Voice transmission in an IEEE 802.11 WLAN based access network

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ABSTRACT

IEEE 802.11 contains a mechanism for transmission of data with realtime constraints known as *Point Coordination Function*. This supplementary medium access protocol resides on top of the basic medium access mechanism *Distributed Coordination Function* and uses a centralized polling approach. Due to the complexity of a PCF implementation and the predicted inefficiency of the PCF several proposals have been presented for providing QoS support without the need of a centralized scheduler. Those solutions suffer from the fact that they are shifting implementation complexity from the access point to the mobile nodes. In this paper we compare the suitability of the basic DCF and PCF protocols for the transmission of audio data in an interactive scenario. We show that a simple priority mechanism used on the mobiles as well as the access point is suitable for providing improved QoS in terms of bandwidth and without the need of an extended DCF protocol. In combination with the PCF an adequate delay characteristic for audio flows is achievable as well. To overcome the limitations in channel capacity caused by the PCF we suggest an implicit signaling scheme for improving the channel capacity by avoiding unsuccessful PCF polling attempts.

Keywords

IEEE 802.11, WLAN, real–time, best–effort, voice transmission, scheduling, PCF, DCF

1. INTRODUCTION

Wireless LAN (WLAN) products are emerging in todays markets and have become a widely used Internet access technology. In 1997 the IEEE standardized *IEEE 802.11* wireless local area networks [1]. Motivated by the growing use of multimedia applications, support for time-bounded services was also integrated. This has been achieved by an extension of the basic medium access mechanism (*Distributed Coordination Function* – DCF) using a centralized polling-based mechanism (*Point Coordination Function* – PCF). In the last few years multimedia applications have become reality. Voice transmission as the most basic form of human communication has been spread out from the classical telephone oriented networks (POTS) to the packet oriented networks like the Internet (Voice-over-IP, VoIP).

In parallel, different wireless technologies have been developed to provide access for data and voice-based end systems. Next to wireless LANs for data transmission, cordless telephones have demanded attention for the wireless transmission of speech. The usage of multiple wireless technologies for providing telephone and data transmission might lead to a coexistence of different hardware infrastructures hence a multiplication of installation and maintenance costs is likely.

Thus the provision of a single wireless access method for the transmission of voice and data is attractive. In this paper we investigate the suitability of *IEEE 802.11* for the parallel usage for voice and data in an *IEEE 802.11* WLAN using DCF and PCF mode of operation. Previous research has postulated a high overhead in PCF mode of operation [2, 3, 4, 5], so providing real–time support is still an open issue in wireless LANs. Several approaches currently compete for providing QoS enhancements in *IEEE 802.11* :

1. Quality-of-Service might be achieved by using a distributed or centralized coordination scheme. In [6] a QoS enhanced signaling scheme is proposed that uses the basic DCF mechanism without any centralized control. Stations requiring instant channel access jam the channel with pulses of energy thus announcing their next transmissions priority. This approach requires fast switching from transmission to receive state, hence changes to the physical layer are mandatory.

2. Another group of distributed QoS-enhanced proposals introduce some kind of service differentiation in the MAC layer. Different approaches for sophisticated MAC procedures are presented in [7, 8, 9, 10]. Main issue is the QoS adapted tuning of the DCF backoff mechanism.

   All these DCF-based proposals suffer from the fact that they imply changes in the basic medium access mechanism or by modifying the standards protocol behavior.

3. The *IEEE 802.11* working group has provided basic real–time support by introducing the PCF. This centralized polling scheme might be enhanced by using improved scheduling or signaling schemes. In [8] an approach based on time slots during PCF is proposed. The access point is responsible for guaranteeing a fixed window in time for each station requesting real–time service, hence forming a traffic-adapted TDMA-like structure. Due to a stations ability to predict the time instant of the next expected poll, a bounded delay can be guaranteed. This approach suffers from the in-stationarity of the wireless channel with its fast varying error conditions and the unpredictable behavior of the underlying operating system used.

Service class differentiation for providing QoS to flows featuring different QoS requirements can be separated in the following basic problems:

1. Provisioning of a local priority mechanism to provide high priority packets preferred access to the MAC layer on a single mobile node. This can be achieved by a simple separation of audio and non-audio frames[1], but might also involve signaling packets of the real–time flow. Real–time related frames should always dominate best–effort packets.

2. Provisioning of priority MAC channel access for high priority packets in the uplink channel against other stations sending non-real time traffic. This can only be achieved by introducing some kind of service differentiation, that might be based on DCF extensions or on the PCF. While the latter offers a standardized approach, DCF extensions require changes in existing DCF implementations.

3. Provisioning of a scheduling policy on the access point in the downlink channel based on payload information from single packets. This can be done by introducing a downstream classifier for distinguish all incoming frames and using this information as an information feed for the local scheduling entity.

In this paper we show the improvements that can be achieved by using the most simple priority scheduling technique in the uplink and downlink channel. We assume a priority access for real–time packets: all traffic is separated in two different queues (real–time and non-real–time). Whenever packets are stored in both queues a station will transmit all real–time packets before any best–effort packet is served. Within the two queues, packets are transmitted according to a FIFO service discipline. Starting from this simple approach, we show the achievable QoS improvements in terms of bandwidth and delay when running a wireless LAN in DCF operation mode compared to a WLAN without local scheduling.

In addition we present estimations of the QoS achievable by the *Point Coordination Function*. The PCF offers service differentiation when combined with the local scheduling mechanism. For both medium access mechanisms we will compare the channel utilization and will introduce an implicit signaling scheme to improve channel efficiency.

