QUALITY OF SERVICE PERFORMANCES IN AD HOC IEEE 802.11 WIRELESS LANS

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Abstract: WLAN is a wireless network which provides connectivity in a limited area. IEEE 802.11 is the most widespread standard for wireless LANs, but it is not suitable for real time services. The draft standard IEEE802.11e provides solutions for Quality of Service (QoS), and maintains the compatibility with the IEEE802.11 standard. The paper, after a brief introduction on the WLAN technology, describes the IEEE802.e solutions for QoS and provides simulation results in an ad hoc network with different loads. It is shown that QoS in an ad hoc network can be provided with completely distributed techniques even if the network is heavy loaded with real time services.

1 INTRODUCTION

WLAN is a wireless network which provides connectivity in a limited area. IEEE 802.11 is the most widespread standard for wireless LANs, which includes two network topologies:

Infrastructure Network, consisting of a Distribution System (DS) that connects two or more Access Points (APs). Each AP provides a radio coverage and each station (STA) is attached to one AP. One AP with the attached STAs is called Basic Service Set (BSS). The DS with the connected BSSs is an Extended Service Set (ESS). The ESS allows the communication between STAs belonging to different BSSs.

Ad Hoc Network, where the STAs are connected peer-to-peer. An Ad Hoc Network self-creates, self-organizes, self-administrates. The STAs sharing a radio channel form an Independent Basic Service Set (IBSS). Performances of an Ad Hoc Network strictly depend on the STAs number, on the mutual distance, and on their instantaneous position. All the algorithms are completely distributed.

IEEE 802.11 is the most widespread standard for wireless LANs, but it is not suitable for real time services. The draft standard IEEE 802.11e provides solutions for Quality of Service (QoS), and maintains the compatibility with the IEEE 802.11 standard.

The paper is organized as follows. Section 2 describes the IEEE 802.11 WLAN standard, section 3 deals with the required changes to the MAC protocol for the support of QoS.

In section 4 the simulation scenario is described; in section 5 simulation results are provided with comments and conclusions.

2 IEEE 802.11 WLAN

The standard IEEE 802.11 defines the Physical (PHY) and Medium Access Control (MAC) layer for Wireless LANs, both for *infrastructure* and *ad hoc* topologies.

The most important physical layers are:

b, which works at 2.4 GHz and provides up to 11 Mb/s

a, which works at 5 GHz and provides up to 54 Mb/s

g, which works at 2.4 GHz and provides up to 54 Mb/s. The g PHY is backwards compatible with the b standard.

The MAC is unique for each PHY, and is based on Carrier Sense Multiple Access/Collision Avoidance (CSMA/CA). The implemented function is called

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Distributed Coordination Function (DCF), and works as follows. Before the transmission, each STA listens the wireless medium: if it is heard free for a DIFS (*Distributed Inter Frame Space*) time, the STA sends the bits, otherwise it launches the backoff procedure. The backoff procedure calculates a *backoff time* through a random function which takes uniformly distributed values between 0 and *CW* (*Contention Window*), where *CW* is always:

$$CW_{min} \leq CW \leq CW_{max}$$

Initially *CW* is initialised at $CW_{min.}$ The backoff time is calculated as follows:

$$Backoff\_time = CW \times Random() \times SlotTime$$

Standard values for the *SlotTime* are 31µs for 802.11b and 15µs for 802.11a. Because the *CW* value is chosen as a power of two minus one, if *Random()* assumes integer values the *Backoff_time* is:

$$Backoff\_time = [(2^k - 1) \times Random()] \times SlotTime$$

where $CW=2^k$-1. When a STA which has launched the backoff procedure finds the medium as free, it begins to decrement of a slot time the backoff timer until it relieves the medium as occupied. When this timer reaches zero value, the station transmits the Mac Service Data Unit (MSDU). Each MSDU which has been correctly received must be acknowledged with and ACK frame. If the ACK is not received into an *ACKtimeout* time, then the transmission is considered unsuccessful and the backoff procedure is launched by duplicating the previous *CW* value as follows:

$$CW_{new} = 2 \times (CW_{old} + 1) - 1$$

The backoff time at the i-th tentative of access is:

$$Backoff\_time = [(2^{k+i} - 1) \times random()] \times SlotTime$$

When a STA has successful transmitted a MSDU, it launches a post-backoff procedure in order to allow other stations to access the medium.
DCF is the basic MAC for both infrastructure and ad hoc 802.11 networks. An added function, not mandatory from the standard, is the *Virtual Carrier Sense* (VCS), which solves the hidden node problem with RTS/CTS frames. Finally, the PCF is an optional access technique which can be implemented only in infrastructure networks where the *Point Coordinator* (PC) regulates the access to the medium during a time called Contention Free Period.

