VoIP in 802.11

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Postgraduate Course in Radio Communications

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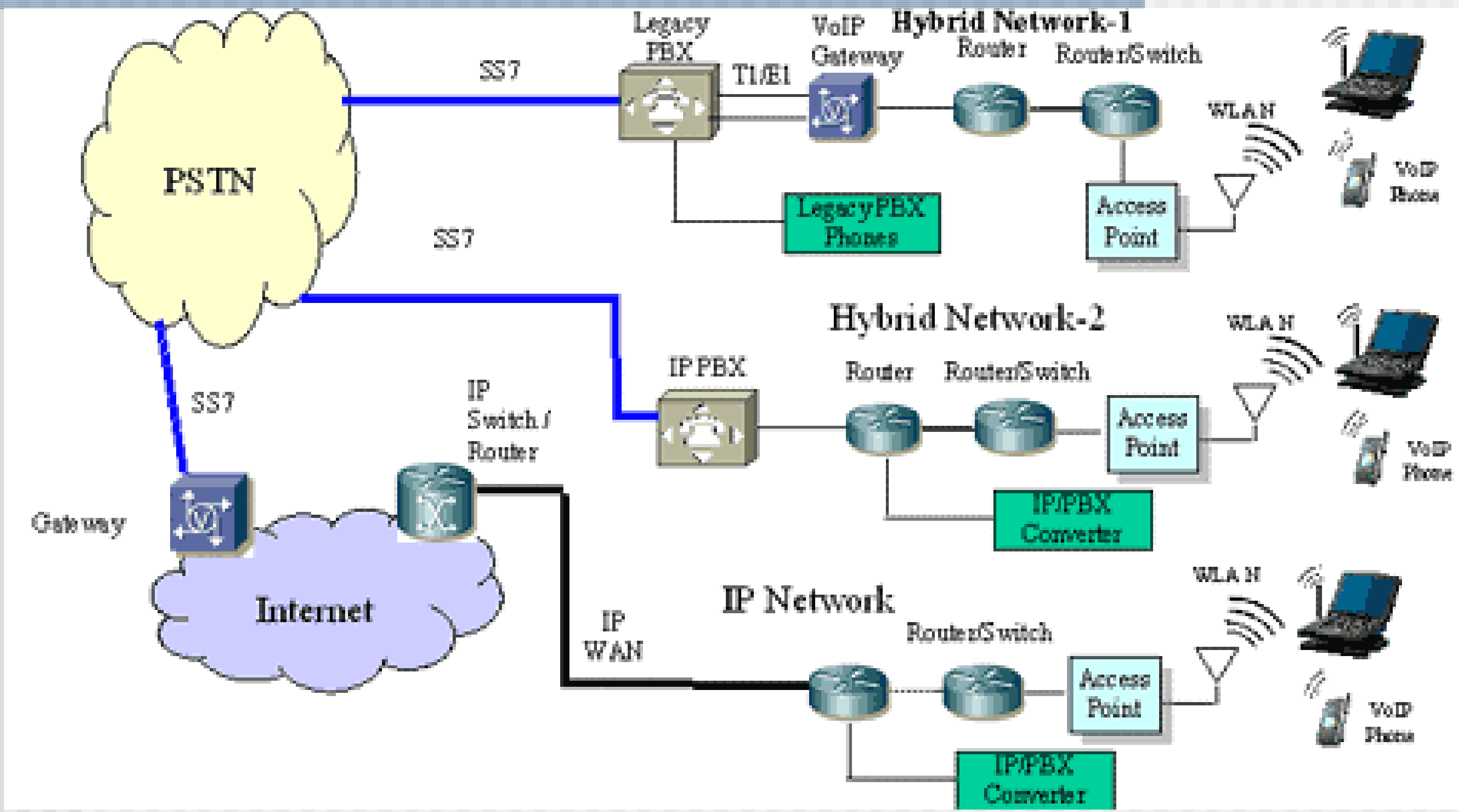
Introduction

- Seamless wireless data and voice communication is fast becoming a reality
- One key capability in the next-generation wireless world will be Voice over Internet Protocol (VoIP) using 802.11 wireless local area networks (WLANs)
- The technology to enable one phone number for broadband wireless data and voice communication is available
- The remaining issues facing handset designers, carriers and service providers as well as enterprise and residential network designers relate to questions of deployment, configuration and network architecture