The structure of the paper is as follows. In Section 2, a short introduction to the asynchronous as well as the synchronous operation of the medium access protocol of the *IEEE 802.11* standard is given. Section 3 covers briefly the architectural scenario assumed and presents audio QoS requirements in a typical wireless LAN environment. This is followed by a presentation of results for a DCF scenario serving as a reference point for further discussion in Section 4.1. The introduction of a priority-access-queueing on the access point as well as the mobile nodes and its impact on the audio QoS provided is shown in Section 4.2. The results obtained so far are compared with values obtained for the *Point Coordination Function* operation mode in Section 4.3. Finally, the improvements of channel utilization by using an implicit signaling scheme in the PCF are shown in Section 4.4.

2. THE IEEE 802.11 MEDIUM ACCESS PROTOCOLS

This section covers briefly the *IEEE 802.11* medium access protocols. Refer to the standard for additional details [1]. The DCF deploys a CSMA/CA approach which is in fact a 1-persistent random access protocol with delay. Such protocols work well under low load conditions, but suffer from significant throughput degradation and increased channel access delay in high load conditions. Transmissions are separated by inter packet gaps known as *Inter Frame Spaces (IFS)*. Channel access is granted based on different priority classes. These classes are mapped on different gap durations: Distributed-IFS (DIFS), Priority-IFS (PIFS), and Short-IFS (SIFS). A short overview of basic medium access is given in Figure 1: A station i which is willing to transmit randomly chooses a slot n_i out of a certain number of slots $(0, n_{max})$ and starts monitoring the channel for $DIFS + n_i$ consecutive slots. The station k with the lowest value n_k wins the contention phase and starts its transmission. All other stations detect a change in channel state (idle $\rightarrow$ busy) and restart the algorithm in the following contention round with a value of $(n_i - n_k)$ instead of randomly choosing a value. This gives higher access priority to stations in the next contention cycle which had to retreat from the current channel access. *IEEE 802.11* uses an im-

[1]This may be done by a simple classifier.

mediate acknowledgment scheme. Acknowledgments obtain channel access using a short interframe space (SIFS) with $SIFS < DIFS$.

For real–time traffic a supplementary medium access scheme called *Point Coordination Function* has been integrated in *IEEE 802.11* that uses a centralized polling scheme. As a consequence a central access point is an essential prerequisite. *IEEE 802.11* offers two different modes for running a wireless LAN: infrastructure and adhoc mode. While the latter offers the opportunity for creating a flexible independent adhoc network, use of the infrastructure mode offers interconnection to wired parts of a local area network. For infrastructure mode an access point is required providing bridging functionality.

The Point Coordination Function is an extension on top of DCF and provides three different basic types of service:

1. contention-free channel access for the access point for solely frame delivering on the downlink. This service is used to allow the AP to transmit any buffered traffic coming from the wired part of the network to the mobiles.
2. contention-free channel access for mobile terminals to support time-bounded traffic.
3. contention-free channel access for best–effort packet transmission. This mode might be used to overcome the limitations of the DCF in highload conditions [3].

Contention free service is provided by using an additional interframe space called PIFS where $SIFS < PIFS < DIFS$. It is used by the AP to gain and retain control of the wireless channel. Support of real–time services is based on a polling scheme avoiding the contention phase as used in the basic access scheme.

The polling strategy used by the AP is not fixed in 802.11, one option might be based on a Round Robin scheduler. In [1] a polling scheme according to the ascending association IDs of the associated mobile nodes is recommended. Stations which are polled and which have no pending traffic transmit a null packet back to the AP. If the contention free period (CFP) terminates before all stations have been polled, the polling list will be resumed at the next station in the following CFP cycle. A typical medium access sequence during PCF is shown in Figure 2. A station being polled is allowed to transmit an MPDU to any station within a wireless LAN. In case of an unsuccessful transmission the station may retransmit the frame after being repolled or during the next Contention Period.

2.1 Superframe structure

The AP controls the actual medium access scheme using a superframe structure as shown in Figure 2. The CFP repetition interval (*CFP Rate*) and length of a CFP (*CFPMaxDuration*) should be determined according to the characteristics of time-bounded traffic that has to be conveyed. The value of CFPMaxDuration shall be limited to allow coexistence between contention and contention-free traffic and must be limited to provide sufficient time to send at least one data frame during the contention period (CP).[2]

Two problems may arise with the *IEEE 802.11* superframe structure:

1. A foreshortened CFP may occur after a CP period when the access point is prevented from accessing the channel due to a busy medium. This may result in a shift of MPDUs to the next CFP cycle, causing additional delay. The maximum time shift is bounded by the maximum size and hence transmission time of a data packet during contention period.
2. A polled station may transmit frames of any length between 0 and 2312 bytes. At the beginning of a PCF cycle the total amount of bytes to be transmitted by the mobiles is not known. Due to the variable payload and duration needed for transmission, the AP may fail to poll all stations in the polling list during one cycle. Stations that have not been polled must postpone their frames queued for transmission to the next CFP causing an additional delay penalty.

[2]This must be guaranteed to allow the transmission of management frames.

