In the second scenario the number of the wireless stations and their wired counterparts increases from 1 to 20 (figure 3.4) with all three MACs as stated above.

In both scenarios, bidirectional VoIP transmissions are used to take the behavior and the QoS requirements of phone calls into account. Voice flows in EDCF and EDCF + CFB are always handled with the highest priority, i.e. with TC[0]. We did not use an error model but the ns' TwoRayGround Propagation model, which considers the line-of-sight path and a ground reflection path. For all our simulations and their results, we determined the confidence intervals with a level of 90 percent.

3.4 Quality Evaluation

Perceptual quality assessment has to take into account the complete end-to-end transmission path, because it reflects human-to-human conversation, starting at the mouth of the talker until the acoustic signal reaches the ear of the listener. Thus, if a part of a transmission path is studied (in this paper the wireless link), anyhow the entire transmission system has to be considered.

The end-to-end quality depends largely on the playout buffer scheme [19]. Playout buffers temporarily store packets at the receiver to play them out in a timely manner. Packets that arrived too late to be played out on time are considered to be lost. In this paper, we apply fixed deadline schemes based on the absolute end-to-end transmission delay and the adaptive algorithms of Ramjee [20], and Moon [21].

Playout buffers adapt the playout time during the transmission. This rescheduling might harm the speech quality because of temporal discontinuities. However, the E-Model does not take into account the dynamics of a transmission but relies on static transmission parameters.

PESQ instead considers playout adaption but does not include the absolute delay into its rating. Therefore, we combine both models. PESQ calculates the speech quality and feeds the MOS value into the E-Model. To combine both models mathematically we applied the formula given in [11]. We describe our assessment method in detail in [12].

In mouth-to-ear transmission paths, also the acoustic components play a major effect, as the encoding scheme, the backbone, the packet loss concealment and the sample content. For simplicity, we consider the acoustic processing as perfect. The encoding/decoding applied μ-LAW [13] (G.711-codec, 64kbps coding rate, 20ms packetization, 200Byte MSDUs). The packet loss concealment algorithms are based on the standard [14] reference implementation. The backbone link is considered loss and jitter free but has a delay of 150ms. Samples are taken randomly from a large ITU speech data base.

Chapter 4

Simulations Results

4.1 Voice and background traffic

We simulate the vulnerability of a voice stream, when parallel TCP or UDP traffic is present. We apply the two node scenario by transmitting a bidirectional VoIP stream between the first pair of nodes. Between the second pair of nodes TCP or UDP background traffic is transmitted from the wired into the wireless network. In figure 4.1 we display the goodput of UDP and TCP flows (by bright bars) versus the R-factors of the voice flows (dark bars). To generate the background traffic we apply the standard ns-2 parameter set. The TCP implementation uses a segment size of 1460 bytes and a windows size of 64. The UDP flows (except for voice) has a packet size of 1500 at a constant bit rate of 11 MBit/s.

With 802.11 DCF and TCP we gain a goodput of about 4Mbps and therefore a small R-factor of 38. With DCF and UDP no phone call is possible due to a conversational quality equal to 0. In both cases, one reason for low quality is the handling of background and VoIP traffic in one queue. Because TCP adapts its sending rate, it does not prevent VoIP traffic entirely. In contrast, the UDP stream avoids any VoIP transmission in time. EDCF and EDCF + CFB highly improve the conversational quality of the voice calls and decrease the goodput of TCP and UDP.

Figure 4.2 allows a comparison of the R-factors for the down- and uplink direction. In this context, down- and uplink directions are defined from the point of view of the wireless nodes. In the downlink, 802.11 DCF with TCP in the background delivers only a minor quality and no quality with the UDP-background traffic as described above. In the uplink, the conversational quality is good for all MAC mechanisms, which is a result of different directions between VoIP and background traffic.

In figure 4.3 the speech quality is shown dependent on different end-to-end delays by variating the playout-buffer delay for the downlink. It is insufficient for DCF together with voice transmissions and any background traffic. This shows the need for a prioritized medium access scheme not only on the wireless nodes but also on the QAP.

If we are increasing the end-to-end delay (i.e. variating the fixed deadline of the playout buffer) up to 0.22s, then we are able to reach a minor speech quality also for DCF with TCP. Without 802.11e EDCF, voice transmissions with UDP traffic do not suffer QoS requirements (figures 4.2 and 4.3) and gain only a low quality with TCP traffic.