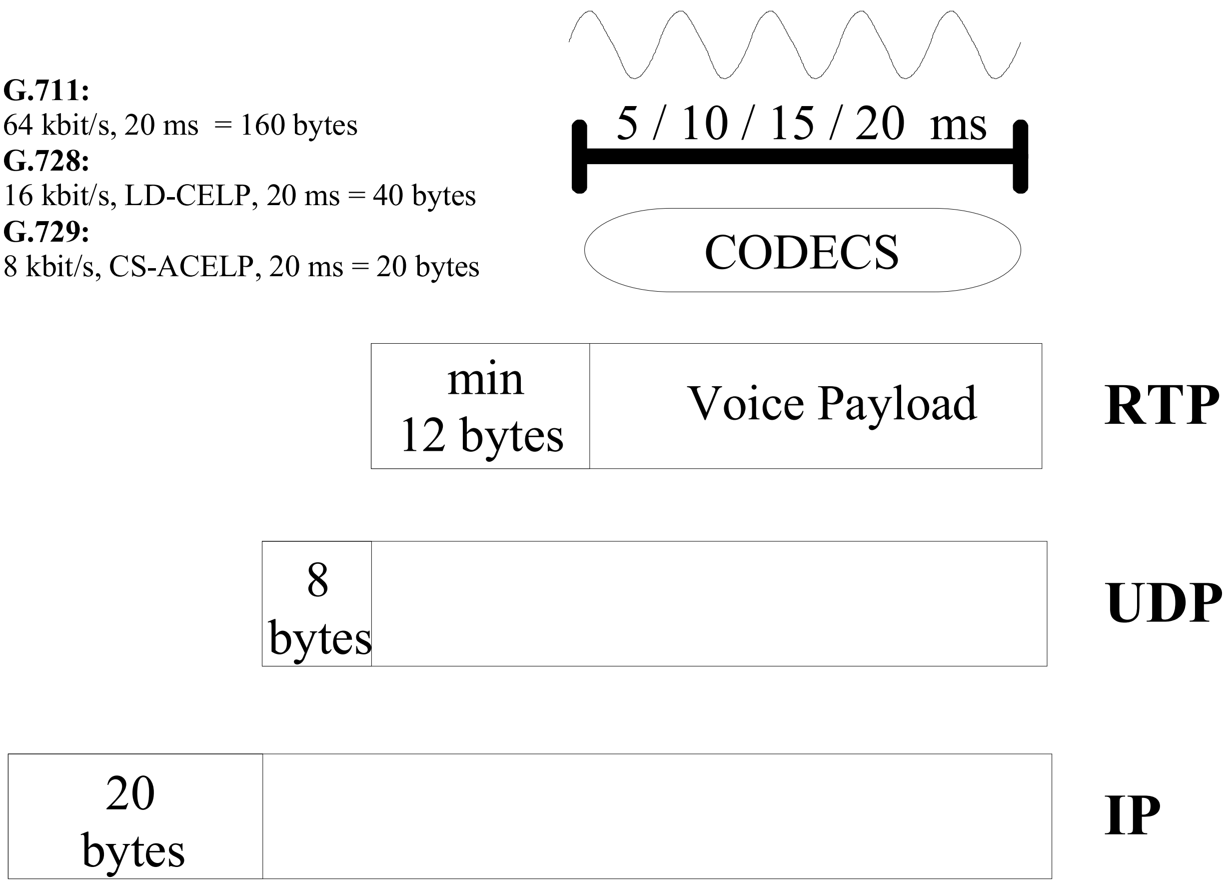
VoIP & WLAN



PSTN
SS7
Legacy PBX
T1/E1
VoIP Gateway
Hybrid Network-1
Router
Router/Switch
WLAN
VoIP Phone
Legacy PBX Phones
Access Point
SS7
Hybrid Network-2
IP PBX
Router
Router/Switch
WLAN
Access Point
VoIP Phone
IP/PBX Converter
SS7
IP Switch / Router
Gateway
Internet
IP WAN
IP Network
Router/Switch
WLAN
Access Point
VoIP Phone
IP/PBX Converter

VoIP & Protocol Stack

G.711:
64 kbit/s, 20 ms = 160 bytes
G.728:
16 kbit/s, LD-CELP, 20 ms = 40 bytes
G.729:
8 kbit/s, CS-ACELP, 20 ms = 20 bytes



RTP = Real Time Protocol (Internet, RFC 1889/1890)

UDP = User Datagram Protocol (Internet, RFC 768)

IP = Internet Protocol [version 4] (RFC 791)

LD-CELP = Low-Delay Code-Excited Linear Prediction

CS-ACELP = Conjugate Structure Algebraic-Code-Excited Linear-Prediction

VoIP & WLAN overhead

- RTP 12 bytes
- UDP 8 bytes
- IP 20 bytes

- 802.11b MAC 34 bytes
- 802.11b PHY with short preamble 15 bytes OR
- 802.11b PHY with long preamble 24 bytes

Admission Control

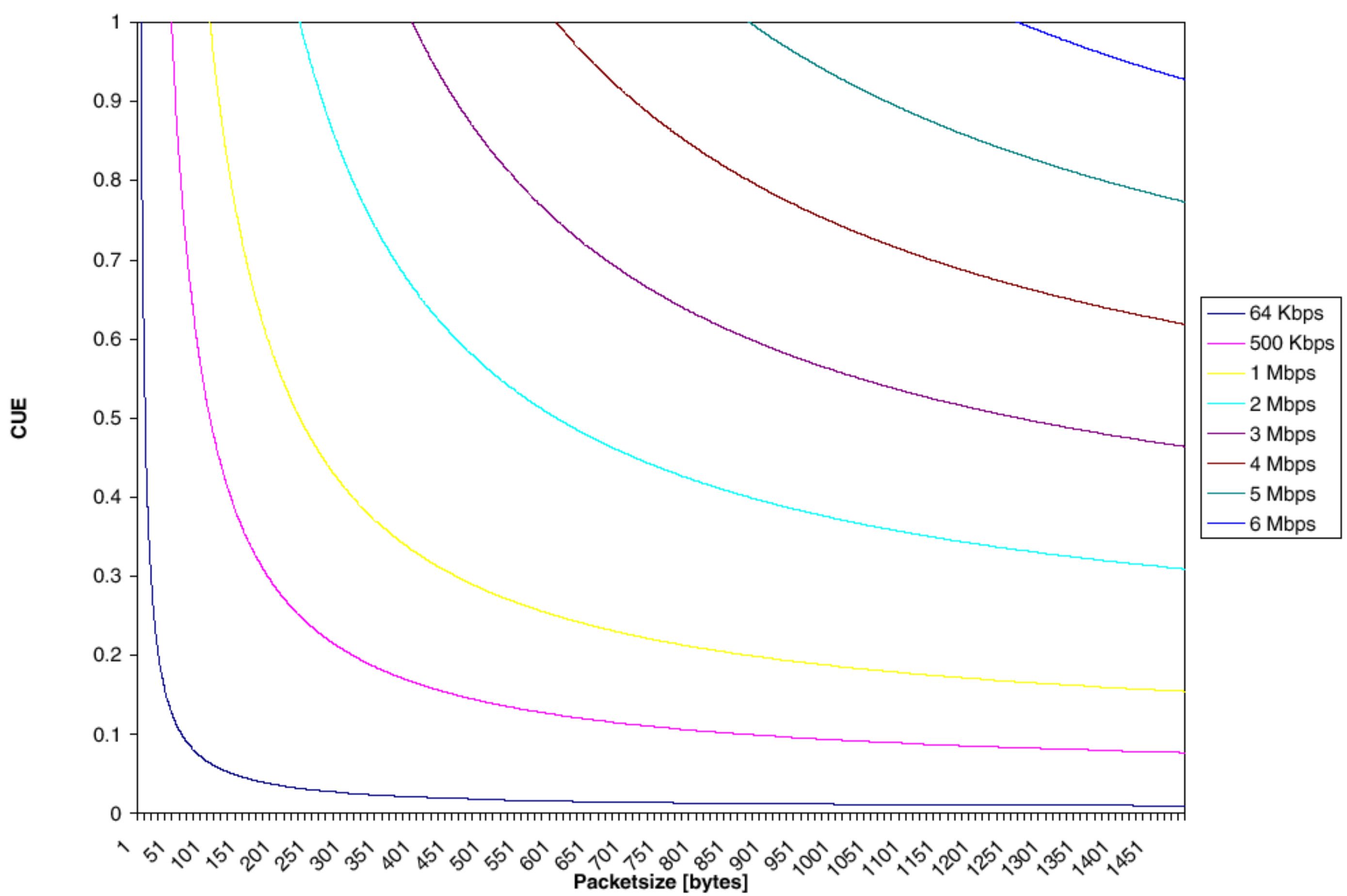
- For VoIP flows, the bandwith and other traffic characteristics of the flow do not change during the lifetime of flow
- For data flows the traffic characteristics may change over time
- When the traffic exceeds the capacity of the wireless network
  → Unacceptable call quality for all ongoing VoIP calls
  (in most cases AP send more traffic than other
  →AP’s traffic is reduced)
  → VoIP flows needs full recources
- Admission control for VoIP flows is necessary, traffic control is sufficient for data traffic