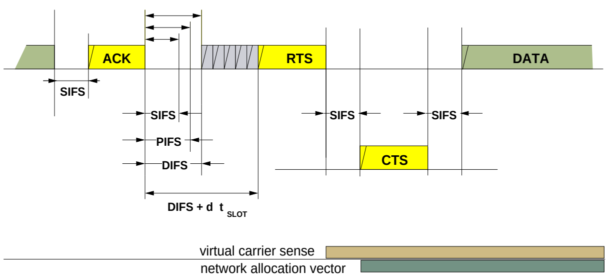

Figure 1: Distributed Coordination Function

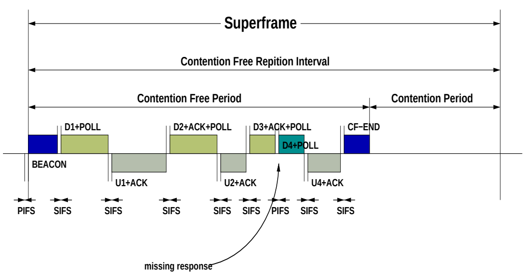

Figure 2: Point Coordination Function

To overcome the delay penalties caused by those time shifts a station must regain channel access as fast as possible. This might be achieved by a reordering of the polling list managed on the access point to allow the station a priority access in the next CFP cycle. As an alternative the station might attempt to transmit the packet during the subsequent contention period and avoiding intervention by the AP.

3. ARCHITECTURAL SCENARIO

In this section we introduce a basic scenario that will serve as a general starting point. Wireless LAN technology forms the access network that offers users connectivity to the Internet. Bridging functionality is provided by an access point that interconnects the wireless cell to the wired infrastructure (e.g. Ethernet-based), i.e. the *IEEE 802.11* WLAN is running in infrastructure mode. For practical reasons we assume a MAC PDU payload not exceeding a size of 1500 bytes to meet the constraints caused by *IEEE 802.3* based LANs.

We concentrate on the transmission of voice as the basic form of human communication. In recent years Voice-over-IP (VoIP) aware applications have been emerged based on the Realtime Transport Protocol (RTP) [11] and UDP/IP resulting in a protocol stack as shown in figure 3. This adds an overall overhead of at least 68bytes to every audio packet[3]. VoIP might be a replacement for classical telephone services working over longer distances. Hence all corresponding endpoints are assumed to be located outside of the wireless cell and all audio flows are passing the access point. The simulation model consists of a wireless cell containing M mobiles including the access point. The number of mobiles running audio connections is limited to N.

3.1 Source modeling

Due to the increasing amount of multimedia traffic generated in todays networks, mobiles located in the wireless cell use a dual source modeling that comprises a best–effort part and a real–time component. The numbers of mobiles running audio and best–effort connections is limited to $N < M$.

3.1.0.1 Audio

Audio flows are represented by packet trains. Users tend to stop their conversation, listen to their counterparts and restart their conversation; this effect is known as TALKSPURT. This behavior is independent from the codec used and is modeled by a two-State-Markov chain. The resulting packet train has exponentially distributed on and off periods with mean values $\lambda_{\text{TALK}} = 1.35$ ms and $\lambda_{\text{SILENCE}} = 1.15$ ms according to [2]. During TALK periods an audio flow is represented as an isochronous source with fixed interarrival times that are determined by the audio codec. According to the employed audio codec the amount of data for conveying speech data and the interarrival times vary as shown in table 1.

3.1.0.2 Audio QoS constraints:

Interactive audio owns several QoS constraints that must be fulfilled: when using echo compensation techniques, the round trip delay is limited to about 250-300 ms[4], i.e., the one-way delay is restricted to at most 150 ms. For PCM[5] encoding the loss rate should never drop under a percentage of 5% of all generated frames to prevent significant losses in quality. Other coding techniques use interframe dependencies to reduce the amount of data that must be conveyed so the acceptable loss rate might be lower for advanced coding techniques.

After creation a residual lifetime is assigned to each audio packet which is stored in a transmission queue until it is conveyed. If the lifetime expires before the packet is transmitted successfully, the packet should be removed from the queue to avoid unnecessary transmission of a worthlessly packet. The overall delay experienced by an audio packet depends on several issues: (1) the mobile nodes operating system support for real–time traffic, (2) the LAN medium access protocol and (3) the delay caused in the wide area network that must be traversed by an audio packet on its way to the destination. According to the distance that must be traveled from the mobile to the corresponding host, some share of the overall one-way delay of 150 ms must be reserved for the WAN and destination system delay, i.e. the max. lifetime of 150 ms must be decreased. As a consequence, we vary the effective max. lifetime of audio packets for each flow between 50 ms and 150 ms in our simulation to cope with the before mentioned requirements.

[3]header sizes without options: IP 20bytes, UDP 8bytes, RTP 40bytes

[4]without echo compensation this drops to 50 ms

[5]see G.711 codec in table 1

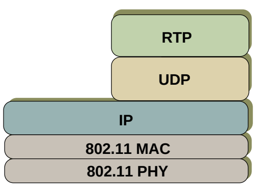

Figure 3: Upper layers in an audio transmission

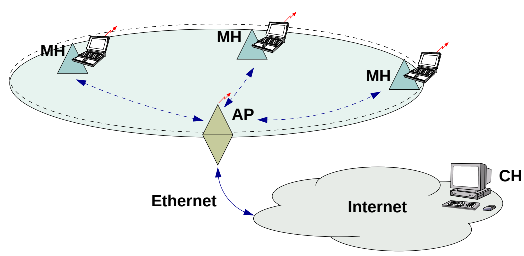

Figure 4: Scenario wireless LAN

TABLE 1
System parameters for different audio codecs

Codec	bitrate [kbps]	payload size [bytes]	frame duration [ms]	pkts [1/s]
G.711	64	160	20	50
G.723.1	5.3	20	30	33
G.723.1	6.4	24	30	33
GSM	13.2	33	20	50