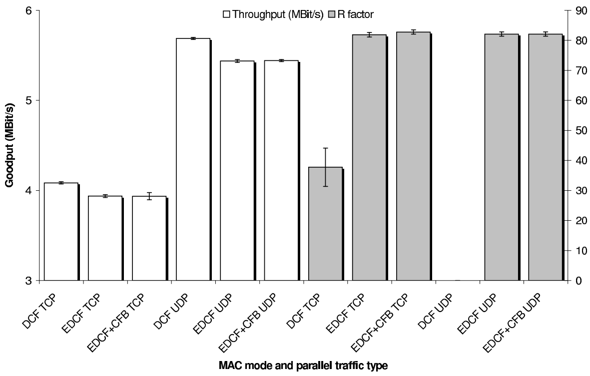

Figure 4.1: Goodput of UDP and TCP traffic vs. R-factors

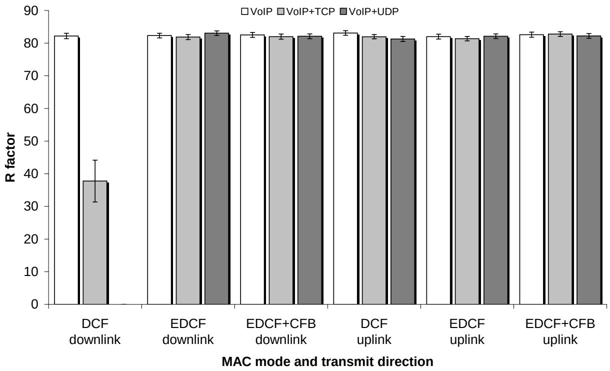

Figure 4.2: R-factors of down- and uplink direction

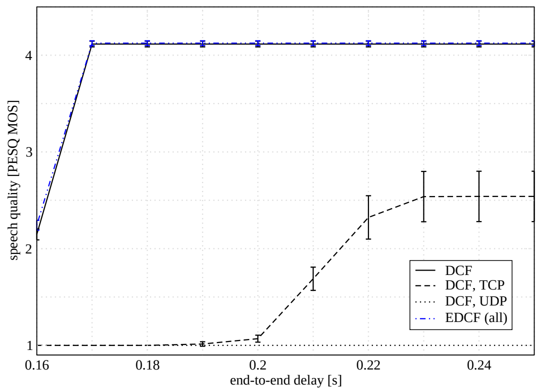

Figure 4.3: PESQ MOS of downlink direction

4.2 Tradeoff between cost and benefit

To know the maximum possible background load, we determined the throughput of the best-effort traffic without any voice transmission in scenario 1. The goodputs of the UDP and TCP flows are shown in table 4.1. Due to a lower priority for best-effort traffic compared to DCF, i.e. AIFS ¿ DIFS, we get lower goodputs for our UDP and TCP flows together with EDCF. The decrease of the TCP-goodput is much greater because TCP's flow control decreases the sending window due to packet losses, i.e. timeouts while waiting for a TCP-ACK. TCP's goodput is reduced by *4.25%* only due to the lower traffic categorization without any influence of voice transmissions. The UDP-goodput has only a very small decrease of *0.85%*. From our point of view, the costs of *0.85%* (UDP) and *4.25%* (TCP) are relative low compared to the strong improvements which with the EDCF comes up.

Table 4.1: background throughput of DCF vs. EDCF, no VoIP traffic

	UDP	TCP
802.11	5.95Mbps	4.47Mbps
802.11e	5.896Mbps	4.28Mbps
Difference	0.85%	4.25%

4.3 Capacity

In scenario two, we simulate how the number of telephone calls on different stations influences the mean R factor of all flows. The results are shown in figure 4.4 (downlink) and 4.5 (uplink). In the downlink direction, EDCF and EDCF+CFB are able to deliver up to 11

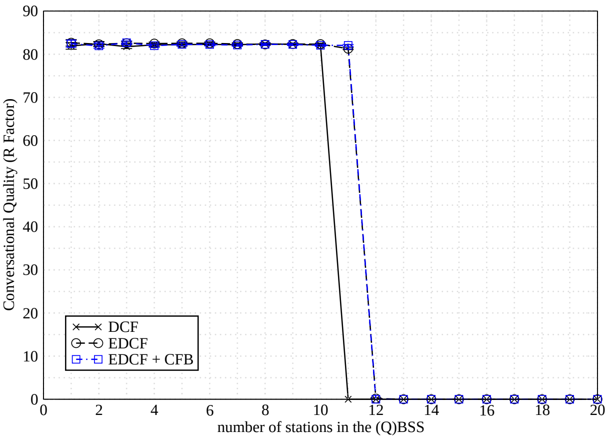

Figure 4.4: R Factors for increasing number of stations in downlink direction

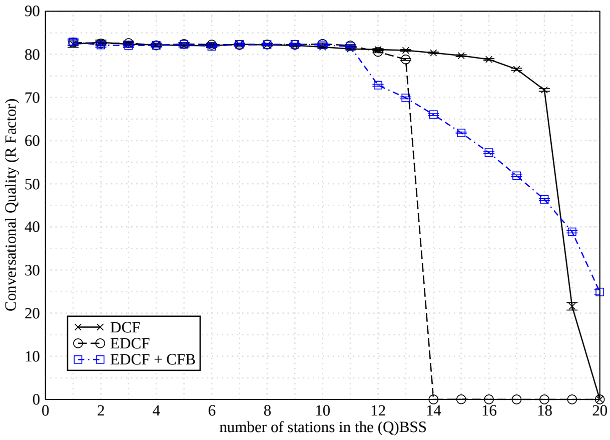

Figure 4.5: R Factors for increasing number of stations in uplink direction

voice transmissions, while DCF can serve only 10 VoIP flows. Due to the high-priority parameters (i.e. small CW_{min} and CW_{max}), the EDCF has a lot less contention overhead than the DCF.

