Studienarbeit

# Research assignment on audio-visual performance of IEEE 802.11e and 802.11n proposals studying Pre-N consumer hardware

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**Abstract**

The goal of this research assignment is to make predictions about the performance with respect to multi-media content of upcoming standards IEEE 802.11e and 802.11n proposed by the IEEE task group TGe and the Enhanced Wireless Consortium[1], focusing mainly on the Quality of Service aspects and limitations with respect to bandwidth of those proposals. The study should contain classifications about maximum achievable transmit rates and service guarantees. Typical Quality of Service and bandwidth scenarios for in-home wireless media distribution will be used to measure the impact of the above proposals. For practical verification, already available and so called Pre-N consumer hardware by Belkin[2] that incorporates MIMO-enabled chipsets by Airgo Networks[3] is used.


[1] EWC `http://www.enhancedwirelessconsortium.org`
[2] Belkin Corporation `www.belkin.com`
[3] Airgo Networks `http://www.airgonetworks.com`

**Eidesstattliche Erklärung:**
Hiermit versichere ich, die vorliegende Arbeit selbstständig und unter ausschließlicher Verwendung der angegebenen Literatur und Hilfsmittel erstellt zu haben.

Die Arbeit wurde bisher in gleicher oder ähnlicher Form keiner anderen Prüfungsbehörde vorgelegt und auch nicht veröffentlicht.

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# Contents

## Glossary

(Roughly in Order of appearance)

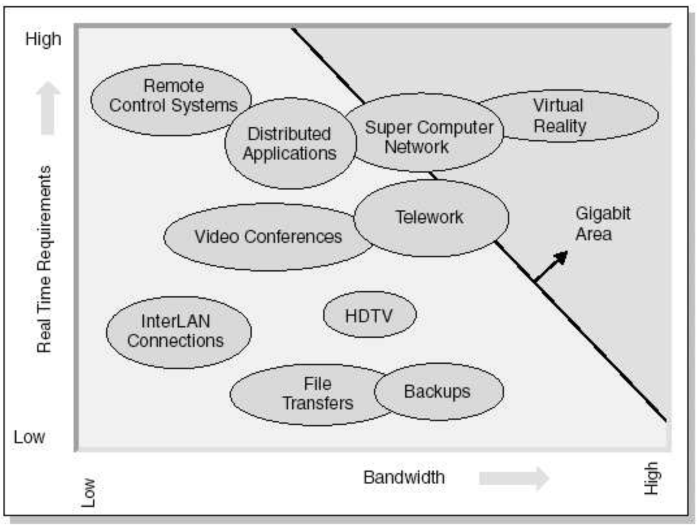
| | |
|---|---|
| QoS/CoS | Quality of Service / Class of Service |
| MIMO | Multiple Input - Multiple Output |
| TGe | IEEE 802.11 Task Group E |
| MAC | Medium Access Control |
| AP | Access Point |
| DCF | Distributed Coordination Function |
| PCF | Point Coordination Function |
| CSMA/CA | Carrier-Sense Multiple Access with Collision Avoidance |
| TXOP | Transmission Opportunity |
| NAV | Network Allocation Vector |
| CP | Contention Period |
| CFP | Contention Free Period |
| TDMA | Time Division Multiple Access |
| DSSS | Direct Sequence Spread Spectrum |
| FHSS | Frequenca Hopping Spread Spectrum |
| OFDM | Orthogonal Frequency Division Multiplex |
| PHY | Physical Layer used in 802.11, either DSSS, FHSS or OFDM |
| MSDU | MAC Service Data Unit |
| HCF | Hybrid Coordination Function |
| EDCA | Enhanced Distributed Channel Access |
| HCCA | Hybrid Coordinated Channel Access |
| UP | User Priority |
| AC | Access Category |
| EDCAF | Enhanced Distributed Channel Access Function |
| HC | Hybrid Controller |
| ARQ | Automatic Repeat Request |
| CBR | Constant Bit Rate |
| NIC | Network Interface Card |

# Chapter 1

# Introduction

With Voice-over-IP or IP-TV targeting the mass-market and being highly demanding applications with respect to the network resources bandwidth and/or latency, the standard wireless network specifications are not sufficient to provide the framework for delivering those formats over a wireless link without loss of quality in whatsoever manner. The means necessary to ensure either a specific bandwidth (e.g. video) or a specific latency (e.g. voice) are missing. Therefore, a new standard is required to enable classification, prioritisation and resource reservation of and for data streams, where one usually distincts the terms "Quality of Service" (QoS) if a certain service quality should be achieved which would be a certain bandwidth, latency or both and "Class of Service" (CoS) when a prioritisation scheme is referred to. The IEEE 802.11 Task Group E (TGe) has made proposals for enabling QoS/CoS[1] in wireless network environments (See Chapter 3 on page 14).

Figure 1.1: Demands of network services - Source: [KC02]




[1] in what follows I will by QoS refer to both if the distinction is irrelevant

Furthermore the bandwidth requirements for a network link are raised steadily with upcoming new standards like HDTV, creating the necessity for new wireless standards defining higher data rates. At the moment there are several proposals for increased data rate in 802.11 networks based on MIMO, a technique that utilises multiple sending and receiving antennas, for an upcoming standard 802.11n (see Chapter 5 on page 22).

The document is structured as follows: First a brief introduction to 802.11 medium access control is given, depicting the impossibility to ensure Quality of Service in the base wireless standard. In the following chapter, the QoS proposals of TGe are introduced and their functionality explained. This is followed by a short summary of MIMO respectively 802.11n. A short chapter about point-to-multipoint transmission should show the possibilities of multicasting and broadcasting for wireless networks and audio/visual content distribution. The expectations on the various access schemes, proposals and other features is then followed by a chapter about the performance of the Pre-N devices and how the performance can be maximised. Then a typical QoS scenario is elaborated the performance of the 802.11n and 802.11e proposals is simulated in this scenario with ns2 and verified with the Pre-N hardware. Finally a chapter about bandwidth requirements of various audio/visual formats is given and a conclusion is drawn from the end-users perspective about the performance of the standards versus the devices implementing those in a typical in-home scenario.

# Chapter 2

# Genuine 802.11 medium access control

In the IEEE 802.11 standard [80299] finished in 1999, two channel access methods were defined: *distributed coordination* in the form of the DCF and *point coordination* in the form of the PCF. The former implementing a CSMA/CA multiple access scheme where each[1] station senses the current power on the channel and transmitting only if the power is below a certain threshold[2]. Thus multiple access interference is avoided in a decentralised manner. PCF on the other hand defines a controlled channel access scheme where a point coordinator allocates a time slot for each station upon request.

## 2.1 DCF - Distributed coordination

The DCF is the basic channel access function in 802.11 networks and it is mandatory for all implementations. As stated above, the DCF allows sharing of the wireless medium through the use of CSMA/CA.

Carrier sense is done by physical and virtual means. Before an attempt to transmit is made, the station physically senses the medium and the channel being idle for a certain amount of time. The amount of time varies depending on the frame type (control, data,...) about to be transmitted but generally in DCF this period is the *distributed inter-frame space* (DIFS) for data frames.

In addition to the DIFS the stations that have a transmission pending draw a random number from a specified interval, the *contention window* (CW) and start to count down. This *back-off time* determines the deferral of the transmission at each station, with the station that chooses the smallest number being able to transmit. Other stations that sense the initiated transmission stop their countdown until the transmission is over. This algorithm helps avoiding collisions and grants every station statistically the same probability of obtaining a *transmission opportunity* (TXOP).

To avoid collisions a virtual carrier sense mechanism is used. Each frame that is transmitted includes a value that specifies the duration of the transmis-

[1]Therein the property "distribution" of the DCF model is justified

[2]The channel is assumed "free" if the power is below this threshold

sion. All stations maintain a *network allocation vector* (NAV) internally and update it according to the duration values received from other stations. Before transmitting itself, a station will wait until the NAV is counted down to zero making sure that all other transmissions have finished.

The standard denotes the time-window where this competition for transmission among the stations occurs the *contention period* (CP). The duration is specified by the AP by periodically repeated beacons between which the contention period lasts and it is optionally preceded by the *contention-free period* (CFP). (See Figure 3.2 on page 15)

## 2.2 PCF - Controlled access

In the contention-free period, the PCF provides for a deterministic transfer mode by means of true TDMA. A *point coordinator* (AP) schedules transmissions for each associated station by a polling mechanism and thus cares for collision avoidance. But being optional and more complex in its implementation, the PCF-mode is not widely used. [God03]

With the DCF providing for fairness among the stations and transmission in a "best-effort" way, prioritising traffic flows is virtually not possible since the transmission is non-deterministic. However, the standard explicitly states the possibility of MSDU reordering making it possible to optimise traffic flow by prioritising for example TCP-acknowledgements or to even elaborate more complex prioritisation schemes. The effect would yet be limited because a TXOP still has to be won, inhibiting service guarantees. On the other hand, with the point-coordinated transmission scheme it would be possible to extend an AP by QoS as well as CoS mechanisms. Still the increased implementation effort for PCF and a missing standardisation prevent the application of such mechanisms.

## 2.3 Optional mechanisms

### RTS/CTS

The request-to-send/clear-to-send handshaking mechanism improves medium usage the when hidden-node problem occurs. A hidden-node is a non-AP stations in a wireless LAN that can not be "seen" by all other stations. Stations that do not "see" each other can not sense each other's medium access. Thus the CSMA/CA scheme becomes more and more ineffective with increasing hidden-nodes. To circumvent this problem the RTS/CTS mechanism can be used. This way every station requests to access the medium prior to transmission and the AP can coordinate the medium access, thus reducing the collisions. In a scenario where no hidden nodes are to be expected, as it is the case here, the RTS/CTS mechanism does only decrease the overall performance.