TABLE 2
System parameters for different PHYs

IFS	FHSS [μs]	DSSS [μs]	OFDM [μs]
SIFS	28	10	13
PIFS	78	30	19
DIFS	128	50	25

3.1.0.3 *Best–Effort*

The Best–effort traffic model used on the mobile nodes is separated in the creation of an appropriate packet length distribution and the generation of interarrival times. The packet length distribution used is extracted from a packet trace made at Harvard University [12]. This trace was made on the university's backbone connection to the Internet, based on a $10Mbps$ Ethernet segment. The traffic observed was limited to $1500bytes$ payload. *IEEE 802.11* provides the ability to transmit data payloads up to 2312 bytes, but due to our scenario assuming the wireless cell as last hop in a wired LAN environment the best–effort payload size is bounded to 1500 bytes. Interarrival times are based on a Pareto distribution (see [13]) with probability density function $f(x) = \frac{ak^a}{x^{1+a}}$.

3.2 Radio channel modeling

IEEE 802.11 provides different PHY technologies. We ignore systems with lower data rates of 1 and 2Mbps and concentrate on higher-rate technologies: *IEEE 802.11b* describes a DSSS system with 11Mbps; *IEEE 802.11a* is an OFDM transmission system with data rates of 2, 11, 24 and 54 Mbps. Transmission speeds of 36,48 and 54 Mbps are optional hence we investigate the highest mandatory transmission speed of 24Mbps as well as the highest optional achievable data rate of 54Mbps. *IEEE 802.11* is based on interframe spaces to provide different channel access priorities. These vary depending on the PHY used as shown in table 2. For speed adaption all PHY headers (sync signal and PLCP header) are sent with a basic rate of 1 Mbps (DSSS, DBPSK) resp. 6 Mbps (OFDM, BPSK) resulting in an additional PHY header delay of 192 bits (DSSS PHY preamble and PLCP header, $192\mu s$) resp. 11 symbols + 48 bits (OFDM sync, PLCP header, $60\mu s$).

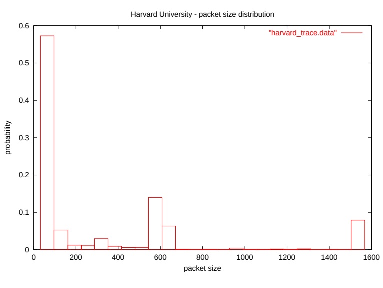

Figure 5: Packet size distribution of harvard trace

3.2.0.4 *Gilbert Elliott Channel model:*

The radio channel is modeled using the Gilbert-Elliott approach as shown in [14]. Spatial distribution of stations in the wireless cell as well as physical phenomena that include attenuation, intersymbol interference, noise and fading are mapped to a two-state-Markov-chain. Each state corresponds to a characteristic bit error rate: BER_{Good} and BER_{Bad}. A dedicated channel is assigned to each pair of stations (i, j) resulting in $n(n-1)$ independent channels.

3.3 Metrics

The definition of all QoS parameters used in our simulations is shown in figure 6. We differentiate among the mean channel access delay (`MCAD`) and the mean channel transfer time (`MCTT`). While the first provides an estimation for the time needed to gain access to the physical channel, the latter shows the mean time needed to traverse the wireless link

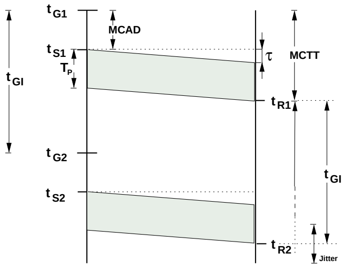

Figure 6: Metrics

(without the notification with an ACK) for the packet including the channel access delay. In addition we measure results for the audio flows jitter. An audio source represents a constant bit rate source in the TALK state, generating packets at a fixed rate. Hence we can determine from the generation timestamps the variation (jitter) of audio packets successfully transmitted to the access point. High jitter must result in an adequate dimensioning playout buffer increasing the overall delay.

4. RESULTS

The current section presents results for a simulation scenario of 12 mobiles nodes (including the access point) and 4 mobiles with audio connections.

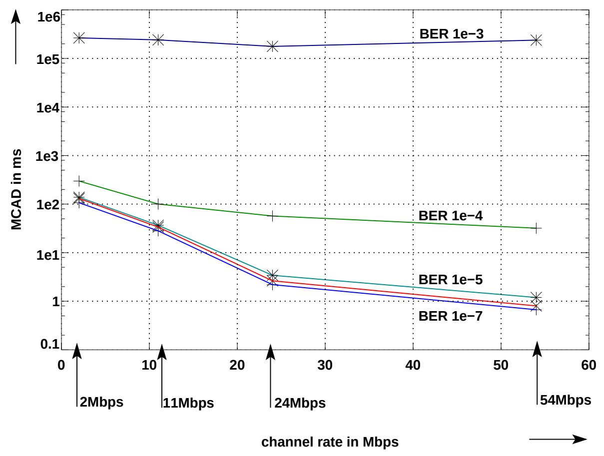

Figure 7: Mean Channel Access Delay (MCAD) for varying BERs in DCF, parameter channel rate

