Contents

Requirements for real-time services, RTP

QoS solutions in 802.11 networks

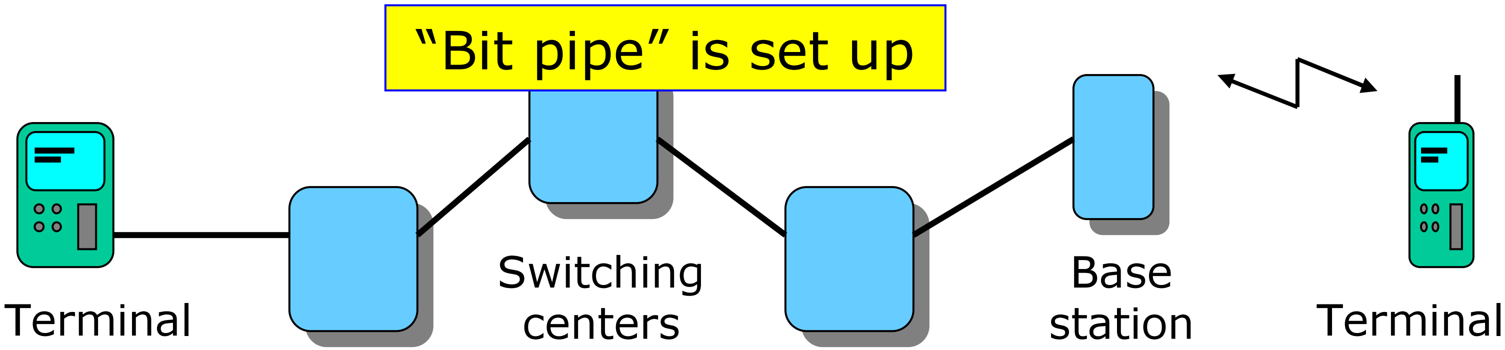
- PCF
- Proprietary solutions
- 802.11e

VoIP over WLAN

- Mobility management and session control
- Voice coding

Circuit switching vs. packet switching (1)

Circuit switching: A constant-capacity “bit pipe” is set up between two terminals through a circuit switched network (usually PSTN and/or PLMN) using call control signalling.



Circuit switching vs. packet switching (2)

Advantages of circuit switching:

Fixed, predictable and guaranteed capacity. Once the connection is established, it is reserved for the duration of the call.

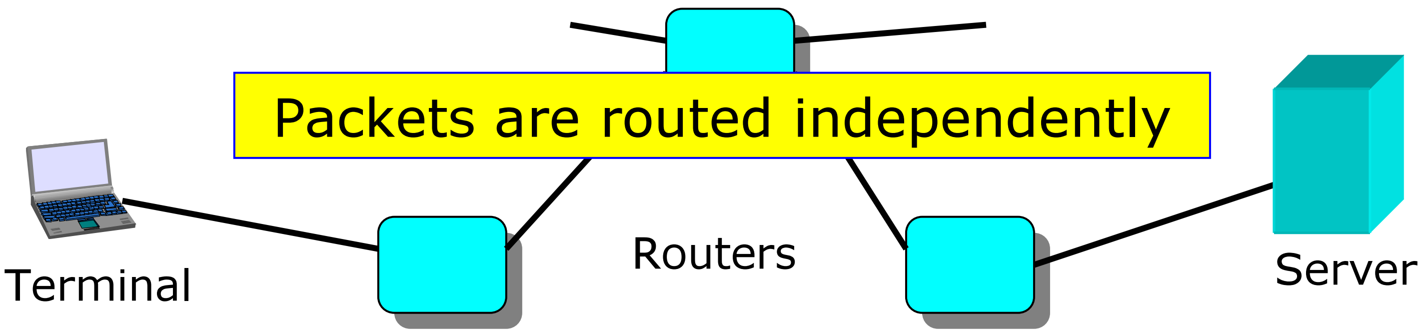
Small delay and small delay variation. There is no buffering (causing delay variations) in the network.

Disadvantages of circuit switching:

Complex signalling, no retransmission possible in case of bit errors, inefficient for bursty traffic.

Circuit switching vs. packet switching (3)

Packet switching: The information is carried in packets (usually IP packets) that are routed independently through the network. There is no call control signalling.



Circuit switching vs. packet switching (4)

Advantages of packet switching:

- Efficient utilisation of network resources in case of bursty traffic (“bandwidth on demand”).
- Retransmission possible (necessary for error-sensitive applications).

Disadvantages of packet switching:

- Delay and delay variations (=> voice traffic).
- No guaranteed bandwidth (=> streaming video).
- Possibility of congestion (call must be dropped).

Performance of an 802.11 network

There is no way of handling circuit switching in 802.11 networks => the disadvantages of packet switching (previous slide) must be taken seriously:

- Delay and delay variations are especially severe when packet technology is combined with radio technology
- 802.11 networks do not offer traffic management, so congestion is a real threat (data and voice traffic have the same priority; voice traffic cannot reserve fixed channel capacity).