### CTS-to-self

As 802.11b devices use a FHSS PHY while 802.11g uses OFDM, 11b devices can not detect OFDM transmissions and thus a protection mechanism is necessary if 802.11b and 802.11g devices participate in the same wireless LAN. Therefore 802.11g devices protect the fast 802.11g frame with a preceding slow CTS frame, thereby prohibiting medium access for other devices and especially also 802.11b devices which would otherwise interpret the 802.11g medium access as noise. The overall throughput is thus lowered if 802.11b and 802.11g devices are intermixed because the CTS frame needs to be transmitted with its 802.11b headers and at a low speed before 802.11g every date frame exchange resulting in a relatively large overhead.

Note that this is the minimum required protection mechanism defined in the 802.11g standard in the presence of 802.11b devices. Another protection mechanism would be the use of RTS/CTS being necessary in the presence of hidden-nodes.

# Chapter 3

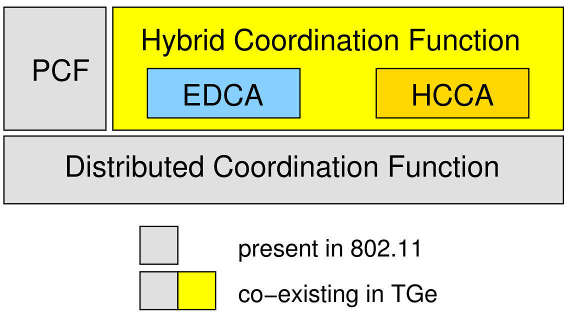
# The TGe proposals for Quality of Service

## 3.1 Enhanced MAC functionality - a hybrid approach

On the basis of the Distributed Coordination Function mainly used in 802.11, the Quality of Service enhancements proposed by Task Group E [TGe05] include the *Hybrid Coordination Function* (HCF) and the *Enhanced Distributed Coordination Function* (EDCF) that are only used in QoS enabled networks. This *Hybrid Coordinated Channel Access* (HCCA) combines functionality of the DCF and PCF and adds several enhanced mechanisms and frame types for QoS networks. With the PCF rarely being used in consumer hardware, it is sufficient to state that the HCF and DCF are present in a QoS-enabled station while in a non-QoS-enabled station only the DCF is present. The MAC architecture is depicted in Figure 3.1.

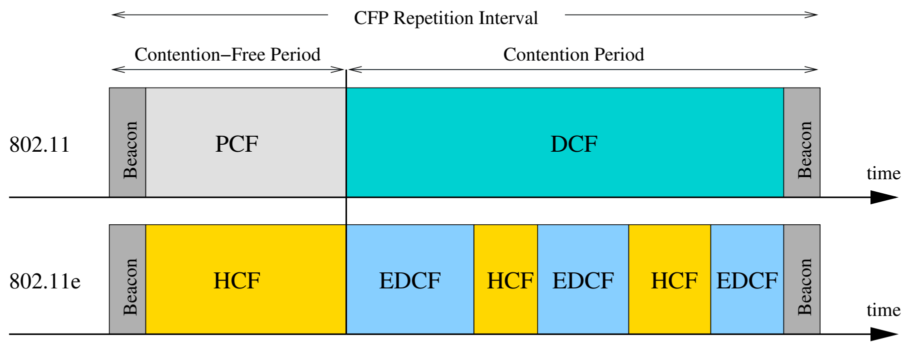
The HCCA adopts the channel access methods *distributed coordination* and *point coordination* with the HCCA being able to serve as both. Thus QoS-enabled data transfer is possible during the CP and the CFP by means of the EDCF and the HCF. Yet the HCCA allows the hybrid coordinator to create

Figure 3.1: MAC Architecture of 802.11 with TGe extension



*controlled access periods* also during a CP which enables the use of the HCF in both the CFP and CP if it is decided to be necessary by the AP. This model versus the legacy 802.11 model is depicted in Figure 3.2.

Figure 3.2: CFP Repetition Interval of 802.11 and 802.11e




This results in the TXOP being defined in two ways

- EDCF TXOP starts when the wireless medium is determined to be idle under the rules of the EDCF. The length of the EDCF TXOP is specified by the AP in the beacon frames.
- HCF TXOP is a polled TXOP that is granted by the HC to a station by a CF-Poll frame in which also the length of the HCF TXOP is specified.

### 3.1.1 EDCA - enhanced contention-based channel access

The *enhanced distributed channel access* is based on the distributed channel access from 802.11 with QoS enhancements. It provides a "best-effort" transmission scheme with prioritised access to the wireless medium. Packets are classified to have a certain *user priority* (UP), which is the same as defined in 802.1D, and are mapped accordingly to one of four *access categories* (AC) as depicted in Table 3.1.

When data to be transmitted arrives at the station, the MAC associates each MSDU to an AC depending on its UP signalled in the corresponding Ethernet frame. Each AC is associated with its own transmit queue. At the head of each queue, multiple MSDUs contend internally for the TXOP as every queue has its own *enhanced distributed channel access function* (EDCAF). The winner then again contends for the TXOP on the wireless medium utilising the same functionality. The model is depicted in Figure 3.3 on the next page.

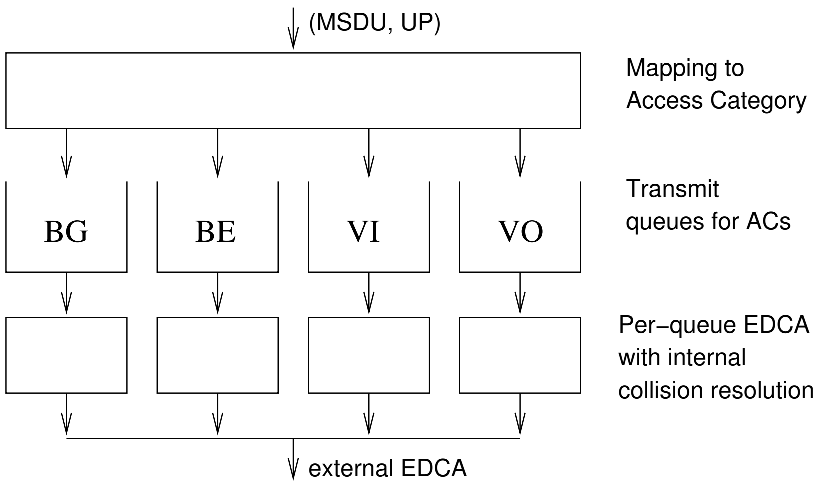
**EDCAF** both the internal and external collision resolution grant the TXOP consecutively to the MSDU that is associated with the smallest randomly chosen delay by means of the same mechanism described by the EDCAF.

The *arbitration inter-frame space* (AIFS) is the analogon to the DIFS in DCF where the AIFS is not a constant but a distinct value depending on the

Table 3.1: Priority to Access Category mappings from [TGe05]

| Priority | User priority (Same as in 802.1D) | Access Category (Informative) |
|---|---|---|
| lowest | 1 | Background |
| | 2 | Background |
| | 0 | Best Effort |
| | 3 | Best Effort |
| | 4 | Video |
| | 5 | Video |
| | 6 | Voice |
| highest | 7 | Voice |

Figure 3.3: Reference Implementation Model adopted from [TGe05]



AC (denoted AIFS[AC]). Also the contention-window ranging from aCWmin to aCWmax from which the back-off-time is chosen randomly is depending on the AC. This leaves the delay produced by the EDCAF to be the combined AIFS and back-off-time in the external case. As in DCF, the station that chooses the smallest combined delay wins the TXOP.

Table 3.2: AIFS[AC] and CW[AC] with $aCW[min, max] = 2^x - 1$

| Access Category | AIFSN | CWmin | CWmax |
|---|---|---|---|
| Bulk/Background | 7 | aCWmin | aCWmax |
| Best-Effort | 3 | aCWmin | aCWmax |
| Video | 2 | (aCWmin+1)/2-1 | aCWmin |
| Voice | 2 | (aCWmin+1)/4-1 | (aCWmin+1)/2-1 |

The EDCA mechanism is designed to provide a certain Class of Service rather than Quality of Service since it only controls transmission probabilities depending on a priorisation scheme. A certain latency or bandwidth is neither adjustable nor reserve-able since the EDCA mechanism is still based on the distributed coordination. In EDCA-mode, the QoS station simply tries to transmit earlier than other QoS stations (AIFS[AC]) and non-QoS stations (DIFS) if it has a MSDU with high priority pending - and vice versa - resulting in transmission probabilities that reflect the AC.

### 3.1.2 HCCA - enhanced contention-free channel access

In *hybrid coordinated channel access* a *hybrid controller* (HC) uses a polling mechanism similar to PCF to assign TXOPs to the attached stations. The priority by which the HC selects a station for a TXOP is determined by the *traffic specification* (TSPEC) the station has registered for. This TSPEC determines a certain bandwidth and/or latency a service requires.

## 3.2 MAC frame format

TGe introduces a Quality of Service Control field to the MAC frame as depicted in figure 3.4 on the following page. The QoS Control field is a 2 byte field that is mandatory for every QoS enabled station. It includes a subfield for the ACK policy that the sender of this frame expects from the receiving end. Also the priority bit for EDCF (TID) and the TXOP duration, either as a request from a non-AP station or a limit from the HC, is included in the QoS Control field. The whole functionality is depicted in table 3.3 on the next page. .