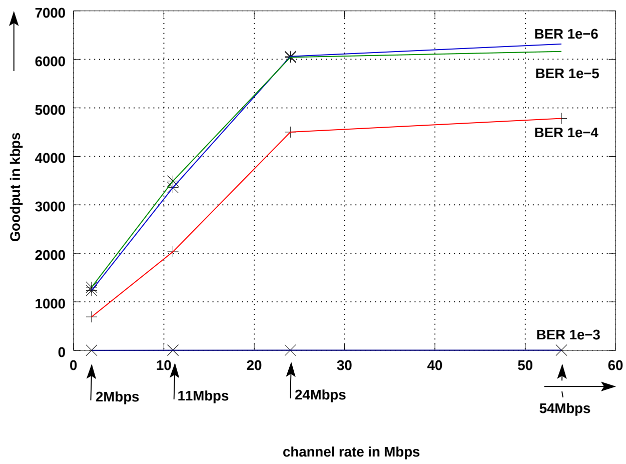

Figure 9: Goodput for varying BERs in basic scenario DCF, parameter channel rate

4.1 DCF

As a reference point we provide results obtained for the DCF mode of operation. Both the mobile hosts as well as the access point use one common queue for storing packets to transmit thus real–time packets

have to compete locally against other (non-real–time) frames and other mobile hosts within the wireless cell attempting to access the channel. Figure 7 shows the impact of mean BERs between 10^{-7} and 10^{-3} resp. on the mean channel access delay while varying the channel rate as a parameter. Channel access delay values remain stable for bit error rates up to 10^{-5}, but increase significantly for BERs exceeding 10^{-5}. This is caused due to the augmented occurrence of *Extended Interframe Spaces* for stabilizing the erroneous wireless link after bit errors have been detected due to corrupted frame check sums. Extended Interframe Spaces are used to prevent a station from transmitting when it has detected an error in the frame check sequence contained in the last received packet.

The results obtained so far demonstrate:

1. Due to the mean channel access delay of at least 100 ms a data rate of 2Mbps is not suitable as an access speed for mobile nodes with interactive audio flow requirements even for low bit error rates of 10^{-7}.

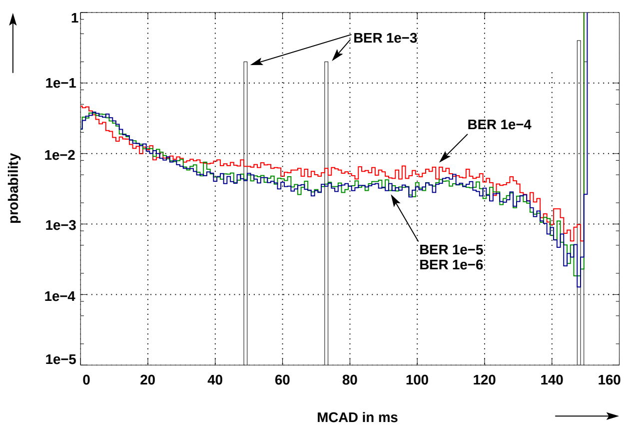

Figure 8: Probability density function of Mean Channel Access Delay (MCAD) for DCF at channel rate 11Mbps

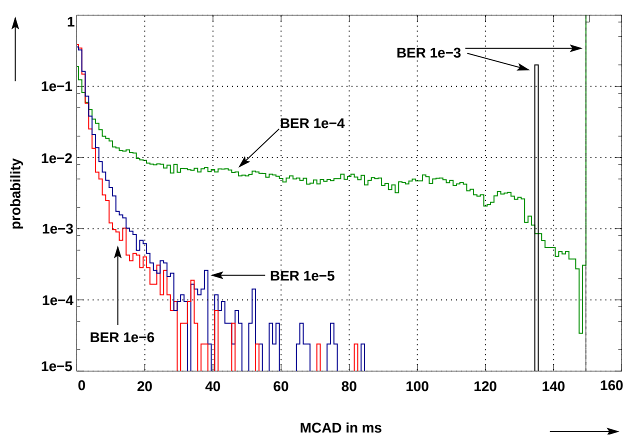

Figure 10: Probability density function of Mean Channel Access Delay (MCAD) for DCF at channel rate 54Mbps

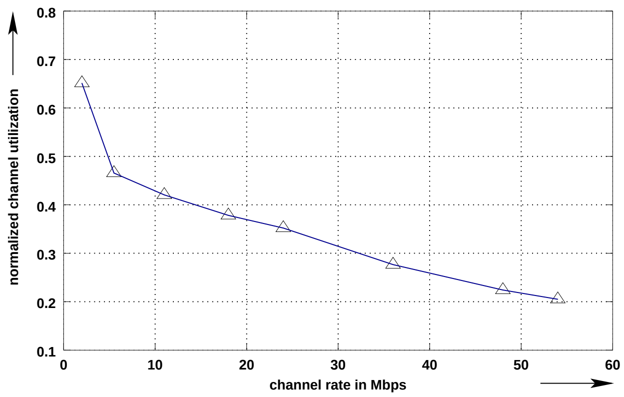

Figure 11: Normalized channel utilization vs. channel rate for a non-erroneous wireless link