Channel Utilization Estimation

- In 802.11 wireless networks, the channel utilization of a flow and remaining network capacity cannot be measured by bandwith

  For example: In 802.11b fixed overhead per frame transmission is 765 μs at 11 Mbit/s (single client case)
  - 100 byte payload → max. 1193 frames/s → 954 kbit/s
  - 1000 byte payload → 670 frames/s → 5,36 Mbit/s
- Question to answer: "Can the network support one more VoIP flow?"
- Proposal:
  The use of fraction of time needed to transmit the flow over the network as a indicator for network usege of a flow
  → Channel utilization Estimation (CUE)

Packet size vs. Channel utilization



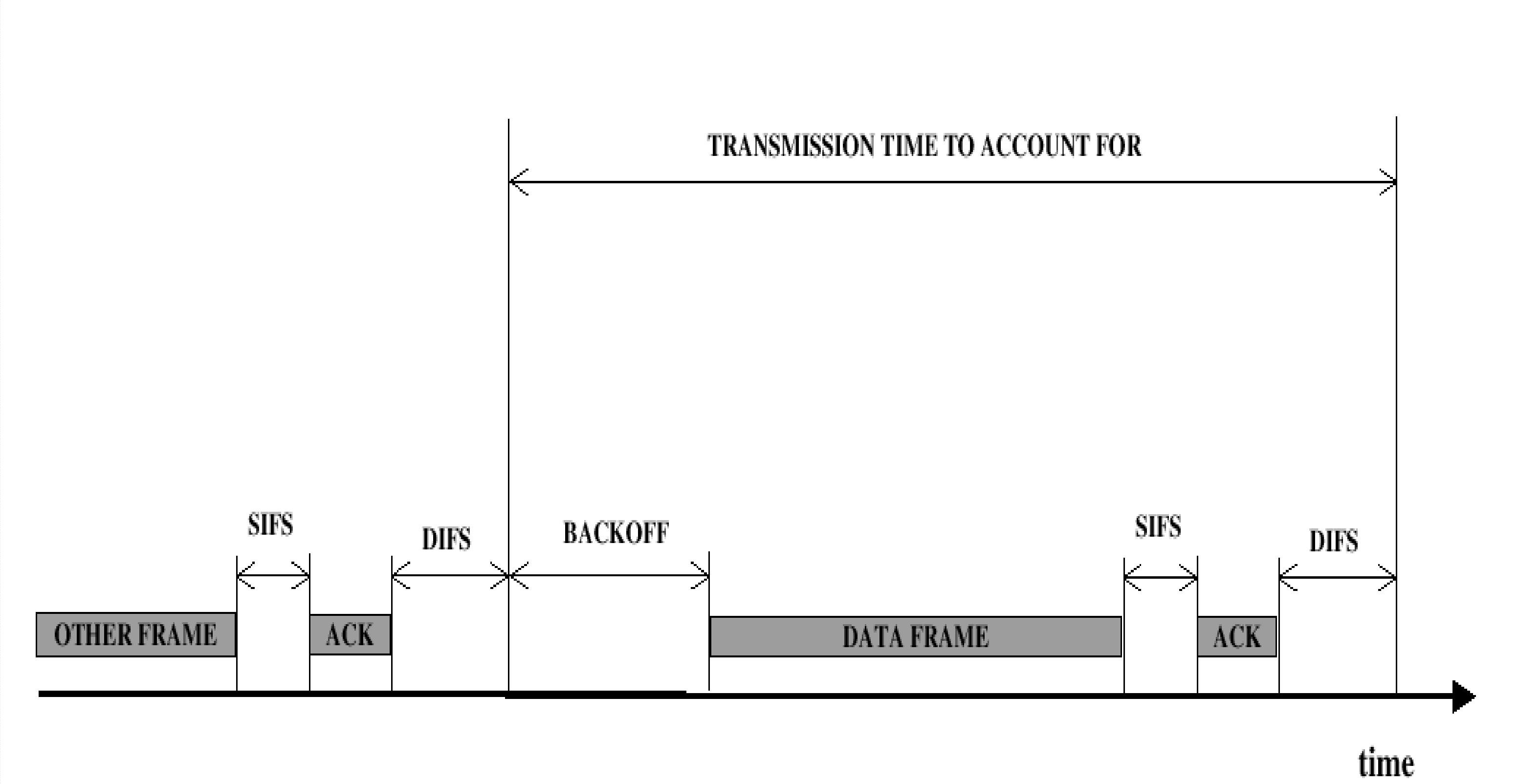
- Single client sending at 11 Mbit/s

CUE

- CUE of the flow is the fraction of time the network is busy transmitting data for that flow
- The sum of the all flows (CUE_{total}) is the fraction of time the network is busy transmitting all flows
- Fully loaded media $CUE_{total} = 1$
- Measuring the CUE in standard DCF MAC scheme:
  - Data frame size b? [bytes]
  - Data rate R? [bit/s]
  - Back-off time?

Part	Time [s]
Data Frame	$192\mu s + b \cdot 8/R$
SIFS	$10\mu s$
ACK	$192\mu s + 14 \cdot 8/R$
DIFS	$50\mu s$

IEEE802.11 CSMA/CA medium access scheme



Using CUE for Admission Control

1. Detecting new flows:

- VoIP streams can be detected monitoring for traffic initiatiate the call (packets to H.323 port or packets containing SIP messages
- New TCP flows can be detected from SYN/ACK bits in TCP packet headers.