Delay (1)

In most cases, the term QoS (Quality of Service) refers to the delay or delay variation in voice transmission (or other delay-sensitive applications).

In most data applications, QoS (i.e. small delay) is not important.

ITU-T Recommendation G.114 states that the round-trip delay should be less than 300 ms for telephony.

802.11 networks operating near (or at) their capacity limit may cause significant frame transmission delay.

Delay (2)

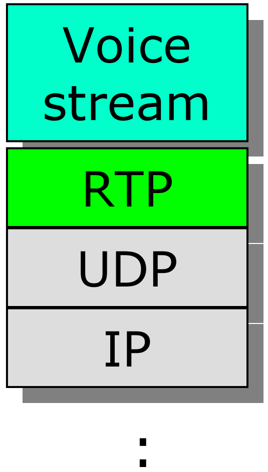
Various mechanisms contribute to the total transmission delay of a packet connection (including the WLAN):

- The CSMA/CA protocol (deferring, backoff) even without retransmissions
- Retransmissions (if allowed)
- Buffering delay (terminal, AP, routers in the packet network) => significant in high load situations
- Signal processing in the terminals (voice or video coding and decoding).

Real Time Protocol (RTP)

RTP is used for carrying real-time data (e.g. coded voice) over IP networks. RTP offers two features:

- The correct packet order is maintained at the destination
- RTP packets include a time stamp that records the exact time of transmission.



Time stamps can be used at the destination to ensure synchronised play-out of (e.g.) voice samples.

Delay variation => use RTP

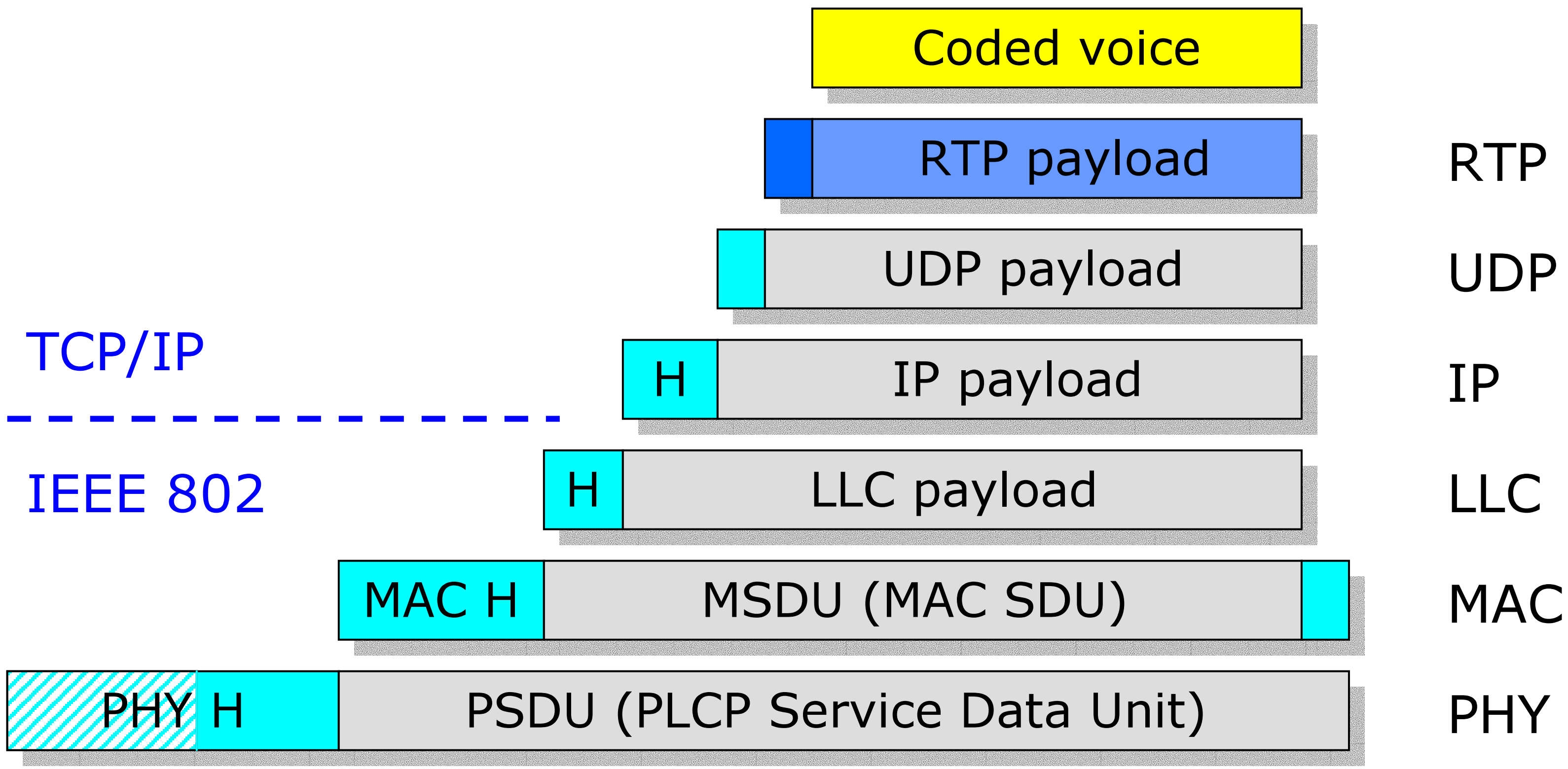
Naturally, RTP cannot affect the total transmission delay in the network.