The influence of the high-priority parameter set is also crucial for the uplink. The EDCF can provide good conversational quality for 13 stations. EDCF+CFB delivers a toll quality for 13 stations too, and decreasing quality for more than 13 nodes. 802.11's DCF realizes a good quality in the uplink direction for up to 18 stations due to better collision resolution. Comparing both directions, 802.11e EDCF and EDCF+CFB are sufficient for up to 11 voice calls, which we assume to be enough for most wireless Internet telephony scenarios.

In addition, the question arises whether different priority-parameter sets on wireless nodes and QAP will not lead to an increase in the maximum number of phone calls, too. Because the QAP has to serve all nodes in the BSS, from our point of view it should have a more frequent access to the medium than an ordinary node.

Our DCF results differ to the figures of the related work because the simulation scenarios differ. For example, different physical, encoding and packet schemes have been chosen. Thus, a direct comparison is difficult.

Garg [8] considered a quite similar DCF scenario as we did. He also applied DCF, G.711 coding, IEEE 802.11b with 11 Mbps, and multiple parallel telephone calls. However, he applied a packetization of 10 ms instead of 20 ms. Thus, the packet frequency is twice as high. However, the throughput of IEEE 802.11b is - in a first approximation - limited by the packet rate. Thus, one telephone call with 10 ms consumes about as much as two calls with 20 ms packetization time because in both cases the large overhead due to framing and contention is almost equal despite the size of the PDU. Consequently, the capacity that Garg has measured is about half (six calls) as the capacity that we gained (ten calls). Therefore, our DCF results confirm related work with respect to the number of maximal calls. Hence we conclude from this that also our EDCF results lie in a reasonable area.

Chapter 5

Conclusions and Outlook

In this paper, we present our modular and verified simulation model of the 802.11e EDCF for ns-2.26. We considered the influence of UDP and TCP background traffic, which prevents sufficient speech quality when using 802.11 DCF. 802.11e EDCF and EDCF + CFB come up with significant improvements that lead at least to a toll speech quality.

Taking not only the benefits but also the costs into account, we compare the goodput of single UDP and TCP flows in combination with the 802.11 DCF and 802.11e EDCF MAC. The TCP flow looses a little more than 4% of its goodput due to the different MAC This is a result of the lower priority of best-effort traffic in the EDCF, which leads to a smaller TCP-sending window and rate. UDP shows only a minor decrease of its goodput (less than 1%).

Finally, we determined the maximum possible number of simultaneous phone calls for the different MACs without any background traffic. With 802.11 DCF we reached a maximum number of 10 calls while 802.11e EDCF and EDCF+CFB are able to carry 11 (bidirectional) voice calls. Only for a higher number calls, the easy contention based medium access seems to be not sufficient because the contention overhead increases rapidly. The much more complex approach of the HCF might lead in this case to a better quality due to the polling based medium access although we assume that these scenarios will happen quite rarely.

Hoene et al. [11] brought into attention that the importance of VoIP packets differs greatly, depending on content and context of the packet (see figure 5.1). Also, the authors presented an algorithm which allows to measure the importance of VoIP packets. The importance of a speech frame is defined as degration of speech quality that will occur if the packet gets lost. Interestingly, about one third of all packets during voice activity can be dropped without a major degration in speech quality. Of course, this fact is valid only if the lost packets are unimportant. This effect can be tested via a public web site using a interactive Java applet [9].

As a next step, we will utilize the importance of VoIP packets to enhance the communication efficiency. Currently, we assign each data flow a static priority: Voice flows are transmitted at high priority; data flow at a lower priority. As a next step, we will assign each packet a priority, which depends on the type of flow but also on each packets' importance and the networks' background load. For example, important voice packet will get the highest priority and unimportant voice packets will be assigned to a a best effort traffic class.

First experimental measurements show that the speech quality can be enhanced [10].

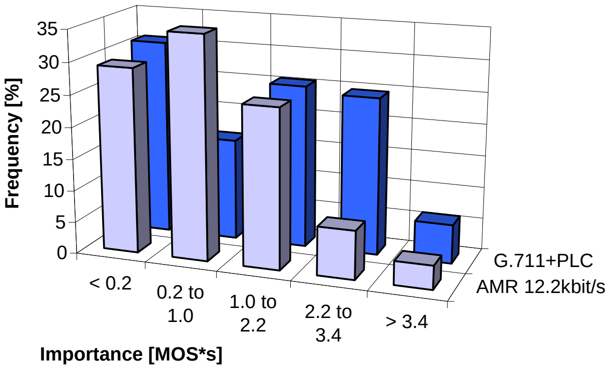

Figure 5.1: Importances of speech frames (higher values are more important)

Further studies, which use the simulation model described in this paper, will show the amount of performance gains that the so called selective packet prioritization approach can achieve.

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