2. Calculate CUE_{total}
3. Estimate CUE for new VoIP-flow

- If $CUE_{total} + CUE_{new} < CUE_{totalMAX} \rightarrow$ New VoIP-flow

Bandwidth restrictions for non-VoIP flows may be needed

Voice services in IEEE802.11b:
DCF vs. PCF I

- DCF mode is the fundamental access method of 802.11 MAC sublayer and it uses CSMA/CA
  -> supports data services
  -> large/unbounded delay when load is high
- PCF mode uses polling and offers a "packet-switched connection-oriented service"
  → + well suited for telephony traffic
  → + CBR or VBR mode
  → - support for PCF is not so commonly available

```
DCF     = Distributed Coordination Function
PCF     = Point Coordination Function
MAC     = Media Access Control
CSMA/CA = Carrier Sense Multiple Access
          with Collision Avoidance
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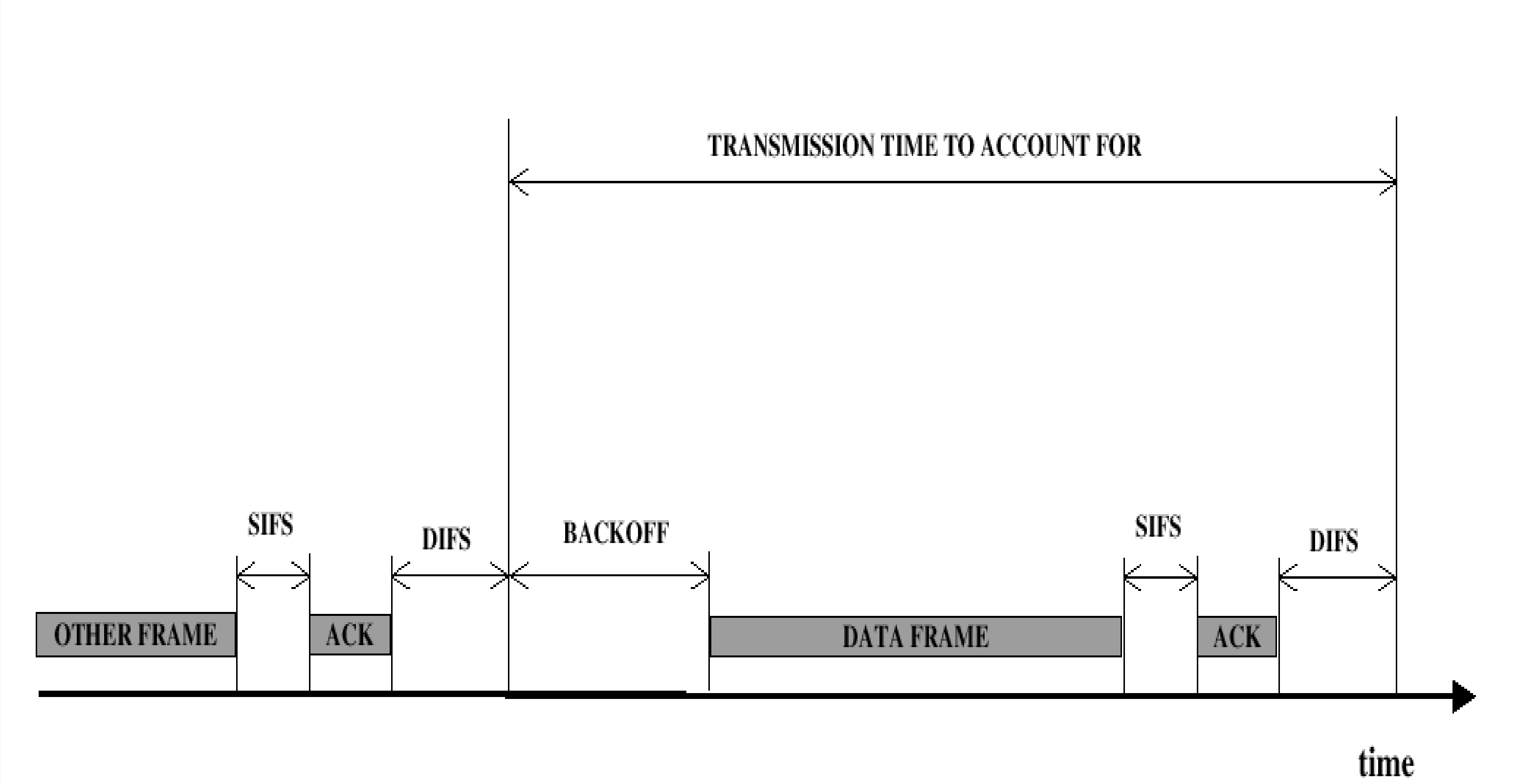
