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VoIP over Wireless
Opportunities and Challenges

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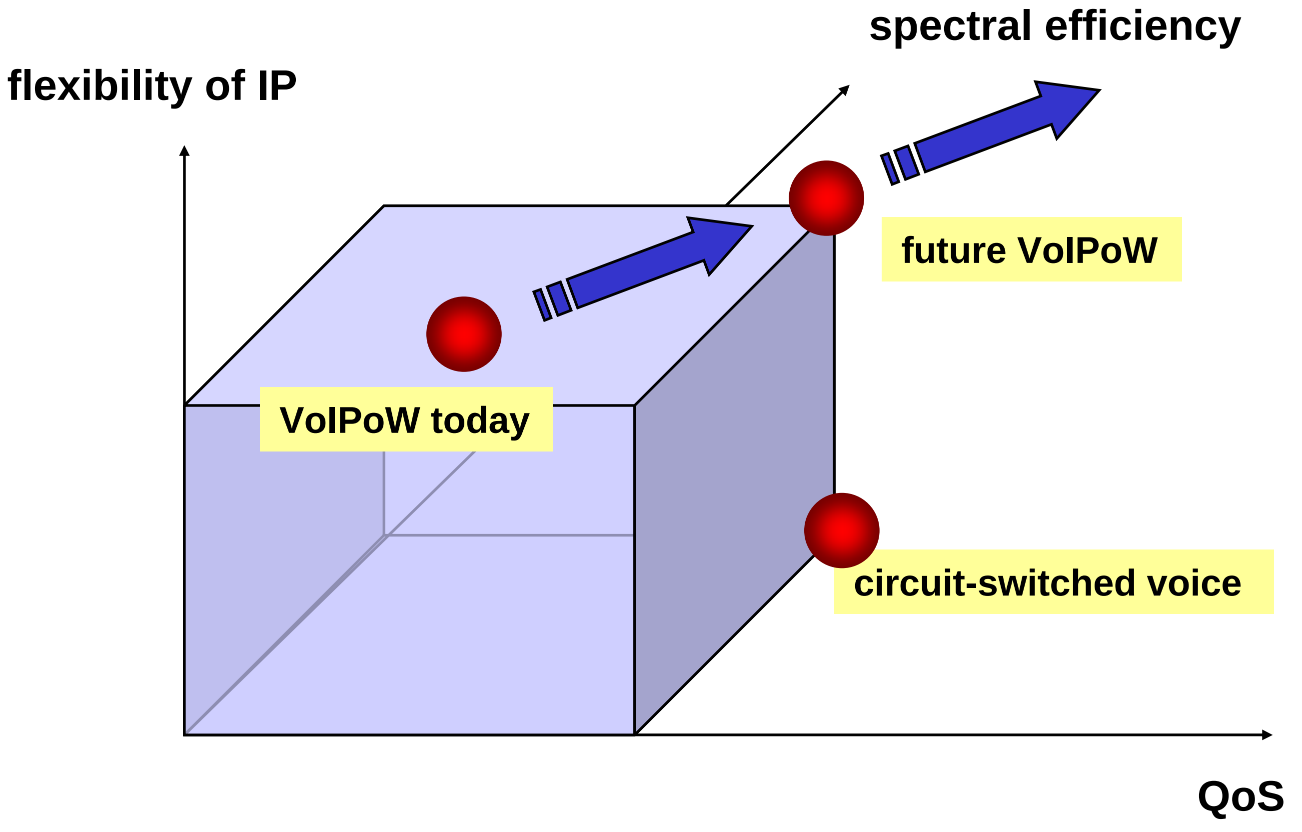
Voice-over-IP over Wireless (VoIPoW)

G723.1
UDP
IMS
QoE
SIP
H.323
G729
HSDPA
Play-out buffer
Skype
WiMAX
RTP
G711
IP
WiFi
Header Compression
UMTS
Codec
Packet Loss Concealment
AMR
LTE
PTT
QoS

Overview

- ▷ Why Voice-over-IP over Wireless?
  - Motivation and Advantages
  - Current problems and challenges
- ▷ VoIP transmission in different radio access technologies
  - IEEE802.11 WLAN
  - IEEE802.16 WiMAX
  - UMTS
- ▷ An outlook to the future/ current research
- ▷ Conclusion

Why VoIP over Wireless?