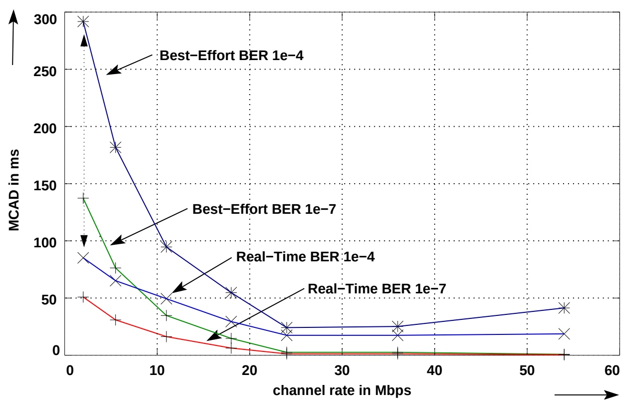

Figure 12: Mean Channel Access Delay (MCAD) for different BERs in DCF with real–time priority access scheduling

2. According to our simulation assumptions of 12 mobile nodes including 4 audio flows a data rate of at least 11 Mbps is mandatory. This guarantees a mean channel access delay of about 40 ms for our scenario, a residual time of $\approx$ 100 ms remains for passing the WAN. Hence, higher channel rates seems to be attractive to increase the cells capacity for audio flows. Figure 9 shows goodput for the same simulation series: an almost identical throughput can be observed for channel rates of 24 Mbps and 54 Mbps, raised by the overprovisioning of bandwidth to the offered load. A bit error rate of 10^{-3} leads to packet losses of nearly 100% resulting in zero throughput.

QoS support might be achieved by simply introducing higher data rates in *IEEE 802.11* DCF for reducing the mean channel access delay. Bandwidth enhancements in *IEEE 802.11* can be expected in the near future like the development for wired LANs has shown in the last years. But this approach has limitations as the application throughput gain will not increase linearly according to the improved wireless link speed due to the characteristics of PHY transmission in *IEEE 802.11* . While the MAC PDU is transmitted with full speed, all PHY PDU parts are sent with a basic rate[6]. In addition all interframe spaces are of fixed length thus a static overhead independent from the actual maximum channel rate must be taken into consideration limiting the achievable link capacity. Higher data rates will increase the inefficiency of the DCF, which motivates the use of an centralized access scheme. The centralized scheme shows also advantages in high load conditions (see [3]).

For determining the physical boundary of link capacity for different channel rates and comparing the normalized capacity in the next simulation run, the overall load offered to the wireless cell was maximized by using again the Ethernet traffic mix, but combined with a reduced packet interarrival time set to zero. Figure 11 presents the normalized goodput for a non-erroneous wireless link ($BER = 0$) vs. channel rate. The curve reveals the impact of the physical layer on the overall performance. A simple increase in channel speed in *IEEE 802.11* DCF leads to a deteriorate normalized channel capacity, e.g. an increase in channel speed of a factor of ≈ 5 (11Mbps $\rightarrow$54Mbps) leads to an application usable capacity improvement of factor $\approx$ 2.2 (effective goodput 45% of 11Mbps $\approx$ 4.95Mbps $\rightarrow$20% of 54Mbps $\approx$ 10.8Mbps).

The end-to-end delay experienced by an audio flow is caused by the originating LAN, the intermediate WAN nodes and the destination local area network resulting in a highly variable end-to-end delay. Hence a strict limitation of delay bounds in the WLAN does not make sense, except the overall threshold of 150 ms for interactive audio. For revealing results for audio losses Figure 8 and 10 show probability density functions of the mean

4.2 DCF with audio priority access

As mentioned in the previous section, real–time packets have to win two different competitions: (1) If a single transmission queue is used, real–time packets are mixed up with best–effort packets. An interactive audio flow with its strict delay requirements should always gain higher priority access to the MAC layer than other non-delay-sensitive traffic. (2) Stations giving audio packets priority MAC access must compete with other stations probably sending non-real–time traffic. To assure a priority adapted negotiation of channel access some proposals have been made, based on changing the basic medium access mechanism used in *IEEE 802.11* . This section shows the performance improvements that are achievable by using a simple classifier to determine audio packets and giving them priority MAC access on the mobile hosts and the access point.

[6] 1Mbps for DSSS, 6Mbps for OFDM

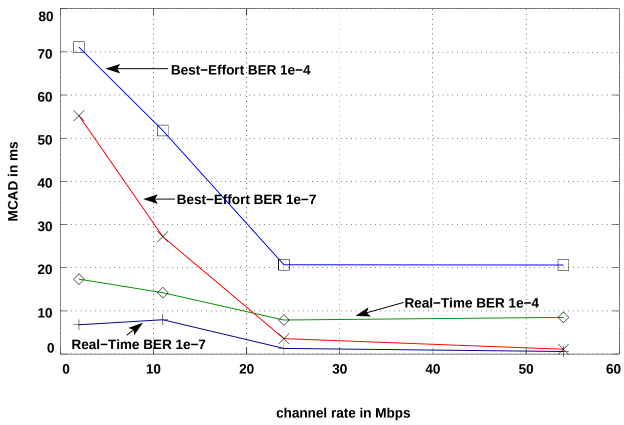

Figure 13: Mean Channel Access Delay (MCAD) for BERs 10^{-4} and 10^{-7} in PCF for real–time and best–effort traffic

Figure 12 shows mean channel access delay vs. channel rate for two different bit error rates: 10^{-4} and 10^{-7}. Comparing these curves with those from Figure 7 on page reveals a significant performance gain for real–time packets. An example: while the channel access delay is at about 100 ms for a single queue (BER 10^{-4}) for a channel rate of 11Mbps, the mean delay for real–time packets with priority access drops to about 50 ms. Channel rates of 24Mbps or higher do not benefit from the separated transmission queues: Best–Effort and Real–Time traffic perceive nearly identical channel access delays.