However, the usage of time stamps helps to reduce the time variation or jitter at the destination.

RTP in itself cannot reduce the time variation. This is the task of the application (by utilising the time stamps provided by RTP) at the destination.

RTP is able to carry a large variety of coded information (audio or video) => the standard solution for VoIP.

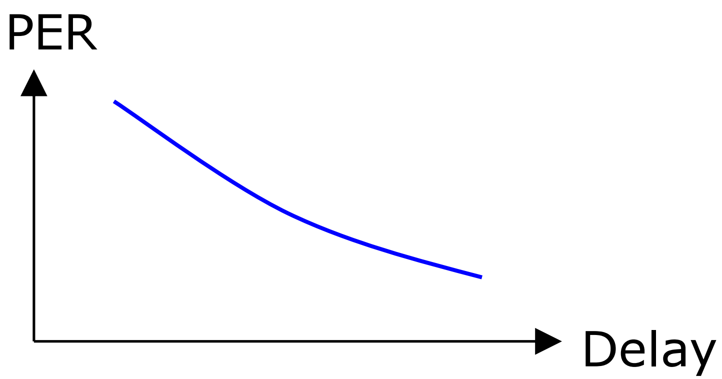
Typical “VoIP over WLAN” protocol stack



Packet Error Ratio (1)

The packet error ratio (PER) depends on the quality of the channel (signal attenuation, interference within the channel bandwidth) and the bit rate (higher bit rate => lower receiver sensitivity).

When retransmissions are allowed, there is a trade-off between PER and delay (qualitative illustration =>)

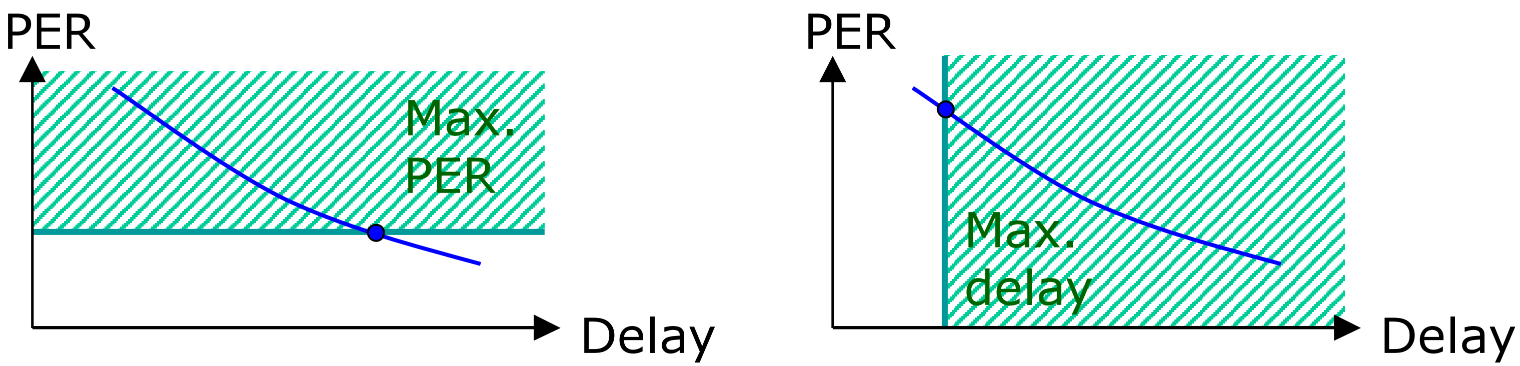


Packet Error Ratio (2)

The optimal PER/delay choice (in practice: maximum number of retransmissions) depends on the type of service (data, voice, multimedia…):