3 QOS IN IEEE 802.11

DCF is for best effort services, because it does not provide QoS. In fact all the stations belonging to a BSS or an IBSS compete with the same priority to access the same wireless medium.
In the standardization bodies was accepted that QoS mechanisms had to be added in the 802.11 standard, and in 1999 the task group TGe was created, that later gave birth to the draft standard IEEE 802.11e. This new MAC maintains the compatibility with 802.11.
In IEEE 802.11e, the stations are named QSTA, the BSS and IBSS are QBSS and QIBSS. It is worth to underline that the existence of a QBSS or a QIBSS does not preclude the good functioning of the non-QoS stations.
With IEEE 802.11e the MAC is enhanced by the *Enhanced Distributed Coordination Function* (EDCF), which is a completely distributed technique that provides a service differentiation based on 8 priority levels, named *User Priority* (UP). Each QSTA can manage 4 *Access Categories* (AC), and each AC has a different value for DIFS, which in IEEE 802.11e is renamed AIFS (*Arbitration InterFrame Space)*, CW_{min} and CW_{max}. In general, the higher is the priority, the lower are AIFS[AC], CW_{min}[AC] and CW_{max}[AC] parameters.
The mapping between UP and AC is represented in table 1.

Table 1: mapping between UP and AC

User Priority (UP)	IEEE Definition	**Access Category (AC)**
0	Best Effort (BE)	0
1	Background (BK)	0
2	-	0
3	Excellent Effort (EE)	1
4	Controlled Load (CL)	2
5	Video	2
6	Voice	3
7	Network Control (NC)	3

The backoff interval is given by:

$$Backoff\_time = Random() \times SlotTime$$

where *Random*() assumes uniformly distributed values in (1, *CW*[AC]+1).
In EDCF there are 4 traffic queues, where are mapped the eight defined User Priorities.

Each queue is characterized by different access parameters.
Each time the backoff procedure is relaunched because of transmission failure of internal collision, the CW value is updated as follows:

$$CW_{new}[\mathrm{AC}] = \min\{CW_{\max}[\mathrm{AC}], CW_{old}[\mathrm{AC}] + 1) \times 2 - 1\}$$

Once a STA has won the competition to access the medium, it transmits MSDUs for a period whose maximum depends on the AC and is TXOPlimit[AC]. Therefore the in EDCF a STA is allowed to transmit more than a frame without having to regain access to the channel.

4 SIMULATION SCENARIO

Simulations have been performed with Network Simulator *version2*, which is an open source simulator for different kinds of telecommunication networks.
The simulated scenario is the ad hoc network represented in figure 1. The ad hoc network is composed of 6 wireless stations in fixed positions.
Each STA performs one or more transmissions of real time (voice, videoconference), multimedia (video and audio streaming) and best-effort (data transfer) applications.

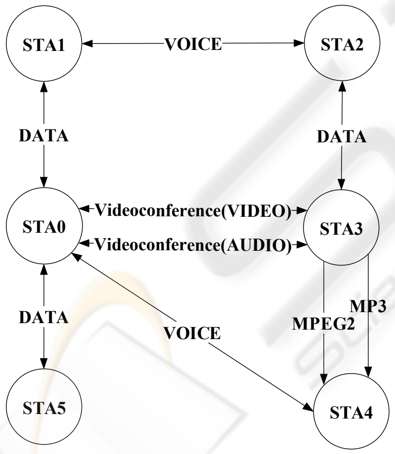

Figure 1: Simulation scenario 1

The simulation time has been divided into three sub-intervals where the different traffic sources start, until the network load reaches the full load.
As shown in figure 2, in the first time interval, voice and videoconference start; in the second time interval, streaming audio and audio-video are added; in the third time interval, the network is loaded with best effort data transfers until the saturation point of the network. The network load is 17% in the first time interval (30 s), 88% in the second time interval (30 s), 104% in the third time interval (30 s).

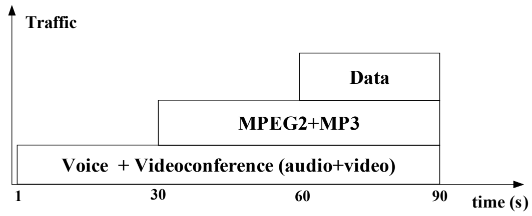

Figure 2: Source timing

For each traffic source the following models have been used.

Voice: bi-directional CBR source at 64 kb/s, packet length 160 bytes, packet interarrival time 20 ms

Videoconference: the traffic is composed of video and audio, and is modelised as follows:
Video: bi-directional CBR source at 98.4 kb/s, packet length 1500 bytes, packet interarrival time 122 ms

Audio: bi-directional CBR source at 6.3 kb/s, packet length 160 bytes, packet interarrival time 203 ms

MPEG-2 video stream: unidirectional CBR source at 4 Mb/s, packet length 1504 bytes, packet interarrival time 3 ms

MP3 audio stream: unidirectional CBR source at 128 kb/s, packet length 512 bytes, packet interarrival time 32 ms

Data traffic: unidirectional CBR source at 128 kb/s, packet length 1500 bytes, packet interarrival time 9.6 ms

The transport layer implemented for simulations is UDP for voice, videoconference, MPEG-2 and MP3, and TCP for data transfer.