Figure 3.4: 802.11e MAC frame format

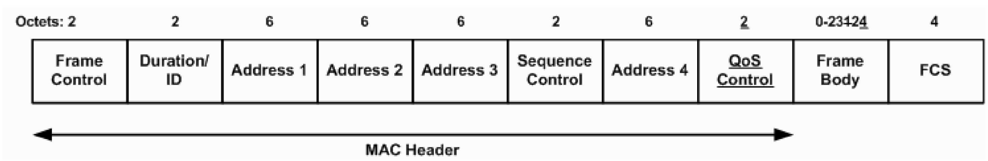



Table 3.3: QoS Control field

| Applicable Frame (sub) Types | Bits 0-3 | Bit 4 | Bit 5-6 | Bit 7 | Bits 8-15 |
|---|---|---|---|---|---|
| QoS (+)CF-Poll frames sent by HC | TID | EOSP | Ack Policy | Reserved | TXOP limit |
| QoS Data, QoS Null, and QoS Data+CF-Ack frames sent by HC | TID | EOSP | Ack Policy | Reserved | QAP PS Buffer State |
| QoS data type frames sent by non-AP QSTAs | TID | 0 | Ack Policy | Reserved | TXOP duration requested |
| | TID | 1 | Ack Policy | Reserved | Queue size |

## 3.3 Optional performance improving parameters

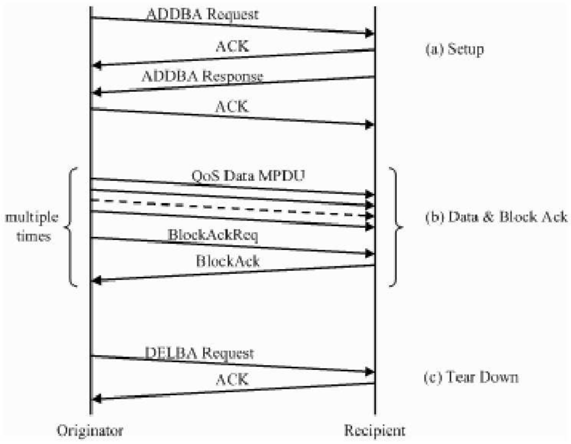
### 3.3.1 Block acknowledgement

Block-ACK (also referred to as “Burst-ACK”) can be requested by the sender to have multiple MSDUs acknowledged at a time as opposed to the immediate acknowledgement used in previous 802.11 standards. This reduces the overhead of the single acknowledgement per MSDU method. Note that this is not the same as *contention-free bursts* described in section 3.3.4 on the facing page. The Block acknowledgement model is depicted in figure 3.5 on the next page.

### 3.3.2 No acknowledgement

The No-ACK switch reduces the acknowledgement overhead even further by completely abandoning the ARQ error correction of the MAC protocol. The resulting throughput and latency is increased but a loss of control is introduced. This leads to unrecoverable errors at the receiver if a MSDU is not correctly received, which affects the higher layer protocols. UDP transport streams would suffer unrecoverable packet losses but this is to be expected since unreliability is characteristic for UDP whereas TCP can recover a packet loss since ARQ is a native aspect of TCP. For sporadic losses the effect would be limited since the behaviour is comparable to packet loss on a saturated Ethernet where a packet can be discarded if the number of retransmission attempts exceed a certain value.

Figure 3.5: Block/Burst Acknowledgement (source: [TGe05])




### 3.3.3 Direct-link protocol

The direct-link protocol enhancement allows stations to exchange data frames between each other directly, bypassing the AP. Without this feature packets travelling from a source station A to a sink station B would be sent from station A to the AP and after contending for the medium again from the AP to station B. Therefore, as it is the case for the standard 802.11 MAC, such packets are on the air twice. Thus the transmission between multiple stations is also referred to as multi-hop, where the expectable data rate is the single-hop data rate divided by the number of hops. With the direct-link protocol enabled the station-to-station throughput is expected to increase up to a maximum that is achievable in a single-hop station-to-AP path.

### 3.3.4 Contention-Free bursts

Although being not specifically defined in the 802.11e draft standard the possibility of contention-free bursting is an option mentioned in [God03] cohesive with TGe. As an optional method to reduce contention a station or AP with a TXOP, either received by the EDCF mechanism or polled by the HC, is able to transmit more than one MSDU if it still has time left in the TXOP to transmit at least one additional MSDU. In the legacy 802.11 MAC it was mandatory for the station to contend for the wireless medium again to gain another TXOP. Thus a station should, if possible, reduce the contention and transmit multiple frames in one TXOP. The frames would then be acknowledged with a single immediate ACK. Unlike the block acknowledgement a *contention-free burst* can also be implemented in an immediate ACK model where during one medium access more than one MSDU is transmitted while during a block acknowledgement frame exchange sequence other stations could also access the wireless medium.

# Chapter 4

# Quality of Service signalling

For signalling the Class as well as a Quality of Service[1] various approaches are possible. The following are some of those possibilities which are either related to 802.11e or easy to implement. Table 4.1 shows on which OSI layers and network protocols such approaches would hold.

- Signalling on the MAC layer with TSPEC frames

  A TSPEC is sent to the Hybrid Coordinator to reserve a resource for a certain stream. A protocol like RSVP can be used at the application layer to signal certain resource needs to the Hybrid Coordinator. Details on this are not subject of this research assignment as the HCCA mode is not mandatory for 802.11e stations and it is as of today not implemented in available hardware.

- Signalling on the MAC layer on a per-frame basis

  The MAC frame header contains a field that identifies each packet's priority class. This form of CoS signalling which is the signalling method for EDCF based transmissions is mandatory for 802.11e. By this signalling method the Ethernet equivalence is achieved since the 802.1D priority field of an Ethernet frame can be copied to a corresponding 802.11e MAC frame. Thus e.g. dedicated hardware like VoIP telephones signal their desired packet priority on the Ethernet frames they generate and the priorities are maintained if the link (VoIP-phone to VoIP-phone) includes a wireless LAN.

- Signalling on the application layer

  A higher layer protocol like RTP is used to specify priorities on a per-frame basis. This can be regarded as the most easy way for an application to indicate that a higher or lower than usual priority is required.

- Assuming UDP is for media, TCP for bulk

  This assumption is the simplest form of adding CoS to a wireless link as the firmware of the CoS-enabled station needs only to distinguish between

[1] Which is also referred to as "admission control"

Table 4.1: Network layers overview

| | OSI layers | Protocols | QoS options |
|---|---|---|---|
| 7 | Application | | |
| 6 | Presentation | HTTP, FTP, etc. | TSPEC (HCCA) |
| 5 | Session | | |
| 4 | Transport | TCP, UDP | UDP media assumption (EDCA) |
| 3 | Network | IP, ICMP, IPX | - |
| 2 | Data Link | Ethernet, 802.11 MAC | 802.1D (EDCA) |
| 1 | Physical | DSSS, OFDM | - |

the two packet types and assign a priority level accordingly, where UDP would get a higher priority/transmit probability than TCP. Though being most simple, the assumption is not far from the truth since media distribution focuses mainly on the UDP protocol where important features like multicast transmissions are possible whereas TCP only provides for point-to-point transmissions, thus increasing traffic requirements for broadcast systems unnecessarily.

# Chapter 5

# 802.11n

## 5.1 MIMO

MIMO uses multiple transmit and receive antennas to increase bandwidth and range. While multi-path spreading was considered a problem in previous wireless systems, MIMO now gathers the echoes introduced by multi-path spreading to increase the signal-to-noise ratio. The data rate is increased by using multiple transmit antennas to create spatially separated channels onto which the signal is multiplexed. The number of separate channels $n$ that can thus be achieved is given by

$$n = min(N_T, N_R)$$

where $N_T$ is the number of transmit antennas and $N_R$ is the number of receive antennas.

### Pre-N

The so called "Pre-N" wireless modems by Belkin are using two antennas for transmission and three antennas for reception. Thus the expected (and advertised) data rate is

$$min(2, 3) \cdot datarate(802.11g) = 2 \cdot 54Mbps = 108Mbps$$

## 5.2 MAC enhancements

**Ethernet frame aggregation** Increasing the payload size of a MAC frame from 2304 bytes to over 8000 bytes would enable a wireless station to aggregate multiple Ethernet frames (1500 bytes max) and send them during one TXOP. This would reduce acknowledgement overhead significantly.

**MAC frame aggregation** Another proposal for frame aggregation is sending multiple MAC frames within one PHY frame (after a single PLCP).

**Block-acknowledgement** Similar to TGe the proposals for 802.11n include the use of a block-ACK mechanism instead of immediate acknowledgement.

# Chapter 6

# Point-to-multipoint transmission

With respect to classical forms of media distribution like standard TV or radio, broadcast is a natural property of distributing audio-visual content - a sending station transmits a signal to multiple clients. In modern IP-based communication systems like the Internet the possibilities of broadcasting services are limited. The nature of the IP protocol is a point-to-point transmission scheme, be it connection-oriented or connection-less. Broadcasts are available for the connection-less UDP protocol but these broadcasts usually do not traverse routers or gateways as they would otherwise have a flooding effect on the entire net. Multicast is also available for UDP and there are is a special address range (224.0.0.0 to 239.255.255.255) reserved for so-called multicast groups. Limited support for multicast in Internet routers used all over the world prohibits a multicast-enabled world-wide-web but for a local area network the above limitations do not hold. Therefore the multicast ability of local area networks is an interesting factor for audio-visual content distribution.

## 6.1 Multicast and broadcast in 802.11 networks

The 802.11 standard defines special frame types and mechanisms suitable to enable efficient broadcast or multicast transmissions. For unicast transmissions every station can send a packet to a certain receiver (via the AP) and expects an acknowledgement (Immediate-ACK) in return from the AP. On the other hand an AP can broadcast a frame with broadcast/multicast flags enabled to its associated stations who will (hopefully) receive the packet correctly without acknowledging the reception to the AP. As it is not guaranteed that every station "sees" every other station broadcasts can only be made by a station which is an AP that "sees" all associated stations. Therefore a point-to-multipoint source on the Ethernet behind an AP can expect the broadcasted data to arrive at each station at the same time and with the delay of one medium access phase by the AP without acknowledgement. Such a source residing at a non-AP station can also expect the data to arrive at all other stations at the same time but with the delay of two consecutive medium access phases since the frame has to be transmitted to the AP (with acknowledgement) and then from the AP to all associated stations by a broadcast (not acknowledged).