Error-sensitive services

Delay-sensitive services



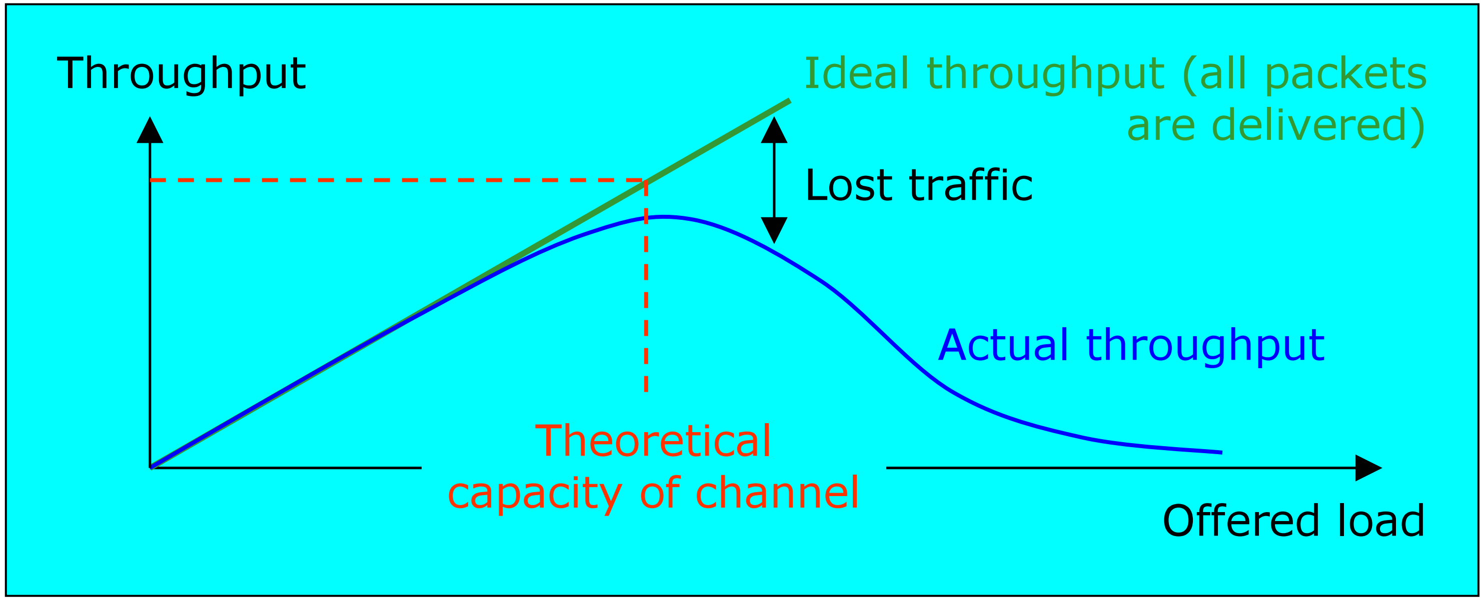
Throughput (1)

Medium sharing protocols (like CSMA) perform well as long as the network load is light. When the offered load approaches the theoretical capacity of the network, there will be congestion. If this happens, packets will accumulate in the buffers of the AP and wireless stations => large delays and lost packets due to buffer overflow.

In contrast with packet errors in the radio medium (where the 802.11 MAC takes care of retransmission) lost packets due to buffer overflow must be handled by higher protocol layers (e.g. TCP).

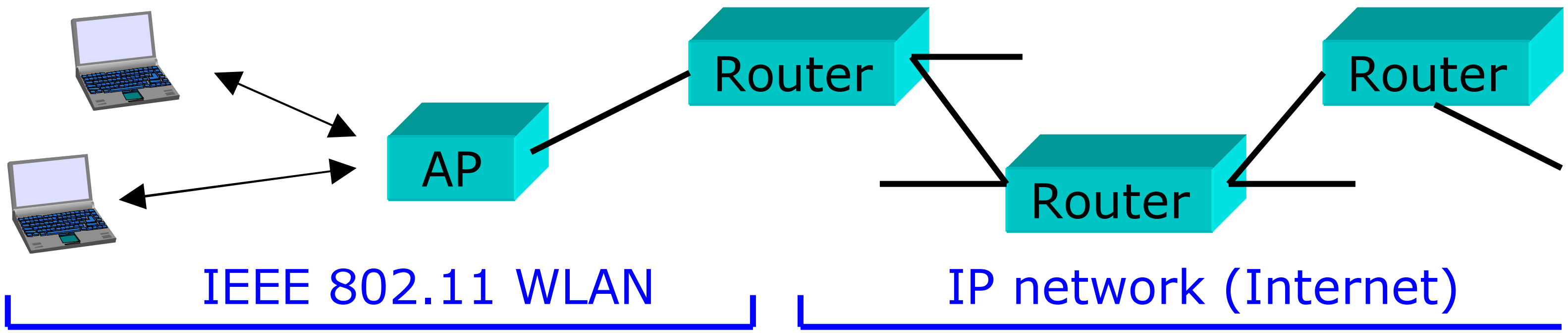
Throughput (2)

A qualitative illustration of the situation:



QoS (Quality of Service)

QoS means in practice that real-time traffic experiences small delays and small delay variation in the network. Streaming applications assume guaranteed bandwidth.