VoIP and VoIPoW

- Aspects of VoIP
  - signaling and connection management (SIP, H.323, Skype, IMS…)
  - transport protocol (RTP)
  - voice codec
  - play-out buffer
  - packet loss concealment (FEC, …)
  - jitter, packet loss, delay

must fit to each other

- Additional aspects of VoIPoW
  - mobility management (MobileIP, VHO, IMS, …)
  - properties of radio transmission
    - high bit error rate
    - time-variant channel
    - limited, expensive bandwidth
  - different radio access technologies

Problems, Challenges, Solutions

- Desired: high spectral efficiency
  - problem: small packets, large header
    - RTP/UDP/IP header avoidable through header compression
  - problem: MAC layer overhead
    - frame aggregation
  - problem: no delivery of erroneous packets
    - voice codec could deal with rare bit errors
    - MAC/UDP require correct packets
    - higher SIR, more robust transmission, more retransmissions

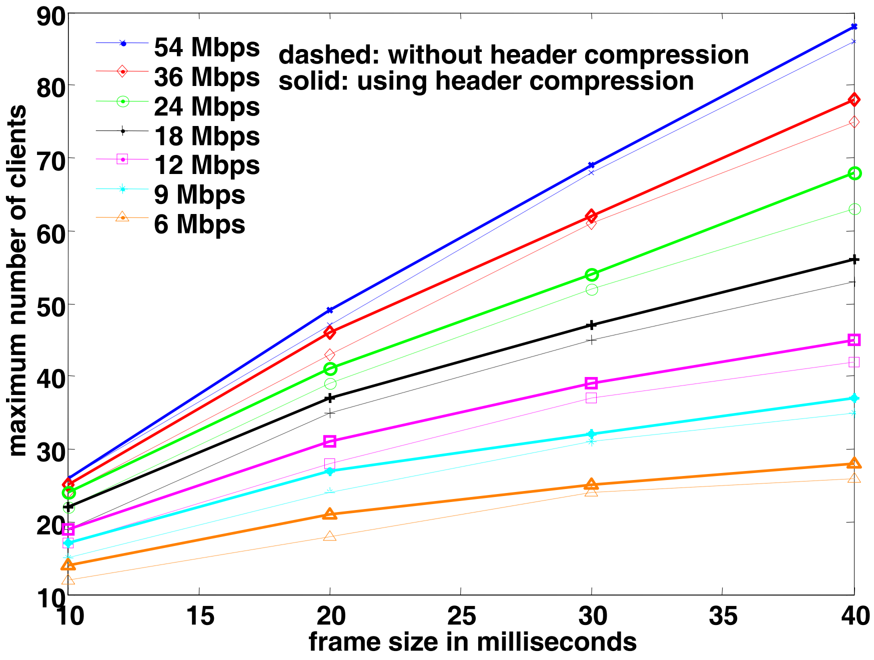
- Desired: low packet loss, delay, and jitter
  - problems:
    - retransmissions
    - random access/medium access
    - scheduling
    - time-variant channel quality
  - solutions:
    - play-out buffer, adaptive codec, packet loss concealment

VoIP over Wireless LAN

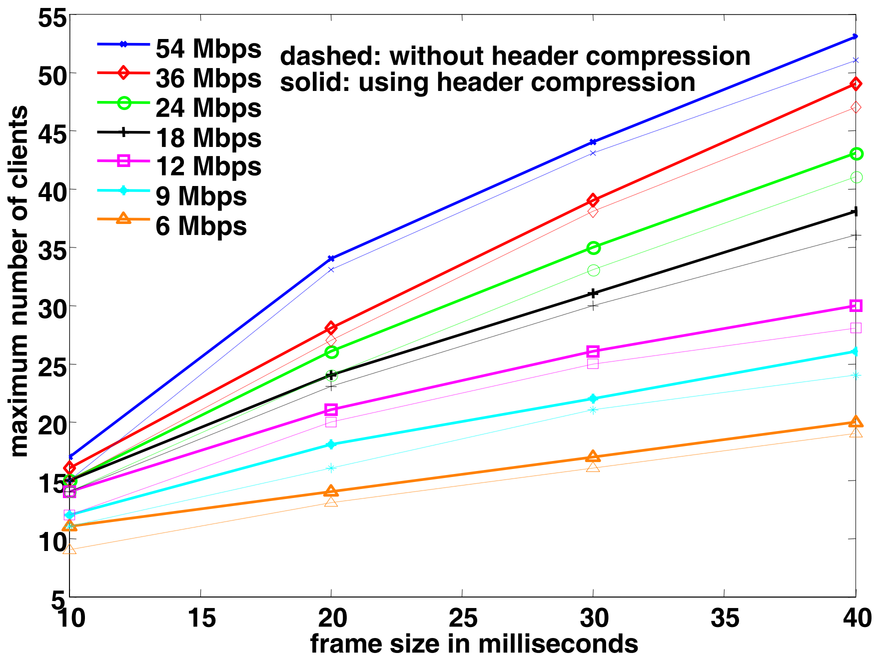
- IEEE802.11abg
  - random access on up- and downlink
  - no service differentiation
  - bad spectral efficiency
  - alternative: polling with PCF (point coordination function)
- IEEE802.11e
  - service differentiation
  - dedicated resource allocation with HCCA
    (Hybrid Control Function Controlled Channel Access)
- Header compression is possible but not used
- Future challenges
  - admission control
  - adaptive contention parameters