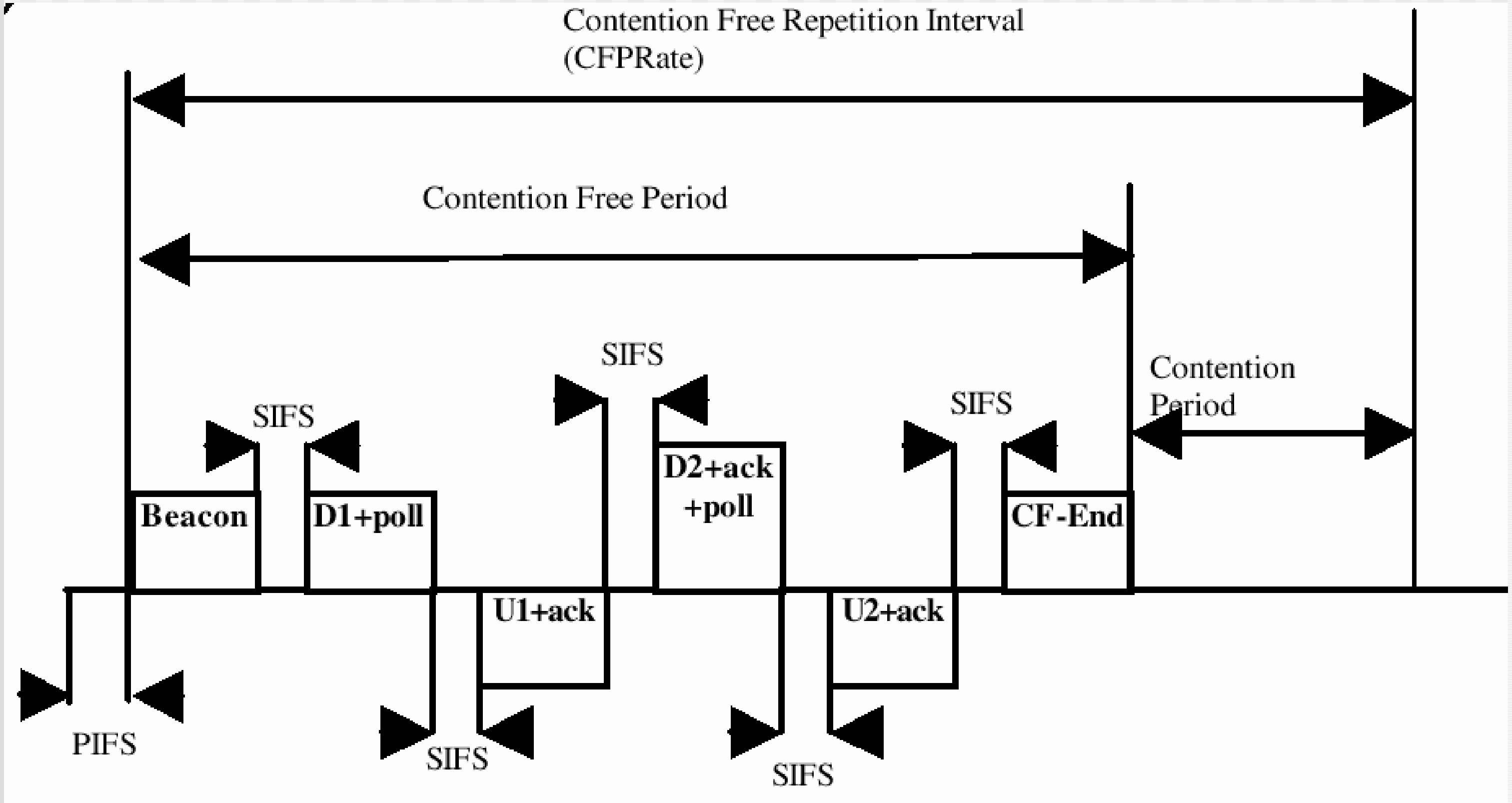
DCF vs. PCF II



SIFS = Short InterFrame Space

PIFS = Point Coordination Function - InterFrame Space

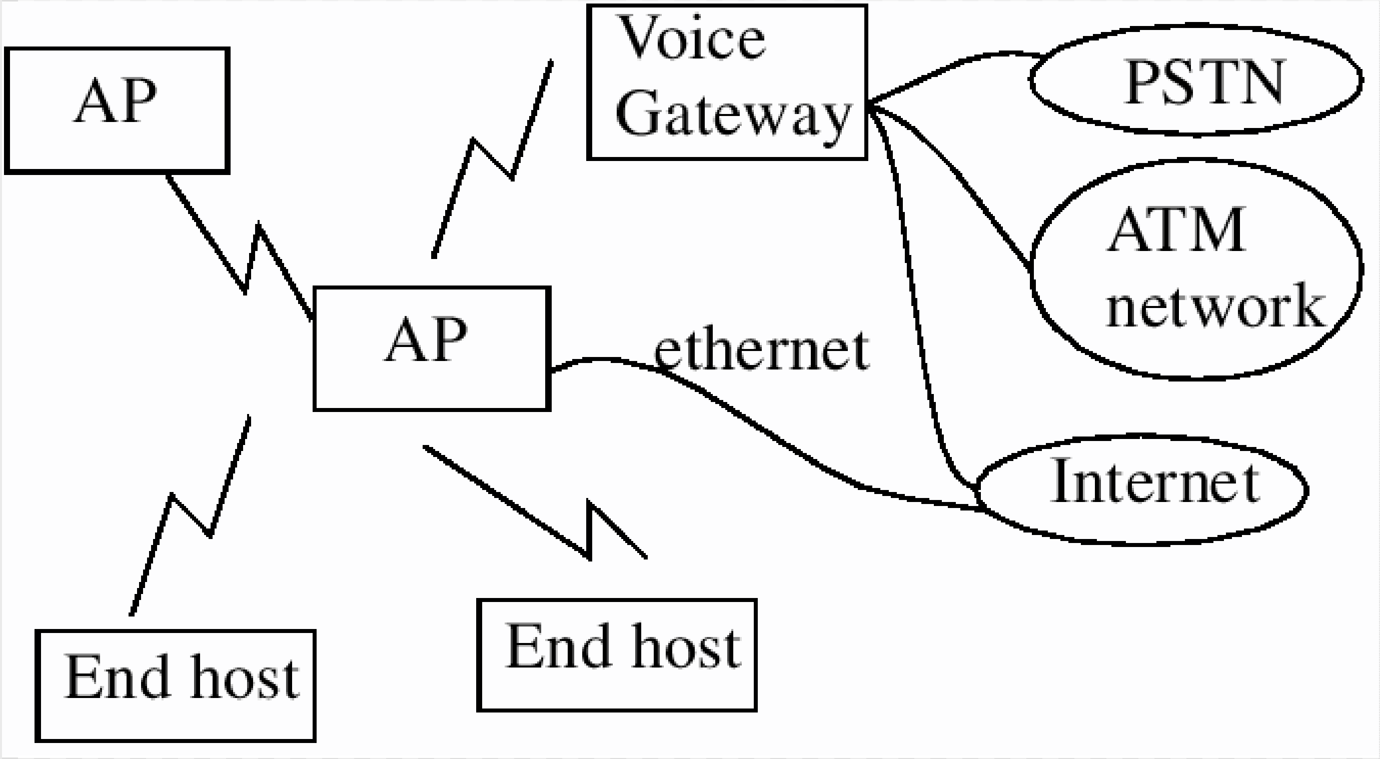
DCF vs. PCF III



SIFS = Short InterFrame Space

PIFS = Point Coordination Function - InterFrame Space

Network Architecture I



Network Architecture II

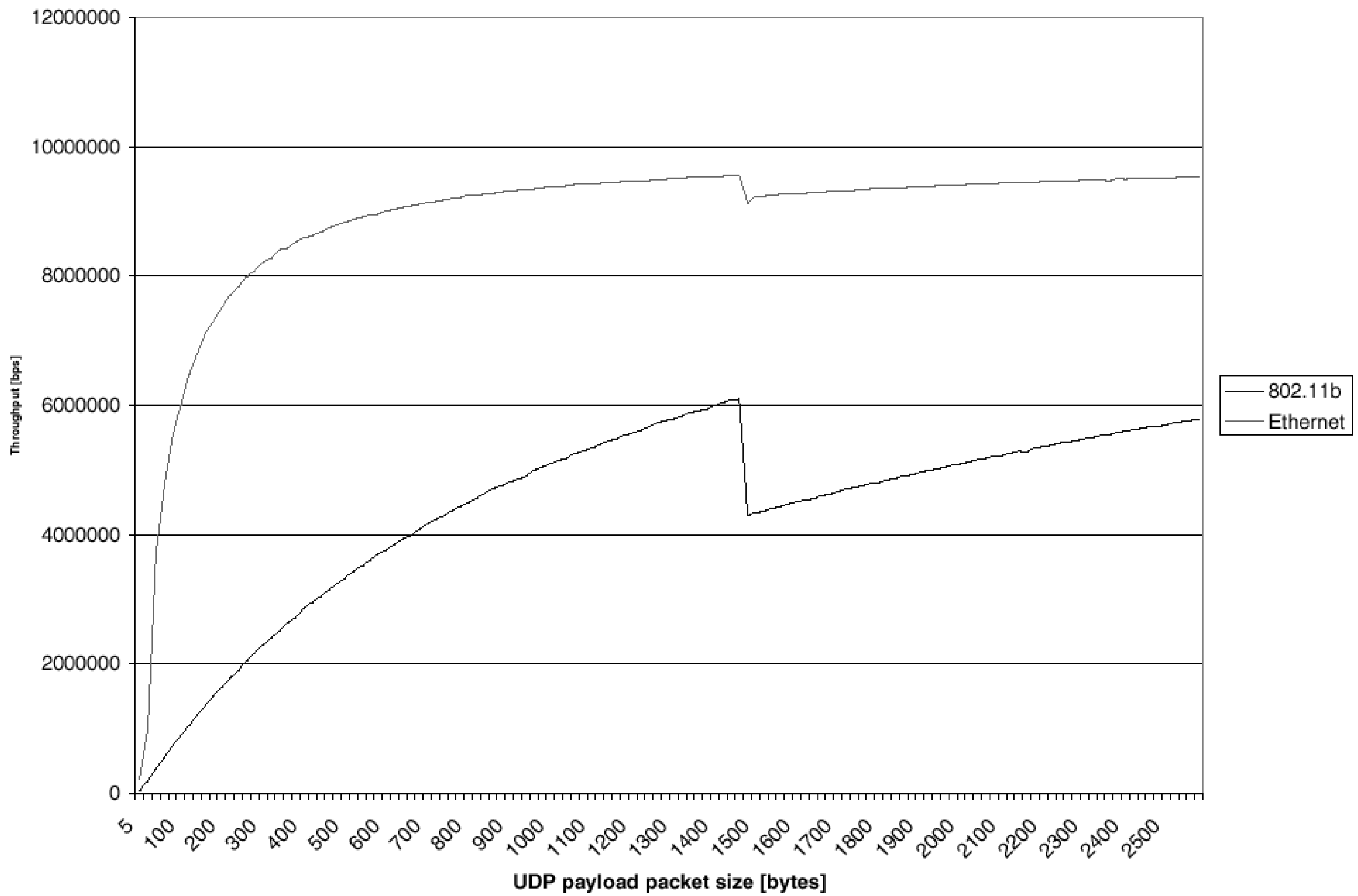
- Voice gateway converts 802.11 protoc stack to:
  - PCM voice for use on the PSTN
  - Voice over ATM Adaptation layer for ATM-networks
  - Voice over Real-time Transport Protocol for IP networks
- PCF-mode between AP and voice gateway

VoIP throughput in IEEE 802.11b

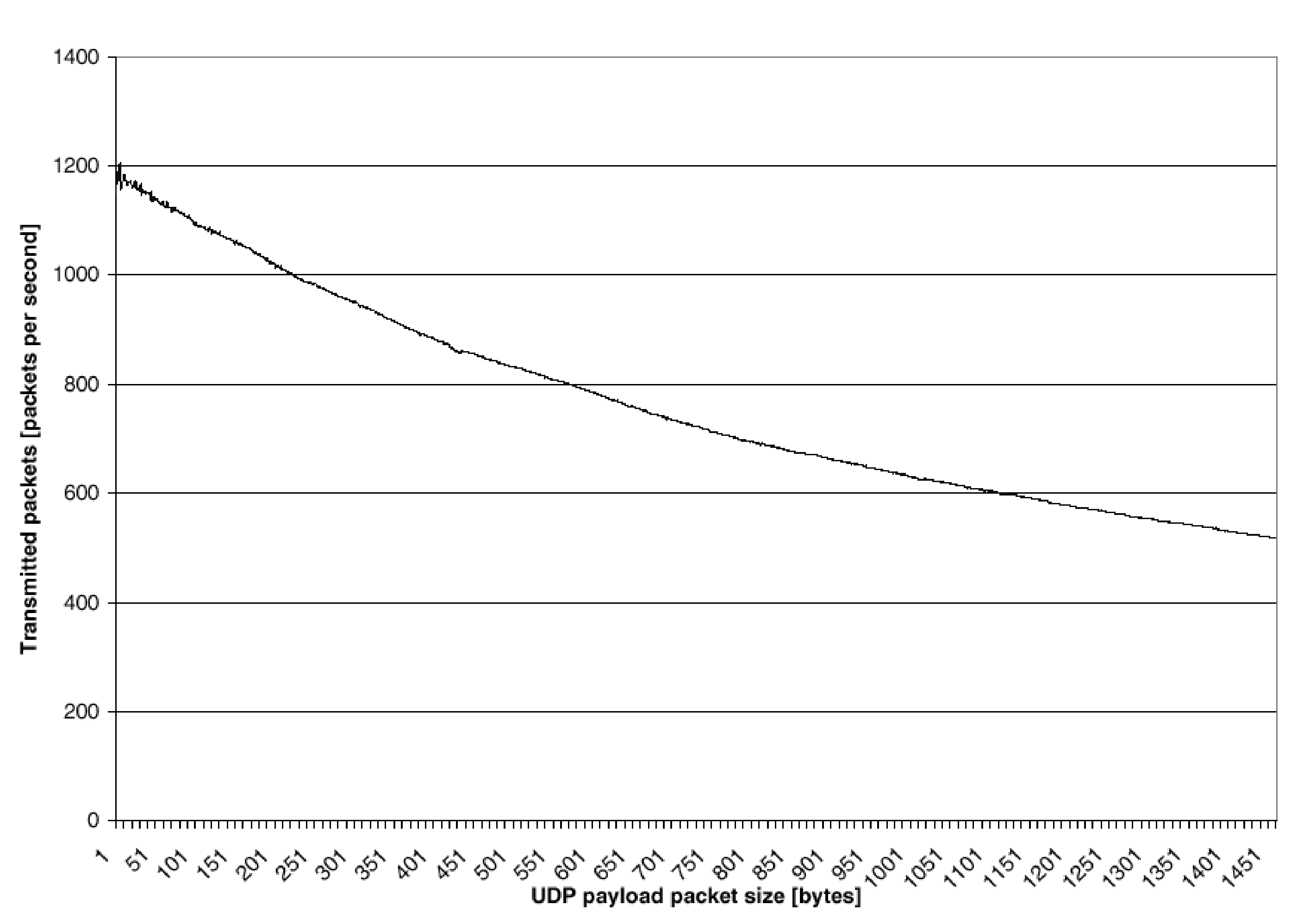
Experimental Stydy - Garg & Kappas [2]

- SETUP:
  - 8 clients, all PC:s, win2000
  - One AP connected to IEEE 802.3 (ethernet) LAN