QoS support in the WLAN (especially radio interface)

QoS support in IP networks is out of scope of 802.11

QoS solutions in IP networks

The following QoS solutions are available for IP networks in general:

- DiffServ: The traffic is divided into different priority classes. The priority class is indicated in the IP header. DiffServ-capable routers handle the traffic in different priority classes differently.
- Multi-Protocol Label Switching (MPLS): Routing in the IP network is connection-oriented (i.e. based on OSI layer 2 MPLS labels instead of layer 3 IP addresses). MPLS-capable routers are required.

QoS solutions in 802.11 networks

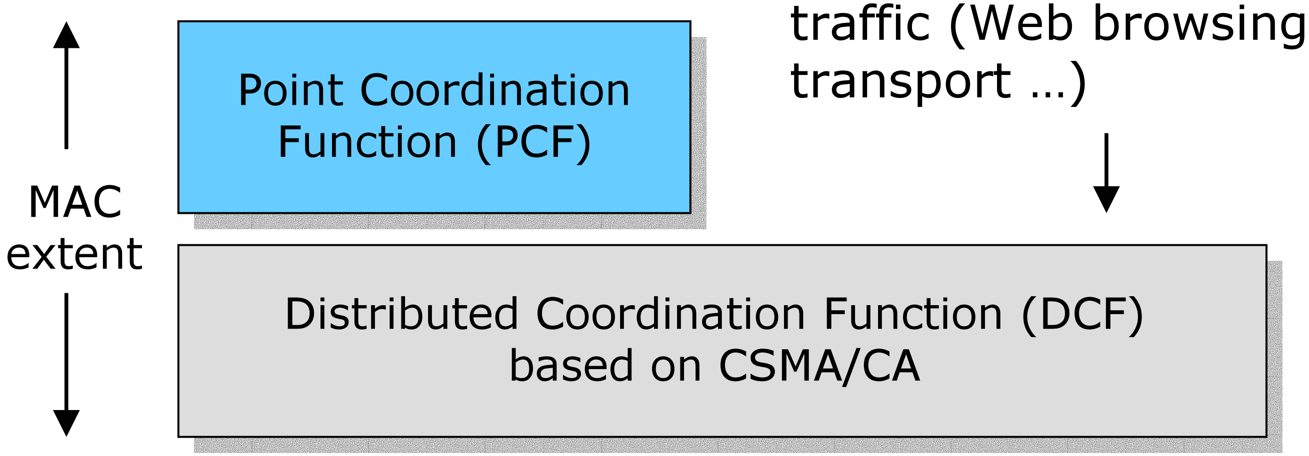
Since traffic routing in WLAN networks is not based on IP, there must be different QoS solutions available:

- The 802.11 standard defines the Point Coordination Function (PCF) for carrying real-time traffic. This solution has not been widely implemented.
- There are proprietary solutions that try to differentiate real-time and non-real-time traffic in the WLAN.
- A number of advanced QoS solutions have been defined in the 802.11e standard (approved in 2005).

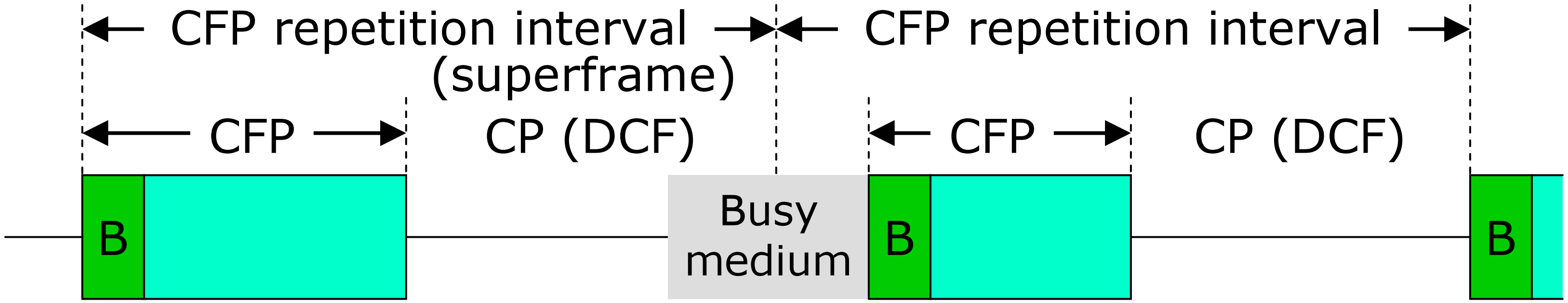
PCF (Point Coordination Function)

Included in the 802.11 specifications, PCF was especially designed for delay-sensitive real-time services

Intended for non-real-time traffic (Web browsing, file transport …)