## 6.2 Power-save queueing problem

With broadcasts arises a problem if power-saving enabled clients are participating in a wireless network. The AP can not distinguish to which associated stations a multicast (and broadcast, of course) packet has to be sent since it does by default not know about the IP addresses or multicast groups the stations have registered for. Therefore it needs to transmit a broadcast/multicast packet to every associated station. A MAC frame incorporates a flag to indicate that this frame is destined to all associated stations - and this flag can only be set by an AP station because the AP by default sees every associated station which does not always hold for non-AP stations. If a station now enables power-saving it will only wake up if it has a packet pending. In the presence of broadcast traffic it would thus be always up which is not desired. To circumvent the problem an AP needs to buffer broadcast packets for power-saving stations - and in the presence of power-saving stations an AP usually does the buffering for every station. Thus the broadcast data rate will be severely affected in this case.

# Chapter 7

# Performance expectations

## 7.1 DCF

**Transmission probability**

Assumptions:

- Saturation

  Every considered station has enough data frames pending so that the saturated condition holds.

- High amount of transmitted frames

  Therefore the effect of the back-off-freeze, i.e. the hold of the count-down when a transmission is sensed during the back-off phase, does not influence the statistical properties of the system.

- No hidden stations

  Every station shall sense a channel access of any other station so the hidden-terminal problem is neglected. Therefore the RTS/CTS algorithm is not required and shall not be covered here since it would only degrade performance in this case.

A station $S$ that has a transmission pending is to be observed. The probability that the same time-slot is picked by at least one other station with a total of $N$ stations (that have transmissions pending) can be denoted as the collision probability $\beta$ at station $S$ which is equal to the probability that the time-slot picket by $S$ is not picked by no other station

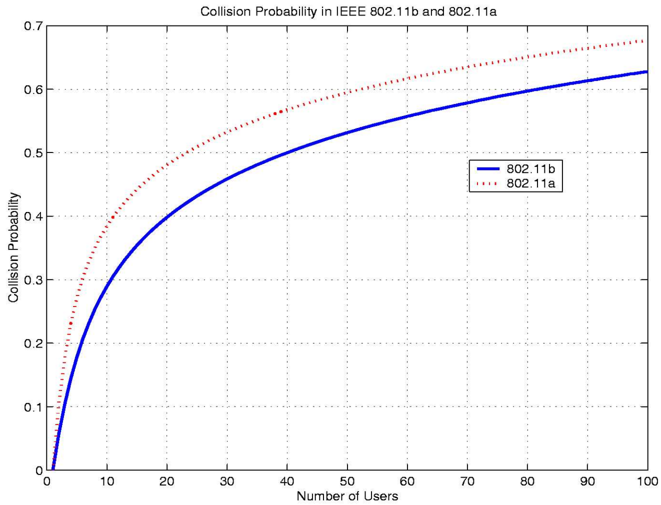
$$\beta = 1 - \left(1 - \frac{1}{\alpha}\right)^{N-1} \tag{7.1}$$

where $\alpha$ is the probability that a station transmits in a randomly chosen slot

$$\alpha = \frac{2(1-2\beta)}{aCWmin(1-2\beta) + \beta(aCWmin+1)(1-(2\beta)^7} \tag{7.2}$$

as it is derived by Bianchi in [Bia00].

Figure 7.1: Collision Probability in 802.11b and 802.11a. Source [KM05]




As defined in [80299] the back-off-time is randomly chosen from the interval $[0, CW] \subset \mathbb{N}$ in case the medium is non-idle for the amount of time specified by the DIFS. The contention window CW is upper bounded by $aCWmax$ (1023 for 802.11b) and lower bounded by $aCWmin$ (31 for 802.11b). Upon collision a retry attempt is made with the CW doubled in size. Thus CW can be defined as

$$CW = 2^{4+i} - 1$$

where $i$ is the back-off stage and $i-1$ is the number of consecutive collisions. Thus $max(i) = 7$ is the number of back-off stages or maximum retransmission attempts.

As an approximation it is assumed that the number of stations contending for the medium is low and thus the number of retransmissions and accordingly the collision probability is low. One can elaborating a trade-off between neglected retransmissions and expectedly chosen slot and introduce a fixed contention window slot $e_{slot}$ depending on the number of stations and the protocol used. For five to ten stations it is sufficient to do this linear approximation as can be seen when comparing figure 7.1 from [KM05] with figure 7.2.

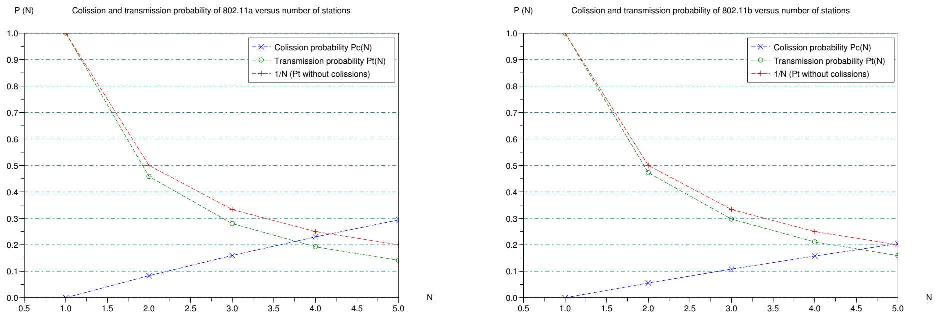
In figure 7.1 the collision probability for 802.11a and 802.11b has been numerically computed solving equations 7.1 and 7.2. With a linear approximation the collision probability in equation 7.1 can be simplified to the approximated collision probability

$$p_c = 1 - \left(1 - \frac{1}{e_{slot}}\right)^{N-1}$$

and thus the transmission probability can be approximated to

$$p_t = \frac{1}{N} \cdot p_c$$

Figure 7.2: DCF transmission probabilities for few stations




The resulting probabilities are depicted in figure 7.2. The values chosen for an approximated expected time slot are

| Protocol | aCWmin | eSlot |
|---|---|---|
| 802.11b | 31 | 18 |
| 802.11g | 31 | 18 |
| 802.11a | 15 | 12 |
| 802.11n | 15 | 12 |

**Latency**

Assumptions:

- Beacon and control frames

  Since those short frames occur rarely compared to data frames their impact on performance is negligible

The one-way latency of a wireless link is defined by a minimum delay $d_{MAC}$that is generated by MAC layer synchronisation and distributed coordination issues

$$d_{MAC} = PLCP + IFS + d_{CW}$$

where $PLCP$ is the preamble, $IFS$ the inter-frame-spacing for the corresponding frame type and $d_{CW}$ the delay produced by the back-off mechanism, measured in (number of slots) $\times$ (slot time) of the corresponding protocol (11b, 11g, ...). Because $d_{CW}$is a randomly picked slot depending on the size of the contention window and thus on the number of collisions that have occurred, $d_{CW}$is defined as the expectation value for the slot multiplied with the slot-time of the protocol

$$d_{CW} = \frac{e_{slot} + 1}{2} \cdot t_{slot}$$

The overall delay $d_{min}$ produced by the MAC layer is the MAC layer delay preceding a data frame plus the MAC layer delay of the corresponding ACK frame, where no contention occurs.

$$d_{min} = PLCP + DIFS + d_{CW} + PLCP + SIFS$$

The delay produced by the actual transmission of the data frame $d_{TX}$is the amount of data divided by the data rate of the used protocol and the overall delay is the MAC delay plus the transmission delay.

$$d_{TX} = \frac{FrameSize + AckFrameSize}{DataRate}$$

$$d_{tot} = d_{TX} + d_{min}$$

**Throughput**

The raw throughput, neglecting higher layer protocol overhead, is depending on the number of stations that are transmitting, the used data rate (per 802.11 protocol) and payload per frame. The protocol data rate defines the time $d_{tot}$ that is needed to transmit a packet and receive a returning ACK, as defined in the latency subsection where the payload defines the parameter $FrameSize$.

$$FrameSize = ProtocolOverhead + Payload$$

$$t = p_t \cdot \frac{Payload}{d_{tot}}$$

## 7.2 EDCF

**Transmission probability**

Assumptions:

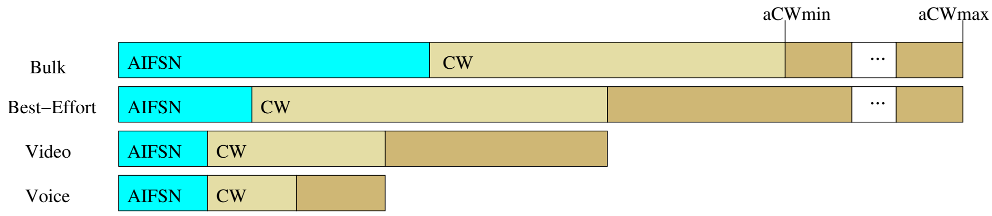
- Though the standard states that frames should contend internally for a TXOP by means of the EDCA mechanism, there shall be no internal collision that prevents a transmission.

In EDCF every AC results in a different transmission probability dependent on the AC of other frames being pending for transmission. Table 3.2 on page 17 shows that the time windows where the contention occurs are not the same for each AC but mutually overlapping with different sizes and offsets. This is also depicted in figure 7.3 on the facing page.

### 7.2.1 DCF-EDCF comparison

As derived in section 7.1 and 7.2 the DCF channel access provides the same transmit probability for every station and thus for every data frame about to be sent. Therefore packet priorisation on the MAC layer is not possible.