Table 2: MAC parameters chosen for DCF and EDCF simulations

DCF and EDCF common values	
SlotTime	20 ms
SIFS	10 ms
PIFS	30 ms
DIFS	50 ms
FragmentationThreshold	2304 byte
RTSThreshold	300 byte
MaxTrasmitMSDULifetime	512*1024 µs
DCF access parameters	
CW_{min}, CW_{max}	31, 1023
EDCF access parameters	
AIFS[3], CW_{min}[3], CW_{max}[3]	2, 7, 31
AIFS[2], CW_{min}[2], CW_{max}[2]	4, 15, 256
AIFS[1], CW_{min}[1], CW_{max}[1]	8, 31, 1023
AIFS[0], CW_{min}[0], CW_{max}[0]	16, 63, 1023

Table 3: traffics and ACs mapping

Priority	**Traffic sources**	**AC**
6-7 (Very High Priority)	Voice, Videoconference (audio and video)	3
4-5 (High Priority)	Streaming audio (MP3) and streaming video (MPEG-2)	2
3 (Excellent Effort)	-	1
0-2 (Best Effort)	Data transfer	0

Table 2 shows MAC parameters chosen for DCF and EDCF simulations. The common values are default values of the IEEE 802.11b physical level; EDCF parameters have been chosen in order to obtain good differentiation among access categories. The RTSThreshold has been chosen such that the RTS/CTS mechanism is not applied to voice and audio of the videoconference traffic.

Table 3 shows the mapping among traffic sources and ACs; table 4 presents physical parameters used for simulations. Carrier sense and Rx ranges have been computed considering antennas height of 1.5 m.

Table 4: PHY parameters chosen for simulations

Definition	**Value**
P_t (dBm)	20
RXThresh (dBm)	-50
CSThresh (dBm)	-56
Propagation model	Two-ray ground
Carrier Sense range (m)	113
Rx range (m)	80
Max Trasmission Rate (Mbit/s)	11
Min Trasmission Rate (Mbit/s)	1

In the simulation scenario, STA1 and STA2, STA3 and STA4, STA5 and STA6 are 50 m away; STA1 and STA3, STA3 and STA5, STA2 and STA4, STA4 and STA6 are 60 m away.

The magnitudes that have been computed for QoS performance evaluation are: throughput, delay, jitter, Packet Loss Ratio (PLR), IP Packet Delay Variation (*IPDV*). The *IPD*V is defined as follows:

$$IPDV = IPTD_{\max} - IPTD_{\min}$$

where $IPTD_{\max}$ and $IPTD_{\min}$ are respectively the maximum and minimum IP Packet Transfer Delay (*IPTD)* measured in the considered interval.

Table 5 shows needed QoS parameters for the described simulation scenario.

Table 5: QoS parameters for the described simulation scenario

Applications	**Data Rate**	**QoS Parameters**		
		Delay end-to-end	**IPDV**	**PLR%**
Voice	64 Kbit/s	<50ms	<50ms	< 5%
Real-time audio (videoconference)	6.3 Kbit/s	<50ms	<50ms	< 3%
Real-time video (videoconference)	98.4 Kbit/s	<50ms	<50ms	< 1%
Streaming audio MP3	128 Kbit/s	<10 s	1 s	< 1%
Streaming video (MPEG-2)	4 Mbit/s	<10 s	1 s	< 1%
Data transfer	128 Kbit/s	<60 s	-	zero

5 SIMULATION RESULTS

Simulations compare ad hoc network performances between DCF and EDCF MAC in terms of throughput, average and instantaneous delay, jitter, *IPDV*.

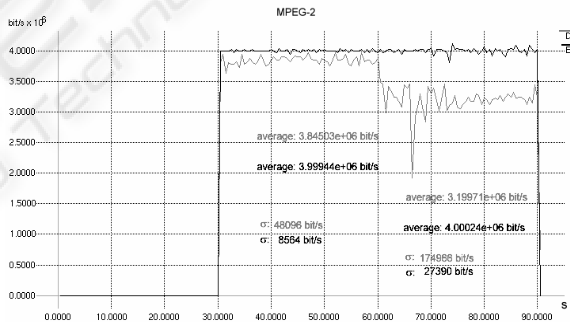

Figure 3: MPEG-2 throughput

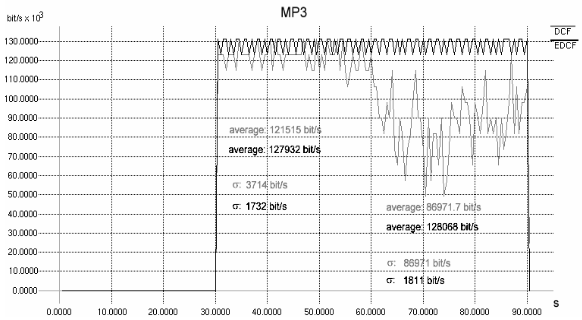

Figure 4: MP3 throughput

Figures 3, 4 and 5 show a comparison of MPEG-2, MP3 and voice instantaneous and average throughputs between DCF and EDCF functions. EDCF maintains good throughputs even at high network loads.