  - AP & clients in same room without physical obstacles
    → no frame losses due to weak signal strenght
    → no hidden terminal problems
  - Traffic: UDP packets → accurate estimate of the actual bandwith that is available in the network
- 1) Single client case
- 2) Multiple UDP senders

Throughput – one client



Transmitted packets/s – one client



Maximal number of VoIP connections using ITU G711 A-LAW codec

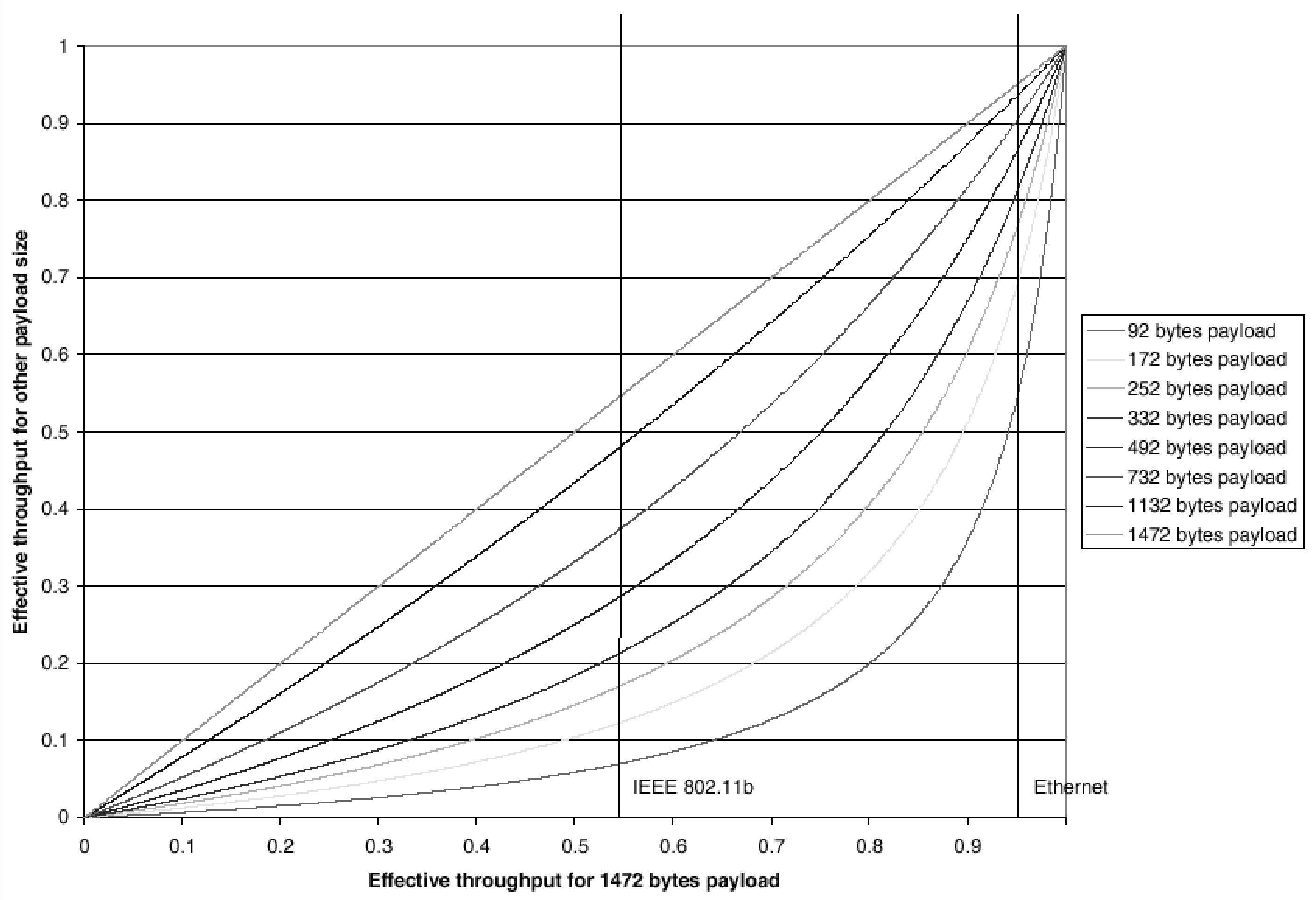
- Frames sent out every 10 ms → 92 bytes payload (12 bytes from fixed RTP header)

Audio (ms)	G711	G729	G723
10	6	7	
20	12	14	
30	17	21	21
40	21	28	
50	25	34	
60	28	41	42
70	31	47	
80	34	54	
90	36	60	61
100	39	66	

VoIP conn.	UDP Throughput
0	6.06 Mbps
1	5.15 Mbps
2	4.26 Mbps
3	3.28 Mbps

Each VoIP connection consisting 2 UDP streams (a’ 74 kbit/s) reduces the throughput of the other UDP-sender app. 900 kbit/s.