Figure 7.3: Contention and access categories




EDCF now provides for a priorisation scheme enabling different transmission probabilities for different access categories. When high priority MAC frames contend for the wireless medium they are likely to win the TXOP against lower priority MAC frames since the inter frame spacing (being the start-delay of the contention mechanism) is smaller and the contention window itself is scaled down for higher priority frames, as depicted in figure 7.3, and thus yielding a higher transmission probability. Therefore one has to differentiate which AC is to be observed and against which AC it contends. Additionally the variable IFS and CW for EDCF frames influences the latency and throughput of data streams under EDCF.

## 7.3 MIMO / Pre-N

With MIMO one can assume that the number of transmit antennas defines the data rate

$$R_{MIMO} = n * R_{11g}$$

where $R_{11g}$ denotes the data rate of 802.11g (54Mbit/s) and $n$denotes the number of antennas.

## 7.4 Multi-hop/direct-link

The multi-hop performance of 802.11e enabled devices is expected to not decrease if the direct-link protocol is implemented. Otherwise the data rate is expected to be

$$r_{MH} = \frac{r_{SH}}{n_{hops}}$$

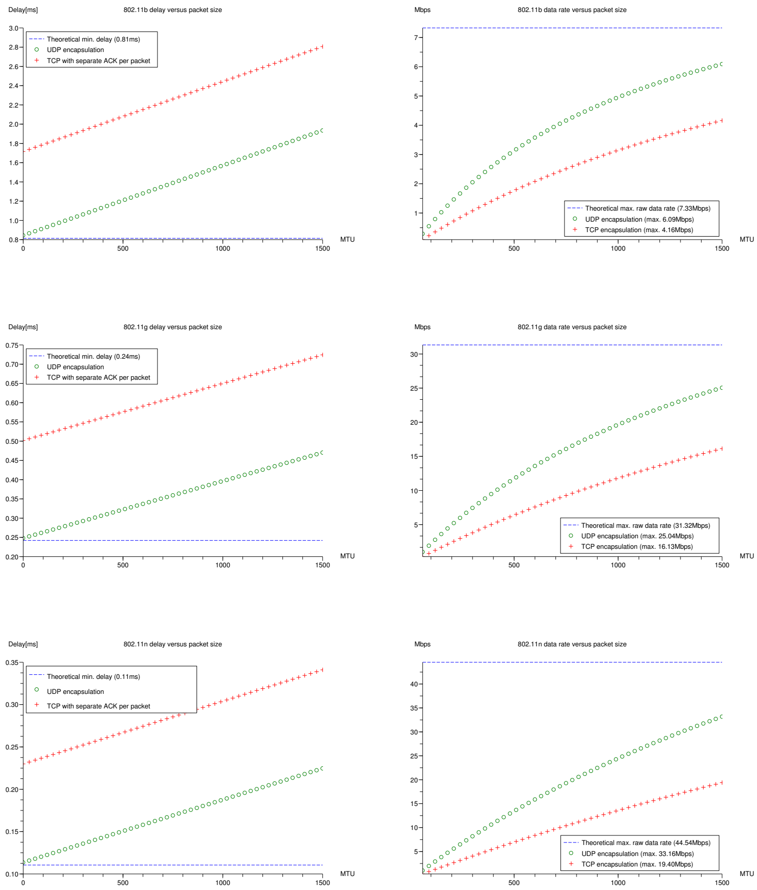
where $r_{MH}$ is the multi-hop data rate, $r_{SH}$is the single-hop data rate and $n_{hops}$is the number of hops. If $n_{hops} = 2$ and direct-link is disabled, the resulting data rate should thus be

$$r_{MH} = \frac{r_{SH}}{2}$$

while if direct-link is enabled the resulting data rate should be for $n_{hops} = 2$

$$r_{MH} = r_{SH}$$

Figure 7.4: Latency and data rate of modes B, G and N




## 7.5 Point-to-multipoint

The point-to-multipoint transmission is as described in chapter 6 a native feature of 802.11. Therefore I expect no performance degradation with broadcast or multicast traffic compared to unicast (point-to-point) performance in a single-hop scenario. A broadcaster at a non-AP station will suffer from a multi-hop degradation since only the AP can broadcast, as described in chapter 6.

## 7.6 Expected latency and data rate

The expected delay and data rate for the various 802.11 modes are depicted in figure 7.4. It is visible that the throughput increases with increasing packet size. The maximum packet size is the Ethernet MTU of 1500 bytes including all headers.

# Chapter 8

# Pre-N performance

The performances are measured in a scenario with two Pre-N stations, one of which is an AP and the other one a non-AP station being the "client" in what follows. The used software/firmware/drivers are the ones that were shipped with the hardware. For the measurements a station sends packets over TCP or UDP[1] for 30 seconds. The packet size is 1500 bytes which is the maximum Ethernet MTU, thus the data rate is maximised. For this task the tool iperf[2] was used. The resulting data rate and latency is measured and the experiment is repeated several times. The focus in this paper is on the data rate aspects - so we are not interested in latencies here. For completeness they may be recorded. The parameter that is to be evaluated and tuned is the achievable throughput (usable data rate). Since providing all performance readings would be too extensive only averaged and reasonable[3] results are provided. The interference level from other wireless-LAN transmitters is negligible[4] as the overall wireless network density is low.

## 8.1 Ideal conditions (LOS)

The client and the AP are co-located in a room where at least some echoes are to be expected. The transmission is therefore not under free-space conditions but a line-of-sight (LOS) exists and the distance between both is 2.5m with no obstacles in the midst. The scenario can be regarded as nearly ideal.

### 8.1.1 Reference DCF performance

The results below were measured with "QoS-enable" set to "no" at the AP's web interface. When comparing the results to the above derived reference values of a standard 802.11g link it seems that this option turns off every performance enhancement including MIMO. The values can therefore be taken as reference 802.11g DCF performance.

[1] For UDP packets are sent at a CBR of 100Mbit/s

[2] http://dast.nlanr.net/Projects/Iperf/

[3] Reasonable in a way that extreme outliers are discarded.

[4] no other AP on the same or an overlapping channel in the vicinity

| Protocol | Rate in Mbit/s ($M = 10^6$) |
|---|---|
| UDP | 28.39 |
| TCP | 23.97 |

### 8.1.2 EDCF performance

#### ACK-Methods comparison with TCP

The throughput is measured in both directions: From the client to the AP and vice versa. In the ideal case the results are expected to be identical except for measurement uncertainties. In the following a table is provided with measurement results and conclusions and observations below.

| | | From client to AP | From AP to client |
|---|---|---|---|
| ACK at AP | ACK at client | Average data rate in Mbit/s ($M = 10^6$) | |
| Immediate | Immediate | 34.76 | 43.34 |
| Immediate | Burst-ACK | 41.44 | 46.22 |
| Immediate | No-ACK | 3.36 | 33.73 |
| Burst-ACK | Immediate | 39.62 | 47.47 |
| Burst-ACK | Burst-ACK | 39.62 | 45.69 |
| Burst-ACK | No-ACK | 3.47 | 33.05 |
| No-ACK | Immediate | 39.60 | 42.21 |
| No-ACK | Burst-ACK | 41.34 | 40.37 |
| No-ACK | No-ACK | 2.36 | 18.43 |

- No-ACK at the sending station reduces TCP performance - severely if client sends. This seems to be a bug in the driver since the effect is not as severe as when the AP is sending. This can be observed in every case where No-ACK is used a the client

- Aside from the No-ACK problems at the non-AP station it is intuitively correct and clearly visible that the ACK method at the sender defines the throughput

- No-ACK is not a preferred option since even under ideal conditions the TCP performance is degraded. This effect will further increase in spatially non-ideal environments.

- Burst-ACK on both sides is the preferred ACK method - data rate is highest with maintained error correction. Although it seems that the combination “burst - immediate” with the latter at the sending end results in the highest throughput this is very likely due to measurement uncertainties or driver issues. Additionally one has to take into account that in this ideal case there is nearly no traffic flowing in the respective opposite direction, so Burst-ACK should give the best performance under non-ideal circumstances.

### UDP throughput measurement

UDP throughput is measured with a server sending UDP packets at a rate of 100Mbit/s over its Ethernet network interface card (NIC) which is connected to the AP. It is to be observed how many packets arrive at the client. Since measurements with packets originating from the client station proved to be unstable in the previous paragraph, the measurement is only done one-way - with the packet streams flowing from the AP to the client station. The unit of the measured data is Mbit/s where $M = 10^6$.

| ACK | Avg. data rate | Min. | Max. | Std. deviation |
|---|---|---|---|---|
| Immediate | 52.4 Mbit/s | 51.8 Mbit/s | 52.9 Mbit/s | 0.4 Mbit/s |
| Burst-ACK | 59,5 Mbit/s | 53.7 Mbit/s | 61.1 Mbit/s | 1.6 Mbit/s |
| No-ACK | 65,6 Mbit/s | 56.5 Mbit/s | 72.4 Mbit/s | 6.6 Mbit/s |

- Immediate-ACK proves to be the most stable acknowledgement function with the lowest standard deviation in throughput. No-ACK enables very high data rates at the cost of reliability and a high fluctuation of 6.6 Mbit/s standard deviation.
- Burst-ACK is again a good compromise where maintained error correction is combined with a fair standard deviation of 2.4 Mbit/s at high data rates.
- The values give hints about how fast a media server on the Ethernet attached to the AP could stream to a client behind the AP in the wireless LAN.