4.3 PCF

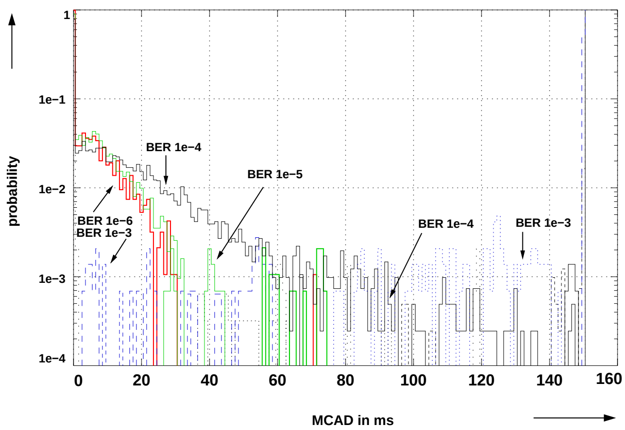

Figure 14: Probability density function of Mean Channel Access Delay (MCAD) for PCF at channel rate 11Mbps

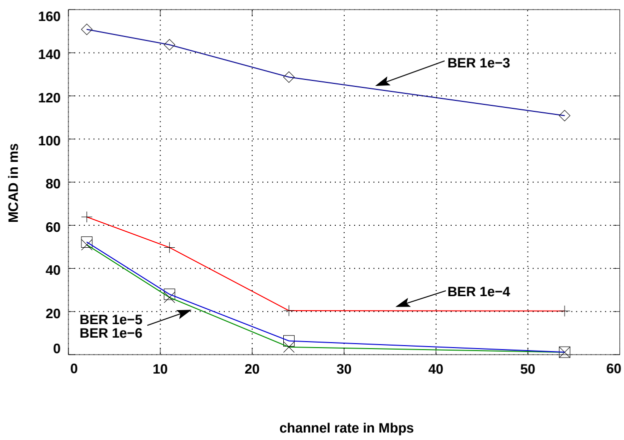

Figure 15: Overall Mean Channel Access Delay (MCAD) for varying BERs in PCF

IEEE 802.11 is the first protocol from the family of IEEE 802.x protocols that has built-in support for traffic with real–time constraints. Based on a centralized polling scheme the high overhead is assumed to lead to a (probably high) but low-variance delay and a dramatic decrease in overall throughput [2]. Due to the centralized scheduler, polling attempts might fail in phases of active talkspurts in an audio conversation hence leading to a control packet exchange with zero payload transmission. This section gives results for PCF mode of operation with a scheduling scheme according to recommendations given in [1]. A RoundRobin style scheduler is used for all stations sending audio packets. Stations with no audio traffic will not be polled during the Contention Free Period. The superframe interval is chosen according to the values given in table 3.1, e.g. 30 ms for a G.723.1 codec (5.3kbps). If not all

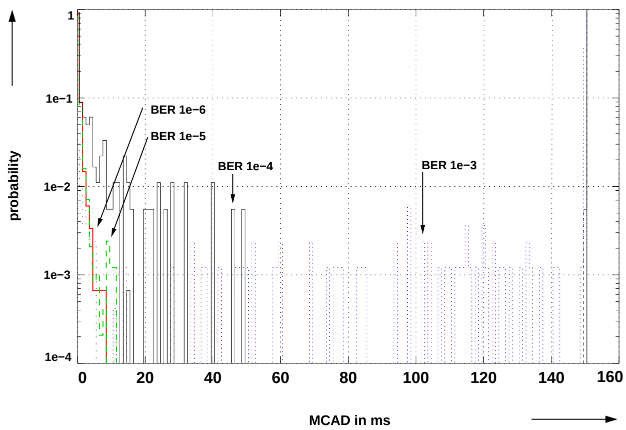

Figure 16: Probability density function of Mean Channel Access Delay (MCAD) for PCF at channel rate 54Mbps

stations can be polled during a superframe interval due to the given CFP length restrictions, the scheduling will restart at the next station that could not be polled in the last interval. The CF phase is not extended for retransmitting erroneous audio frames. Such frames will be retransmitted during the following contention period.

Figures 15 and 13 show appropriate curves for comparison of Mean Channel Access Delay for DCF and PCF mode of operation. PCF offers when compared with priority access in DCF an additional channel access improvement, e.g. for 11Mbps channel rate we obtain: BER 10^{-4} best–effort drops to a mean access delay of $52ms$ compared with $\approx 100ms$ for CBQ style DCF while real–time packets perceive a MCAD of about $15ms$ vs. $50ms$. In addition Figures 14 and 16 present the probability density functions for PCF mode of operation for 11 and 54Mbps resp. These curves reveal a more packed *pdf* in PCF mode for both channel speeds indicating a lower variance up to bit error rates of 10^{-4}. BERs of 10^{-3} and higher cause additional delay due to the increased number of retransmissions resulting in a more uniformly distributed pdf.

The improvements for mean and variance of channel access delay due to the *Point Coordination Function* raise a penalty in overall throughput. Figure 18 shows the resulting goodput for DCF and PCF. Compared with the results from Figure 9 a performance loss can be observed for channel rates of 2 and 11Mbps. The overall goodput drops from 1300 to about 1140kbps for channel rate 2Mbps and 3490 to 3200kbps for 11Mbps for a BER of 10^{-6}.