Effective throughput of network



Effective throughput of a network for 92, 172, 252, 332, 492, 732, 1132 and 1472 bytes (packet sizes used by a G711 a-Law codec with audio data length 10ms, 20ms, 30ms, 40ms, 60ms, 90ms, 140ms, respectively) of payload as a function of the effective throughput for 1472 bytes of payload. The vertical lines mark the values for IEEE 802.11b and Ethernet.

Conclusions

- VoIP call quality is fine as long as the network throughput limit is not exceeded (packet loss, delay & jitter acceptable)
  → admission control needed
- Payload size affect the throughput of WLAN network
  → + increasing audio data length per packet
  → -- delay, quality of voice when loosing a packet
  → → Optimal payload size
- PCF mode can be used to carry telephone traffic
  -> + (almost) fixed delay, optimal payload size

References

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2. An experimental study of throughput for UDP and VoIP traffic in IEEE 802.11b networks Garg, S.; Kappes, M.; IEEE Wireless Communications and Networking, Volume: 3, Pages:1748 – 1753, 16-20 March 2003.
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4. http://www.iec.org/online/tutorials/ti_voip_wlan/

HOMEWORK

Let us assume that compressed digital speech with a bit rate of 12 kbit/s is sent from a VoIP client to AP (and then to another network)
(IEEE 802.11b network, 11 Mbit/s, DCF-mode, RTS/CTS option is not used)

a) What is the packing delay at the sending side if voice payload 48 bytes?
b) What is the packing efficiency when the packing delay is not allowed to exceed 10 ms?
c) Approximate (and give short explanation) how many VoIP flows there can be at the same time in one cell in case b). No other traffic is assumed.