### UDP packet loss measurement

Packet loss can not be measured in the same way as in section 8.1.2 because an Ethernet NIC has a much higher throughput than the wireless LAN. Observing the packet loss with the data stream flowing in the same way as in section 8.1.2 one could not distinguish between packet losses due to the buffer at the AP overflowing and packet loss on the wireless MAC layer. Therefore one has to measure the packet loss at a station where the UDP data rate is limited by the speed of the wireless NIC. For No-ACK this yields a problem as unfortunately there seems to be a windows driver issue with No-ACK leading to very low throughput. Still, the results are as follows:

| ACK-Method | Packet loss | Throughput |
|---|---|---|
| Immediate | 0 | ~ 49 Mbit/s |
| Burst-ACK | 0 | ~ 54 Mbit/s |
| No-ACK | 1% | ~ 5 Mbit/s |

## 8.1.3 Pre-N MAC and ACK gains

When a stream from the AP to a client is observed, the following table that shows the gains of the different MAC and ACK methods can be compiled from the above results. Note that the column “ACK-method” denotes the ACK-methods at both ends of the transmission except for No-ACK that proved to be unstable at the non-AP station - which should be a problem that can be fixed. In this case only the values of the AP were accounted for. Note also that the following are Pre-N gains compared to 802.11g and Pre-N ACK-method gains.

| ACK-method | Protocol | Compared to 802.11g | Compared to Imm. ACK |
|---|---|---|---|
| Immediate | UDP | +84 % | + 0 % |
| Burst-ACK | UDP | +110 % | + 14 % |
| No-ACK | UDP | +131 % | + 25 % |
| Immediate | TCP | + 88 % | + 0 % |
| Burst-ACK | TCP | + 95 % | + 12 % |
| No-ACK | TCP | + 75 % | - 2 % |

- Again it is clearly visible that No-ACK improves performance at the cost of reliability. In the case of UDP the performance gain is the largest but also the packet loss is largest - as can be seen for TCP where a lost packet is retransmitted yielding an overall lower throughput than with Immediate-ACK.

## 8.2 N-LOS environment

As the tests in the previous environment have included a line-of-sight path between the two stations it is now to be observed how the system behaves an environment where multiple signal paths are enforced by the spacial geometry while a direct line-of-sight path is suppressed, yielding a non-line-of-sight (NLOS) environment. In this scenario the signal has to propagate around a corner where parts of the signal should be reflected at the walls in the right direction while other parts should be reflected to different directions and lost. An obstruction in the form of a wooden door is added. The distance is increased to approximately 10 meters.

### 8.2.1 EDCF performance

#### UDP throughput measurement

For comparison with the above results the UDP performance is taken into account since it provides information about the fluctuation of the data rate, which is critical for audio-visual transmission over a wireless link.

| ACK | Avg. data rate | Min. | Max. | Std. deviation |
|---|---|---|---|---|
| Immediate | 45.1 Mbit/s | 39.2 Mbit/s | 49.4 Mbit/s | 4.3 Mbit/s |
| Burst-ACK | 55,7 Mbit/s | 47.5 Mbit/s | 58.2 Mbit/s | 4.4 Mbit/s |
| No-ACK | 66,7 Mbit/s | 46.9 Mbit/s | 70.4 Mbit/s | 7.9 Mbit/s |

- Compared to the previous results the data rate is nearly maintained in this less ideal scenario except for Immediate-ACK which suffers most from the missing of a direct path.

- The channel conditions are fluctuating much more than before with increased standard deviations in throughput.
- Again it is observable that No-ACK has the highest fluctuations and Burst-ACK has the lowest relative fluctuations.

**UDP packet loss measurement**

Refer to section 8.1.2 on page 33 for problems that also hold here.

| ACK-Method | Packet loss | Throughput |
|---|---|---|
| Immediate | 0 | ~ 48 Mbit/s |
| Burst-ACK | 0 | ~ 52 Mbit/s |
| No-ACK | 1.7% | ~ 5 Mbit/s |

## 8.3 Performance with encryption

### 8.3.1 WEP security

The performance impact of WEP encryption was unfortunately not measurable for TCP. It seems that there is another driver issue since the connection was lost every time on multiple installations and the throughput was extremely low - due to the lost connection I believe. Strangely the effect was not noticeable with UDP where the data rate for Burst-Acknowledgement on both sides was around 50 Mbit/s. So, at least for UDP there seems to be no performance impact due to WEP encryption.

### 8.3.2 WPA security

In this section the performance impact of the WPA security mechanism is to be measured in a LOS scenario as above. The WPA encryption implements the basic security features described in 802.11i. Without going into detail the basic idea of WPA is to replace the insecure WEP encryption by an algorithm that reduces the probability of cracking the encryption key by[5]. Also the design rules for WPA were that it should be computable with reasonable effort so it could be integrated into existing WEP-enabled hardware via a firmware upgrade. In the ideal case the performance impact of the computing effort for encryption is negligible. The measurements should show if this holds for the observed Pre-N hardware.

Note: As the encryption adds some overhead to the packet data the throughput is expected to be slightly decreased by encryption but the impact is believed to be minimal.

| Protocol | ACK-method | Data rate |
|---|---|---|
| TCP | Burst-ACK | 16.3 Mbit/s |
| UDP | Burst-ACK | 21 Mbit/s |

[5] Which was possible with WEP in reasonable time only by observing encrypted packets.

- A severe performance drop is observed. The resulting data rate is reduced by a factor of 2-3.
- This leads to the conclusion that computing effort is a limiting factor here. A performance drop due to packet overhead is not noticeable for 802.11g, as shown in [Hig03].

## 8.4 Direct-link protocol test

Ideally it would only be necessary to observe if packets that ought to be transmitted from one non-AP station to another non-AP station are twice in the air. But since wireless sniffing is not possible with the used hardware, the direct link protocol implementation can alternatively be tested when observing the behaviour of the data rate of a stream between two stations. As the scenario of transmitting from one station to another station is a multi-hop scenario with two hops, the expected data rate is either

- the single-hop data rate - which would be a clear indication of the direct-link protocol being enabled
- or it is the single-hop data rate divided by two - which would be an indication that the direct-link protocol is not enabled.

The following table is a comparison of the above single-hop measurements (Burst-ACK) with multi-hop measurements

| | TCP | UDP |
|---|---|---|
| single-hop | up to 40 Mbit/s | up to 60 Mbit/s |
| multi-hop | up to 19 Mbit/s | up to 30 Mbit/s |
| Ratio | nearly 1/2 | fairly exactly 1/2 |

- The direct-link protocol seems to be not implemented. The data rate is around the predicted “single-hop/2” value.

## 8.5 Multicast

The multicast performance was again measured with iperf (for reference see above) in multicast mode. A server on the Ethernet behind the AP sent packets at a controllable rate. Two multicast clients on the wireless LAN were listening.

I will not give specific measurements results but instead only a short summary of the behaviour since the packet loss was very large, in fact to large to be useful at all. Even at very low bit rates of about 1Kbit/s the packet loss was in the range of two to five percent. At a rate of 10Mbit/s the packet loss was at 90%. This is totally surprising since the 802.11 MAC should be very well suited to deal with this kind of traffic as I have stated above in chapter 6. I can only guess that this is a problem with power-save-queueing enabled by default which is discussed in section 6.2.

## 8.6 Additional Belkin parameters

### 8.6.1 At the AP

**Wireless Mode** [g only, b/g mixed] (g)
In "g only" mode no pre-802.11g devices are allowed to associate with the AP.

**Protected Mode** [on, off] (off)
This parameter controls the protection from 802.11b modems. Enabling this option reduces the performance. The cause is discussed in section 2.3 on page 12.

**ACK Mode** [Immediate, burst, no, auto] (auto)
The acknowledgement policy of the AP is set here. The setting chosen here is not mandatory for associated stations. Every station can independently choose an acknowledgement method.

**802.11e QoS** [on,off] (off)
This parameter controls if the 802.11e QoS enhancements are to be used. The parameter seems to also control the use of MIMO. If enabled, only Pre-N stations with medium access set to "EDCF" (see later) can associate with the AP.

### 8.6.2 At the client - driver options

**11g short slot time** [0,1] (1)
This parameter controls the short slot time. Reducing the value to 0 from its default value 1 renders the wireless link unusable. From my understanding it is analog to the AP option "wireless mode" where setting this parameter to 0 corresponds to activating a sort of compatibility to old modems which also do not use the 802.11g short slot time but the "longer" 802.11b slot time.

**Network Density** [low,medium,high] (low)
This parameter presumably controls the use of performance reducing but link quality increasing protection mechanisms. The "low" setting should be the most "unsafe" and therefore the best choice with respect to performance.

**Transmit Antennas** [1-2] (2)
This parameter controls how many antennas are used for transmission. Reducing the value to 1 reduces the performance by approximately 1/2.

**Receive Antennas** [1-3] (3)
This parameter controls how many antennas are used for reception. Changing the parameter affects the performance with 3 being the optimal value.

**ACK timeout** [integer] (30 for 802.11b/g)
This parameter controls the ACK timeout presumably in multiples of short-slot-time. A performance increase is not noticeable tuning the value in one or the other direction.

**Adaptive Threshold Algo Selection** [D0,CD,disabled] (D0)
I could not find out what the function of this switch is. Changing it crashed the system.

**Short retry limit, Long retry limit, RTS Threshold** Since RTS/CTS is not subject of this work and the retries are expected to be very low since low interference and a low number of wireless clients is assumed, the parameters are not considered.

**CCA mode** [Combinations of ED, CS and CD]
CCA is the clear channel assessment function, i.e. the logical function in the physical layer (PHY) that determines the current state of use of the wireless medium. Under the conditions discussed in this text the CCA mode is believed not to have an impact on performance. ED: Energy detection, CS: Carrier sense, CD: Carrier sense and energy detection.

**ACK policy** [Immediate, No-ACK, Burst-ACK] (Immediate)
This parameter controls which ACK method is to be used. It is in detail described in previous chapters.