For higher rates (24 and 54Mbps) an increase in overall throughput in PCF mode is shown by the curves, e.g. 6160kbps to 6850kbps for 54Mbps. The use of the PCF reduces the overall contention phase in DCF due to the transmission of short audio packets in the CFP resulting in a higher DCF and overall throughput. This gain in throughput entails a slight increase in the best–effort channel access delay.

4.4 PCF with implicit signaling

Following the results obtained so far, we have seen a decrease in overall throughput for PCF mode of operation compared to DCF operation when the offered load reaches the wireless cell capacity. The centralized scheduler introduced with PCF avoids contention phases and provides channel access to individual stations. Stations that want to obtain service during the CFP, make an association with the access point to be admitted to the APs polling list. *IEEE 802.11* suggests a RoundRobin style polling scheme. Unsuccessful polling attempts might be caused by different phenomena:

- Due to the occurrence of talkspurts, the application will stop to send further audio frames.
- When an audio packet is ready for transmission, the sending station might choose to wait for the next PCF slot to transmit the frame. Waiting for the next polling cycle increases the overall delay by an additional waiting time, hence if the wireless cell is running in contention period, the audio frame should be transmitted as fast as possible in the CP. This might result in an audio packet free transmission queue in the next CFP cycle.
- A polling frame might be lost in the CFP.

To improve the efficiency of the *Point Coordination Function* , we introduce a signaling scheme to avoid unsuccessful polling attempts due to talkspurts or early packet transmissions. The access point might use the information available from higher layers in the audio frames. E.g., RTP provides an SSRC identifier to distinguish different audio flows as shown in Figure 17. The access point might use this identifier for classifying individual audio flows. RTP provides a marker bit and header extensions to carry payload dependent information like frame boundaries or talkspurt beginnings. Using this mechanisms leads to an explicit in-band-signaling. We have used the most simple signaling scheme available: Audio frames are sent during PCF. When a TALKSPURT phase ends, the next polling attempt by the access point fails and the AP removes the station from its polling list. When the station restarts sending audio packets, the first packet is sent in DCF mode of operation. The centralized scheduler detects the continuation of the audio flow and reassigns the station automatically to its polling list.

Results for determining the improvements by using such an signaling scheme to avoid unsuccessful pollings are shown in Figure 18. The additional curve marked with triangles reveals a bandwidth enhancement of $\approx$ 400 kbps at a channel rate of 2 Mbps and of $\approx$ 300 kbps at a channel rate of 11 Mbps.

5. CONCLUSION

In this paper we have studied *IEEE 802.11* as a local area network access technology and its suitability for the transmission of audio flows. While the DCF offers sufficient audio QoS support in terms of mean channel access delay for channel rates of at least 11Mbps, use of the DCF lacks the ability to provide low variance. High variances may result in a significant loss of audio packets at the receiver due to the strict delay requirements for interactive audio. Increasing the available bandwidth by introducing techniques like OFDM may solve this problem but due to the overhead introduced by

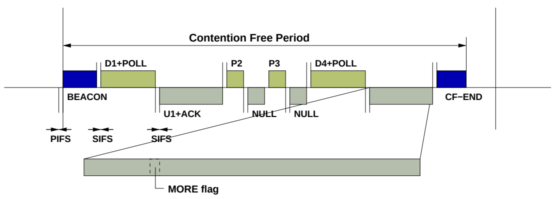

Figure 17: Simple implicit signaling scheme using a `MORE` flag

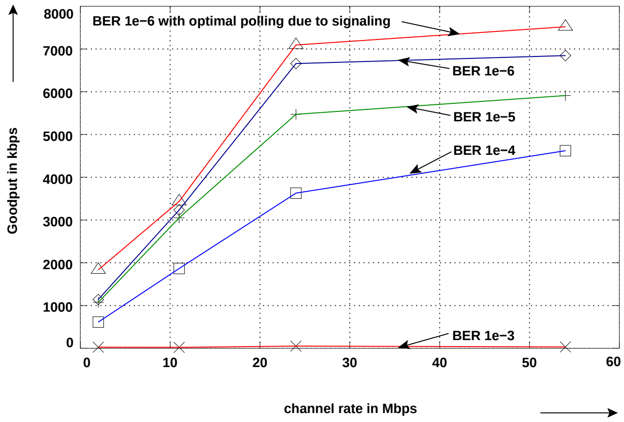

Figure 18: Goodput for Point Coordination Function

the *IEEE 802.11* PHY mechanisms, a significant share of the additional bandwidth will not be available by the application.

Efficient support of audio requires a local scheduling policy on the mobile nodes as well as the access point. A simple priority access mechanism reduces the mean channel access delay without the need of a complex change in the medium access protocol. A separation of audio and non-audio frames must be provided by a classifier and support for multiple transmission queues must be available on the nodes.

The most suitable medium access mechanism for audio is the *Point Coordination Function* that reduces the mean channel access delay once more by avoiding contention phases during CFP and reducing contention in CP. In addition it offers a very low variance. We have also shown that the use of the PCF decrease the efficiency of *IEEE 802.11* for lower data rates up to 11Mbps. But as the efficiency of the DCF decreases with higher data rates above 11Mbps the inefficiency of the PCF plays a less significant role. The result is a by far better performance of the PCF which support the fact that a centralized access control scheme should be used at higher data rates.

By avoiding unsuccessful polling attempts, the PCF can be optimized in terms of overall throughput. A simple scheme of removing and adding a station to the polling list can be used to increase the overall throughput.

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