PCF operation



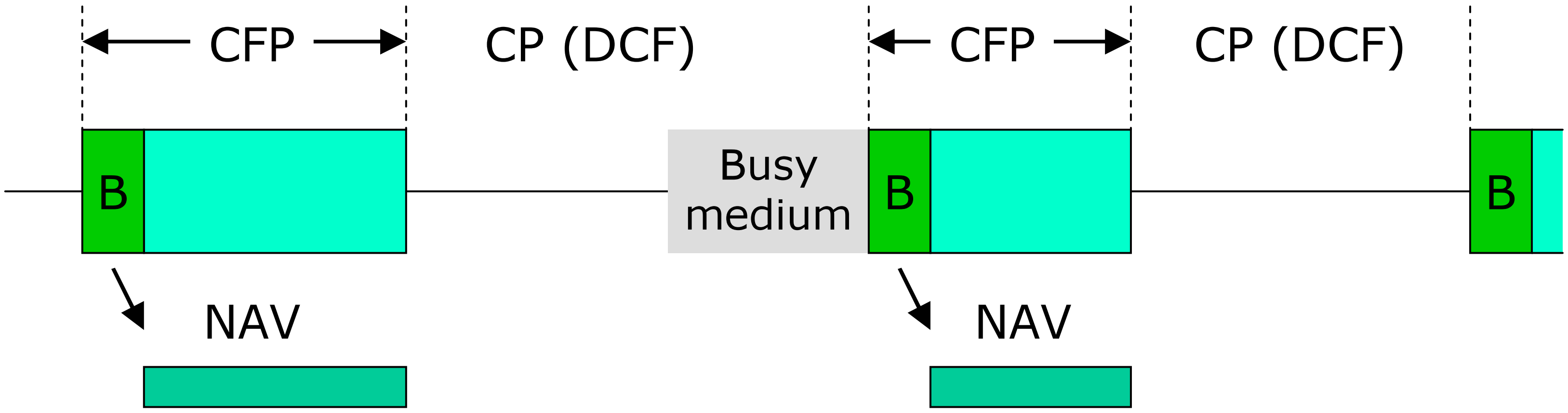
B = Beacon frame (sent by AP to indicate start of CFP)

CPF = Contention-Free Period (reserved for real-time traffic)

CP = Contention Period (normal DCF operation)

Note the foreshortening of the CFP due to the busy medium (it is not possible to cut off active DCF transmissions)

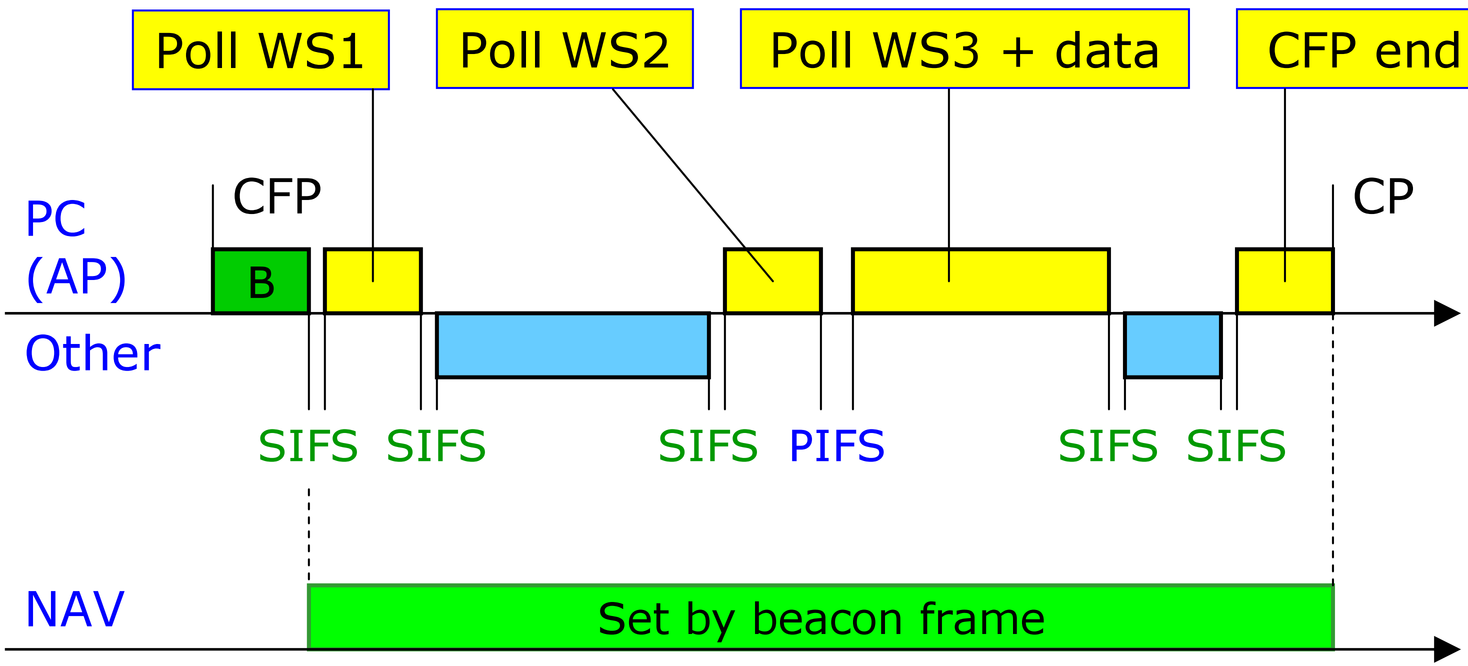
PCF operation (cont.)