Figures 6 and 7 show the delay of voice and videoconference traffics, and highlight that delay limits are not guarantied with DCF even in the second simulation interval. Table 6 shows the videoconference average delay, jitter, and IPDV with DCF and EDCF functions. EDCF guarantees QoS limits even in the third simulation interval, when the network is high loaded. In tables 7 is shown that the probability that the delay of real time traffic is lower than 50 ms is always higher than 99% with EDCF, and is sometimes lower than 50% with DCF.

Finally, table 8 shows PLR for EDCF and DFC, highlighting EDCF better performances.

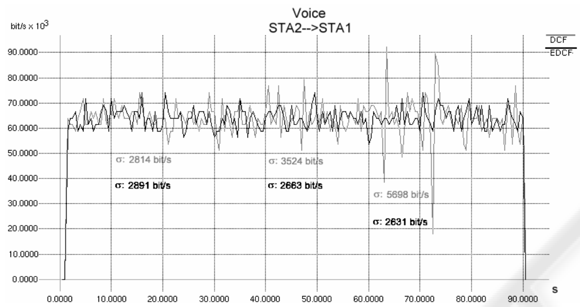

Figure 5: Voice throughput

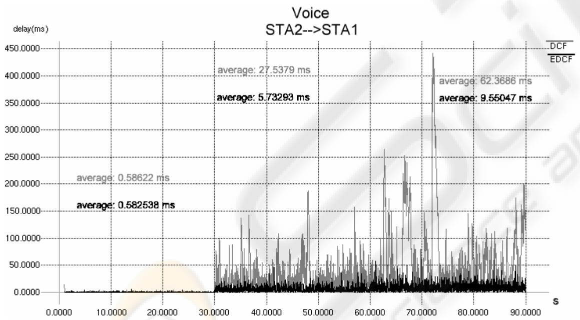

Figure 6: Voice delay

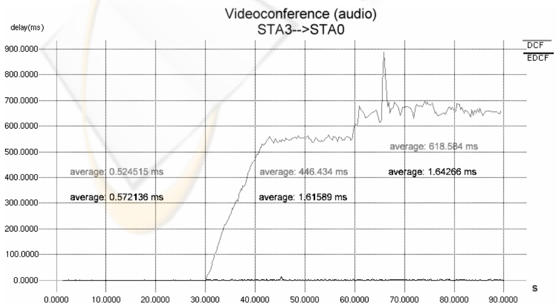

Figure 7: Videoconference (audio) delay

Table 6: Average delay, jitter, IPDV of videoconference traffic

Application	s	MAC	Average delay (ms)	Jitter (ms)	IPDV (ms)
Videoconf. (audio) STA0→STA3	60-90	DCF	6.61	25.8	58.95
		EDCF	1.77	3.48	4.19
Videoconf. (audio) STA3→STA0	30-60	DCF	446.43	46.1	555
		EDCF	1.61	4.47	13.17
	60-90	DCF	618.58	121.	285.7
		EDCF	1.64	3.46	5.89
Videoconf. (video) STA0→STA3	60-90	DCF	6.97	25.2	67.14
		EDCF	3.21	3.94	8.81
Videoconf. (video) STA3→STA0	30-60	DCF	446.12	47.8	559.3
		EDCF	2.93	4.17	8.89
	60-90	DCF	666.73	149.	282.5
		EDCF	3.18	3.92	6.13

Table 7: probability that the delay of real time traffic is lower than 50 ms

Application	**MAC**	**P(delay<50 ms)**
Voice STA1→STA2	DCF	82.34 %
	EDCF	99.88 %
Voice STA0→STA4	DCF	99.64 %
	EDCF	100 %
Voice STA4→STA0	DCF	100 %
	EDCF	100 %
Videoconference (audio) STA0→STA3	DCF	99.77 %
	EDCF	100 %
Videoconference (audio) STA3→STA0	DCF	40.72 %
	EDCF	100 %
Videoconference (video) STA0→STA3	DCF	99.59 %
	EDCF	100 %
Videoconference (video) STA3→STA0	DCF	39.03 %
	EDCF	100 %

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