**Medium Access** [Auto, DCF, EDCF, HCF] (DCF)
This parameter controls which MAC model is used. The EDCF mode enables CoS while the DCF mode provides backwards compatibility. The HCF option is not subject of this work and could not be tested since the AP does not support it.

**11d support** [On,Off] (Off)
Controls if 802.11d, also referred to as “world-mode” should be enabled. This parameter has no performance affecting character.

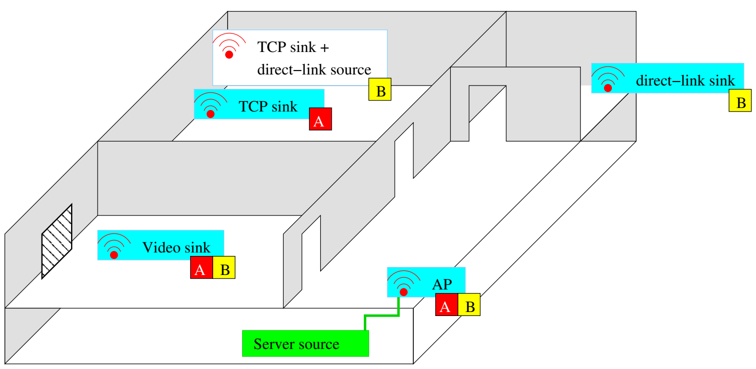
# Chapter 9

# QoS scenarios

## 9.1 Definition

A typical scenario for in-home media distribution would be the one depicted in figure 9.1. A media server is located in the basement and made available to the networking clients by an 802.11e QoS-enabled AP. A QoS enabled station resides in the living-room and is connected to a display and a hi-fi system of some sort. At this station high data rate traffic delivering multi-media content like HDTV is to be received. The goal is to achieve a constant, uninterrupted traffic flow at this point to ensure the quality of this service at any time.

Figure 9.1: QoS scenario




### Scenario A

In another room resides a second QoS-enabled wireless client that will receive a non-otherwise-limited TCP file transfer stream from the server with maximum overall packet size 1500 bytes.

- The audio/visual data stream shall be a CBR 20Mbit/s (e.g. HDTV) UDP stream from the server to the media client in the living room.
- At some point in time the TCP file transfer is initiated.

### Scenario B

A third wireless station is added as depicted in figure 9.1 on the preceding page. From this station a TCP station-to-station (multi-hop or direct-link) stream of the AC "best-effort" is to be initiated which has the same properties as the first stream to maximise the possible influence.

- 20 Mbit/s CBR UDP stream from scenario A
- TCP file transfer stream from scenario A
- Some time later the station-to-station TCP file transfer is initiated.

In both scenarios the impact on the video stream is to be observed.

## 9.2 Simulation with ns2

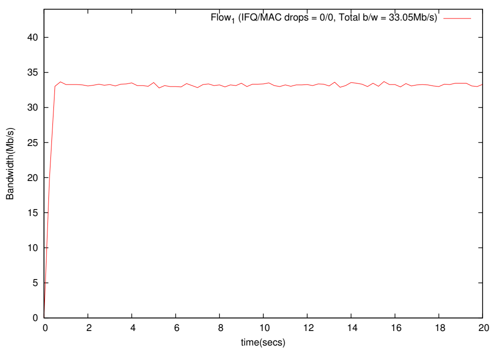
### The ns-edcf implementation

The simulation is done with an EDCF implementation for ns2 that was elaborated by Dr. Ni Qiang of the Planète group at the INRIA Sophia Antipolis research institute.

### Direct-Link

As direct-link may not have been included in the draft 802.11e standard at the time the ns-edcf implementation was done, a quick test of a station-to-station transmission should reveal if direct-link is implemented or not. The resulting throughput is compared to the throughput resulting from a single-hop station to AP transmission under the same conditions. The resulting graphs are depicted in figure 9.2. It is clearly visible that the data rate for the dual-hop transmission is fairly exactly half of the single-hop data rate.

Figure 9.2: Direct-link quick check



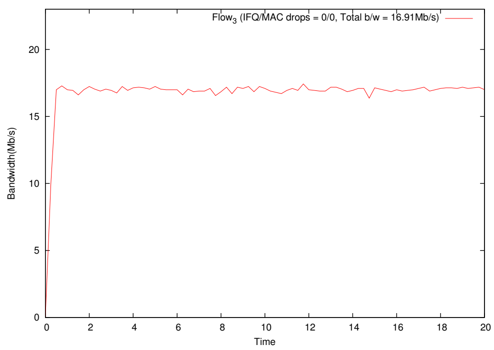
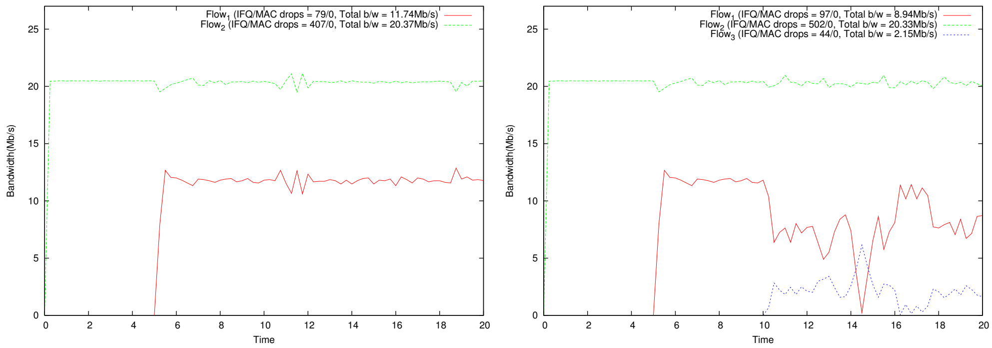



Figure 9.3: Ns2 simulation of QoS scenario A and B




### QoS scenario A simulation

In figure 9.3 it is observable that the HDTV stream which is assigned to the AC Video clearly maintains its constant bit rate while the single-hop TCP stream can not influence the constant data rate of the UDP stream. The throughput graph is depicted in figure 9.3 generated by the ns script on page 49.

### QoS scenario B simulation

Again it is observable that the HDTV stream maintains its constant bit rate while both the single-hop and dual-hop TCP streams of different AC do not interfere. The throughput graphs are depicted in figure 9.3 generated by the ns script on page 49.

### Simulation conclusion

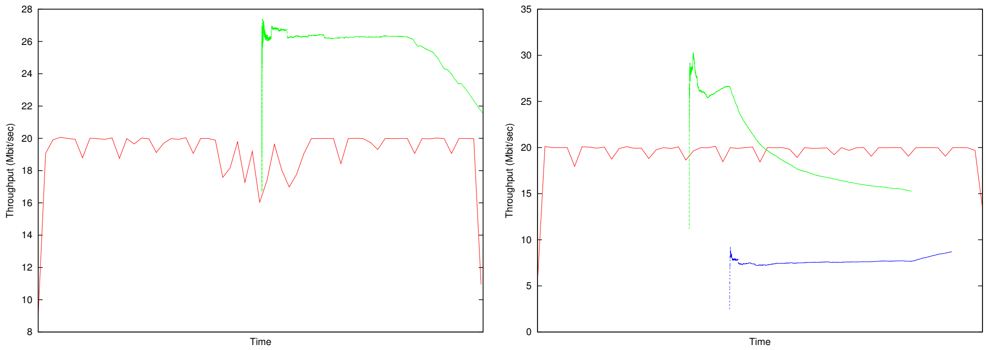
Summing up the average throughput values depicted in 9.3 for scenario A and B results in a total bandwidth of around 30 Mbit/s. One has to note that the simulation was done with all standard values and the ACK model is Immediate. Still the effect of the priorisation scheme of 802.11e is clearly visible.

## 9.3 Pre-N measurement

### QoS signalling

First thing that is necessary to state here is that the Pre-N modems seem to classify data streams that have no previous Ethernet (802.1D) priority marking in a way that UDP streams get a higher priority than TCP streams. This is sufficient since the assumption that in most cases only UDP streams contain data that is delay or bandwidth critical is realistic. TCP on the other hand can be regarded as a container for traffic that is in most cases for purposes where reliability is the main goal - e.g. file transfers. Situations where reliability and e.g. low latency are needed should be rare exceptions in an in-home wireless LAN. An example would be remote logins like ssh.

Figure 9.4: Measurement of scenario A and B




### Direct-Link

As measured in section 8.4 on page 36 the data rate of a station-to-station transmission is equal to the expected multi-hop ($n_{hop} = 2$) data rate under the same conditions. Thus one can assume that the direct link protocol is not implemented

### Measurement results

In figure 9.4 the measurements of scenario A and B are depicted. The graphs are very similar to the simulated results except for an overall higher bandwidth which results from the use of burst-ACK. The overall bandwidth is noted in the following table since it is not visible from the graphs.

| Scenario | UDP | TCP 1 | TCP 2 | Sum |
|---|---|---|---|---|
| A | 19.8 Mbit/s | 21.5 Mbit/s | - | 41.3 Mbit/s |
| B | 19.7 Mbit/s | 15.3 Mbit/s | 8.7 Mbit/s | 43.7 Mbit/s |

There are peaks in the beginning of the TCP transmissions that do not reflect the true data rate on the wireless medium. As far as my understanding goes the peaks are due to the socket buffer being filled to fast and when it is full the measured data rate converges to the data rate the wireless medium can provide. Thus the table above provides average data rate values returned by iperf that are closer to the truth.

# Chapter 10

# Media applications bandwidth requirements

In the following table a couple of audio/visual applications and their respective data rate requirements (peak values) are collected. It is not complete and should only give an overview which formats are already distributable over a standard 802.11g wireless LAN and which formats will only be distributable via a 802.11e + 802.11n enabled WLANs. As an example it should be possible to distribute 3-10 standard definition (SD) DVB streams over a Pre-N wireless LAN under ideal conditions which should be sufficient for an in-home wireless LAN TV setup. Another example would be that HDTV is very likely not to work in conjunction with an 802.11g wireless LAN.