Undisturbed CFP operation is guaranteed in two ways:

- The NAV value in the beacon signal = length of CFP
- Usage of PIFS within CFP (instead of DIFS), PIFS < DIFS

PCF is based on polling, not CSMA/CA



Proprietary QoS solutions

The PCF option has never become popular in the industry. However, some 802.11 equipment vendors offer other solutions for real-time (in practice = VoIP) support.

A solution has been suggested. This solution is effective, as long as the real-time traffic is a small portion of the whole WLAN traffic. The solution is based on:

(a) buffer management at the AP

(b) setting backoff value = 0 in the VoIP station(s)

See: http://www.spectralink.com/products/pdfs/SVP_white_paper.pdf

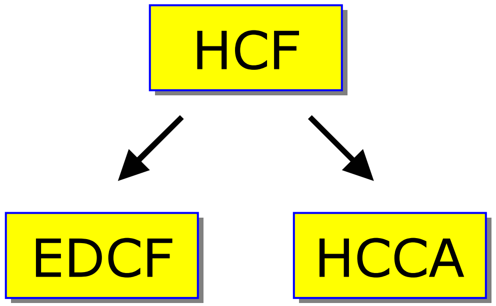
Why 802.11e?

The Point Coordination Funtion (PCF) – although designed for real-time applications – does not offer extensive QoS. The shortcomings of PCF are:

- Differentiation between traffic classes is not possible
- No mechanisms for wireless stations to communicate QoS requirements to the access point
- The contention free period (CPF) length cannot be dynamically changed according to traffic needs
- Different maximum packet lengths cannot be enforced.

IEEE 802.11e

The 802.11e standard defines a new Hybrid Coordination Function (HCF) that offers two modes of operation:



Enhanced DCF (EDCF) is like DCF, but introduces different priority levels for different services.

HCF Controlled Channel Access (HCCA) is a CSMA/CA-compatible polling-based access method (like PCF but without the shortcomings listed on the previous slide).

EDCF

EDCF is based on dividing the traffic in the WLAN into different priority levels. Channel access is controlled by using four differentiating parameters:

- Minimum contention window size (CWmin)
- Maximum contention window size (CWmax)
- Arbitration Interframe Space (AIFS) = variable DIFS
- Transmission Opportunity (TXOP) specifies the time (maximum duration) during which a wireless station can transmit a series of frames.

EDCF (cont.)

The IEEE 802.1D standard defines four Access Categories (AC) for differentiating users that have different priority requirements:

AC	Application
0	Best effort
1	Video probe
2	Video
3	Voice

EDCF (cont.)

The Access Categories can be implemented in the WLAN by using the following parameter values (in addition to using different TXOP values):

AC	CWmin	CWmax	AIFS
0	CWmin	CWmax	2
1	CWmin	CWmax	1
2	(CWmin+1)/2 - 1	CWmin	1
3	(CWmin+1)/4 - 1	(CWmin+1)/2 - 1	1

HCCA

HCCA is based on a Contention-Free Period (CFP) during which the access point uses polling for controlling the traffic in the WLAN, like PCF. The differences between HCCA and PCF are the following:

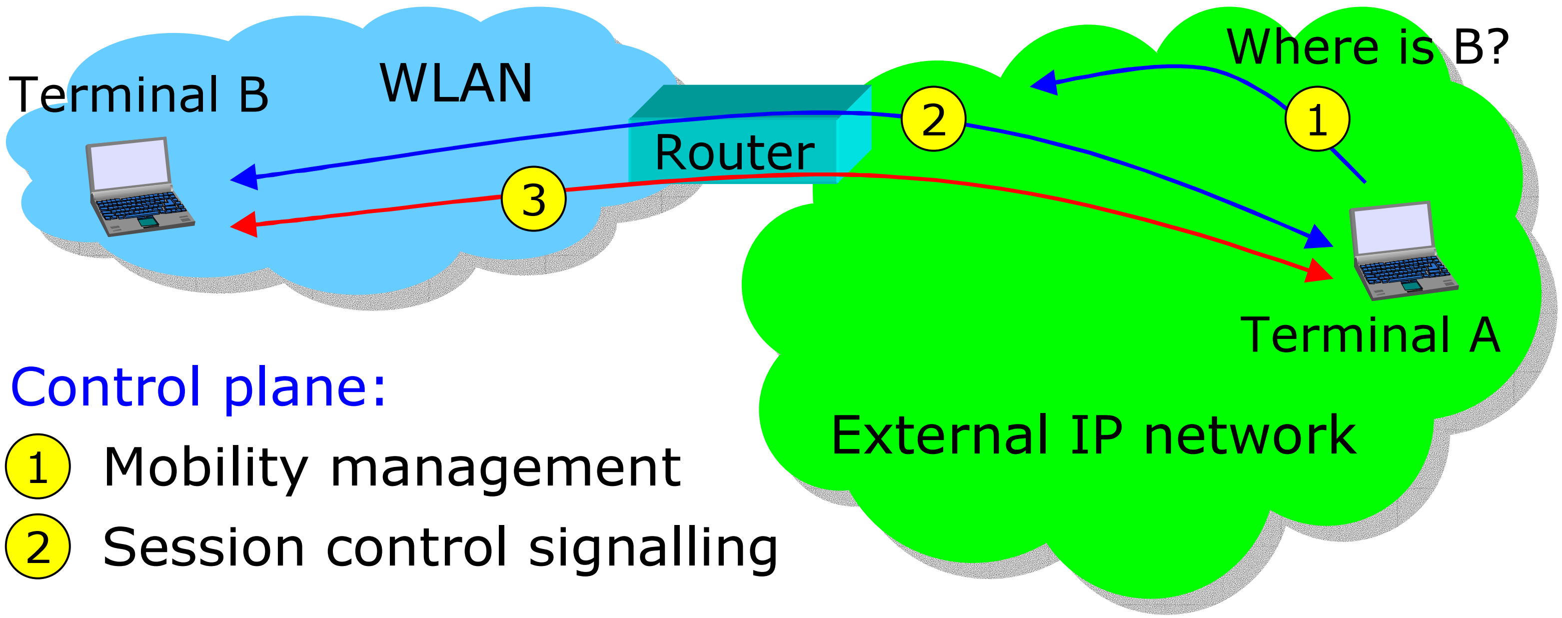
- HCCA can poll stations also during the Contention Period (CP).
- HCCA supports scheduling of packets based on the QoS requirements.
- Stations can communicate their QoS requirements (data rate, delay, packet size…) to the access point.

MAC enhancements in 802.11e

The 802.11e standard also offers MAC enhancements:

- Contention Free Bursts (CFB) allows stations to send several frames in a row without contention, if the allocated TXOP permits.
- New ACK rules. For instance in applications where retransmission cannot be used due to the strict delay requirements, the ACK frame need not be used.
- Direct Link Protocol (DLP) enables communication between wireless stations directly without involving the access point.

VoIP over WLAN: the general picture



Control plane:

1 Mobility management

2 Session control signalling

User plane:

3 QoS, speech coding

The problem of mobility (1)

When a wireless station associates with a WLAN, it is given an IP address (which is stored in the router taking care of the binding between IP and MAC addresses).

However, terminals in the outside world (Internet, another IP subnet on the wired LAN, another LAN or WLAN) do not know this address. Consequently, it is not possible to route VoIP calls (or anything else) to this wireless station.

The problem of mobility (2)

There are at least four ways of resolving this problem:

- Mobility management of 2G/3G mobile networks (not possible before there is seamless integration between WLAN and 2G/3G technology)
- H.323 (ITU-T solution)
- SIP (http://www.ietf.org/rfc/rfc3261.txt)
- Mobile IP (http://www.ietf.org/rfc/rfc2002.txt)

H.323 and SIP also take care of session control signalling (basically giving IP addresses of users to other users).

Voice (speech) coding schemes (1)

Standard PCM (Pulse Code Modulation) produces a fixed bit rate of 64 kbit/s. The encoding/decoding is specified in the ITU-T recommendation G.711.

G.726 specifies an Adaptive Differential PCM (ADPCM) codec which produces various bit rates (16, 24, 32, or 40 kbit/s).

G.729 specifies a speech coder that operates at 8 kbit/s. This is a complex codec based on linear prediction and other advanced concepts.

Voice (speech) coding schemes (2)

Low-bit-rate voice coding is especially important in mobile radio systems (2G and 3G). Two widely used codecs are:

- Enhanced Full Rate (EFR) used in GSM. Although the bit rate is quite low (12.2 kbit/s) the speech quality is surprisingly good.
- Adaptive Multi-Rate (AMR) used in 3G systems, where several bit rates (4.75 ... 12.2 kbit/s) are possible, depending on the channel quality. In fact, AMR at 12.2 kbit/s = EFR.

Voice coding performance

As a general rule, when the bit rate decreases:

- The voice quality decreases (becomes robot-like)
- A certain packet error ratio (PER) causes more severe voice quality degradation.

Efficient voice coding is maybe not so important: When carrying coded voice over IP networks (and especially 802.11 networks) the protocol overhead (especially in the lower layers) is so large that efficient voice coding does not offer substantial capacity improvements.