Table 10.1: Requirements versus capacities

| Application | Format | Format specific properties | Data rate (Mbit/s) | Resource usage | |
|---|---|---|---|---|---|
| | | | | 11g | 11e+n |
| VCD | MPEG-1 | 352x288, 25fps (PAL), CBR, MPEG-1 Audio Layer 2 | 1.5 | 5% | 2.5% |
| SVCD | MPEG-2 | 480x576, 25fps (PAL), CBR, MPEG-1 Audio Layer 2 | 3 | 10% | 5% |
| DivX, Xvid, etc. | MPEG-4/ASP derived | various profiles | 1-2 | 5-7% | 2.5-3.5% |
| DVD | MPEG-2 | Main profile @ Main level | 15 | 50% | 25% |
| HDTV 720p | MPEG-2 | 1280x720 progressive | 30 | 100% | 50% |
| HDTV 720p | H.264/AVC | Level 4, 1280x720 progressive | 20 | 66% | 33% |
| HDTV 1080i | MPEG-2 | 1920x1080 interlaced | 30 | 100% | 50% |
| HDTV 1080i | H.264/AVC | 1920x1080 interlaced | 50 | - | 83% |
| HD-DVD | H.264/AVC or LC-1 (WMV9 | HDTV 1080i | 36 | - | 60% |
| DVB SD | MPEG-2 | MP@ML or MP@LL | 4-15 | 13-50% | 6.5-25% |
| DVB HD | MPEG-2 | Euro 1080 (HD-1) | 30 | 100% | 50% |
| DVB-S2 | H.264/AVC | Level 4.1, 1080i | 50 | - | 83% |

Reference 802.11 data rates are UDP data rates; 11g: 30Mbit/s, 11e+n: 60Mbit/s

# Chapter 11

# Conclusion

The upcoming standards 802.11e and 802.11n provide sufficient measures to provide Quality as well as Class of Service combined with increased data rates.

Being an early implementation of those standards, Pre-N performs quite good. Performance is nearly un if a line-of-sight path is suppressed. This is a major improvement compared to standard 802.11 devices and it shows the power of the MIMO technology. The data rate is also increased by Pre-N compared to 802.11g nearly by a factor of two. A drawback is that enabling EDCF renders the wireless LAN incompatible to previous and widely spread 802.11g or b devices. Also the direct-link protocol seems to be not implemented which would be important in a scenario where a high data rate audio-visual content provider does not lie behind the AP but should stream data from one station to another. A third major drawback is the unusability of multicast transmission which could also be desirable in an in-home wireless network for example if the same TV program streamed over the wireless LAN ought to be received at multiple stations. It is obvious that the Pre-N devices observed in this work do not fulfil the specifications of the upcoming standards 802.11e and 802.11n completely. But some of the problems encountered seem to be correctable with driver and firmware upgrades. In my opinion the compatibility of devices fulfilling the specifications of 802.11e versus devices fulfilling 802.11g should be maintained in Pre-N and future devices for EDCF since it should be possible with reasonable effort.

In selected scenarios the Pre-N modems perform very good and enable the distribution of high data rate audio/visual content over wireless links, even with more than one stream in case of not extremely high demanding applications. The reduction of the CoS determination to TCP and UDP as traffic classes is sufficient in most cases. But the Pre-N modems are also easily brought to their limits with respect to the problems multicast and direct-link. The major benefits come from the use of the block-ACK mechanism, the priorisation scheme and the two spatially separated channels by the use of MIMO, increasing signal strength and data rate.

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# Appendix A

# Scilab source code

## A.1 const.sci - 802.11 protocol specific constants

```
//universal constants
N=5; //number of stations
MTU=1500; //max. ethernet transmission unit
FrameHeaders=32 + 5 + 3;  //MAC+SNAP+LLC
IPv4Header=20;
UDPHeaders=8 + IPv4Header;
TCPHeaders=20 + IPv4Header;

//constants in bytes/seconds for the various 802.11 protocols
protB.PLCP=192e-6;
protB.SIFS=10e-6;
protB.SlotTime=20e-6;
protB.DIFS=2*protB.SlotTime+protB.SIFS;
protB.DCF_IFS=protB.DIFS+protB.SIFS; //data DIFS + ACK SIFS
protB.MaxFrameBody=2312;
protB.AckFrameSize=14;
protB.eSlot=18;
protB.Rate=11; //Mbit/s

protG=protB;
protG.Rate=54;
protG.SlotTime=9e-6; //short slot time, no b/g-mixed-mode
protG.DIFS=2*protG.SlotTime+protG.SIFS;
protG.DCF_IFS=protG.DIFS+protG.SIFS;
protG.PLCP=20e-6;

protA=protG;
protA.eSlot=12;

protN=protG;
protN.Rate=108;

protE=protN; //choose which "bandwidth" protocol to use for 11e here
protE.MaxFrameBody=2304;
```

## A.2 dcf.sci - DCF channel access models

```
clear;
exec("const.sci")

// Contention-Window function
deff('[y]=fCW(i)','y=2^(3+i)-1')
```

```
// DCF constants
DCF_i = 1:7
DCF_CWmin = fCW(min(DCF_i))
DCF_CWmax = fCW(max(DCF_i))

// DCF probabilities
deff('[y]=fPt(N,proto)','y=1/N*(1-1/proto.eSlot)^(N-1)');
deff('[y]=fPce(N,proto)','y=1-N*fPt(N,proto)');

//DCF delays (transmission, minimum=PLCP+IFS+contention)
deff('y=tx_delay(fs,p)', 'y=8*(p.AckFrameSize+fs)/(p.Rate*(1000^2))');
deff('y=min_delay(n,p)','y = 2*p.PLCP + p.DCF_IFS ..
                             + (p.SlotTime*(p.eSlot))'); //expectation

//DCF throughput
function r = DCFtp(n,fs,tx_enc,proto)
    if fs > protB.MaxFrameBody then error('payload too large'), end
    delay = 0;
    if tx_enc == "RAW" then
        fs = proto.MaxFrameBody; //always override
        payload = proto.MaxFrameBody;
    elseif tx_enc == "UDP" then
        payload = fs - UDPHeaders;
    elseif tx_enc == "TCP" then
        payload = fs - TCPHeaders;
        //TCP ACK, size of is IP+TCP headers (40 Byte) + MAC overhead
        delay = min_delay(n,proto) + tx_delay(FrameHeaders+TCPHeaders,proto);
    else
        error('Invalid mode specified')
    end
    frame_size = fs+FrameHeaders;
    delay = delay+min_delay(n,proto)+tx_delay(frame_size,proto);
    r = fPt(n,proto)*(payload / delay)*8;
endfunction;
```

# Appendix B

# ns2 scripts

## B.1 Scenario A

```
# A wireless lan scenario with video/cbr traffic over udp
# and two bulk traffic streams over tcp

set num_nodes 5 ;# number of mobilenodes in the scenario

proc create_scenario { } {
global ns_ node_ AP_ title_ mode

##########################################################
# Setup Video flow V

# period/pktsize of Video Streams based on the blocking factor
        set period 0.0008
set pktsize [expr 2048 * 1]

# Flow V

set V [new Application/Traffic/CBR]
set V_src [new Agent/UDP]
set V_sink [new Agent/UDP]

$V set random_ 0
$V set packetSize_ $pktsize
$V set interval_ $period
$V_src set packetSize_ 1500
$V_src set class_ 2
$V_src set prio_ 3

$ns_ attach-agent $AP_ $V_src
$ns_ attach-agent $node_(1) $V_sink
$ns_ connect $V_src $V_sink
$V attach-agent $V_src
puts "V flows from AP to Node1 over UDP"


##########################################################
# Setup Background flow B1
# B1 flows from the AP to Node4

set B1 [new Application/FTP]
set B1_src [new Agent/TCP]
set B1_sink [new Agent/TCPSink/DelAck]

$B1_src set syn_ 1
$B1_src set packetSize_ 1500
```

```
$B1_src set class_ 1
$B1_src set prio_ 5

$ns_ attach-agent $AP_ $B1_src
$ns_ attach-agent $node_(3) $B1_sink
$ns_ connect $B1_src $B1_sink
$B1 attach-agent $B1_src
puts "B1 flows from AP to Node3 over TCP"


##########################################################
# Setup Background flow B2 (only for scenario B)
# B2 flows from Node3 to Node4 via the AP

set B2 [new Application/FTP]
set B2_src [new Agent/TCP]
set B2_sink [new Agent/TCPSink/DelAck]

$B2_src set syn_ 1
$B2_src set packetSize_ 1500
$B2_src set class_ 3
$B2_src set prio_ 5

$ns_ attach-agent $node_(3) $B2_src
$ns_ attach-agent $node_(4) $B2_sink
$ns_ connect $B2_src $B2_sink
$B2 attach-agent $B2_src
puts "B2 flows from Node3 to Node4 over TCP"


##########################################################
# Start simulation

set access $mode
if {[PLevels set plevels_] > 1} {
set access VDCF
}
$ns_ at 0.0 "$V start"
$ns_ at 5.0 "$B1 start"

set phy_bw [Mac/802_11 set bandwidth_]
set retries [MAC_MIB set ShortRetryLimit_]
set pqlim [Queue/DropTail set pqlim_]
set title_ "set title \"QoS scenario 1\""
}
```

## B.2 Scenario B

The additional node is already included in the tcl file of scenario A. Only the following line needs to be added in the appropriate place:

```
$ns_ at 10.0 "$B2 start"
```