VoIP over IEEE802.11g/e with Header Compression

IEEE802.11g
without service differentiation



IEEE802.11e
with service differentiation



- ▷ Contention parameters for VoIP support decrease VoIP capacity
  - adaptive contention parameters
- ▷ Small benefits from header compression

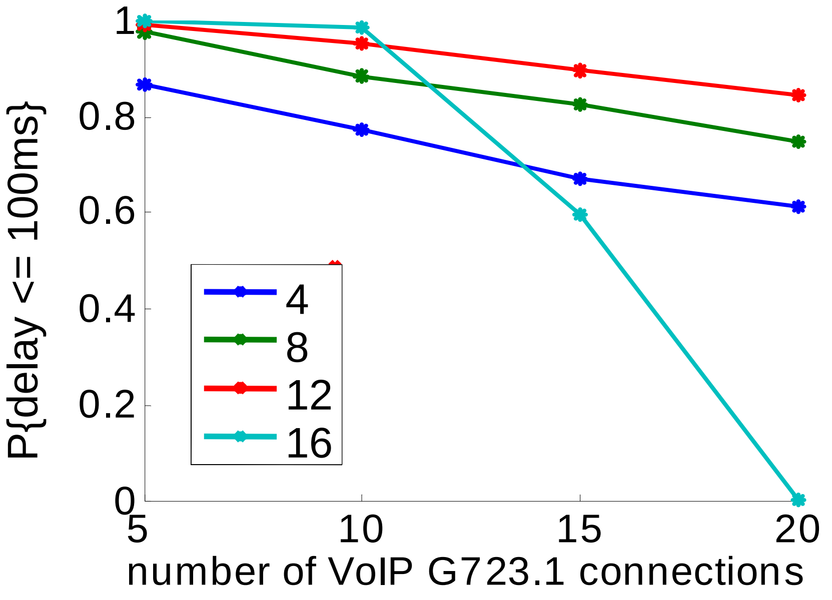
VoIP over WiMAX (IEEE 802.16)

- ▷ Possible scheduling services in WiMAX
  - UGS (Unsolicited Grant Service)
    - essentially a dedicated channel
    - no support for silence suppression on uplink
  - rt-PS (real-time Polling Service)
    - regular dedicated bandwidth request opportunities
    - support for silence suppression on uplink
  - BE (Best Effort)
    - not intended for VoIP
    - contention based bandwidth requests
    - collision free data transmission
    - introduces delay and jitter
- ▷ Problem: Services intended for VoIP (UGS, rt-PS) require detailed traffic characteristics and provide detailed QoS
  - VoIP e.g. Skype transmitted over BE

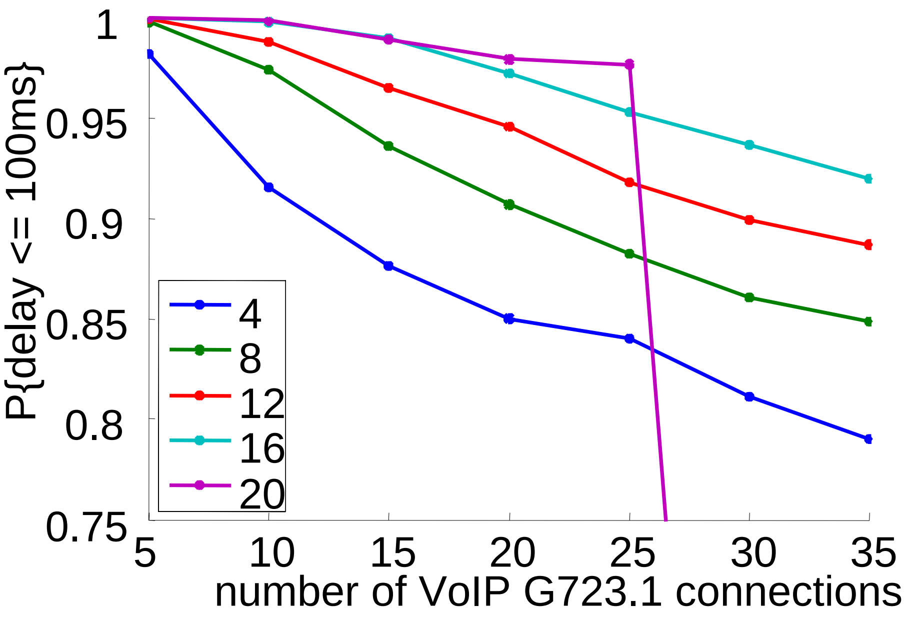
VoIP over Best-Effort Connections in Fixed WiMAX

- Performance of VoIP connections over BE service
- No background traffic, no packet loss, no header compression
- 5MHz TDD
- G723.1 Codec: 480bit every 30ms

10ms frame, QPSK, ½ code rate



5ms frame, 16QAM, ¾ code rate

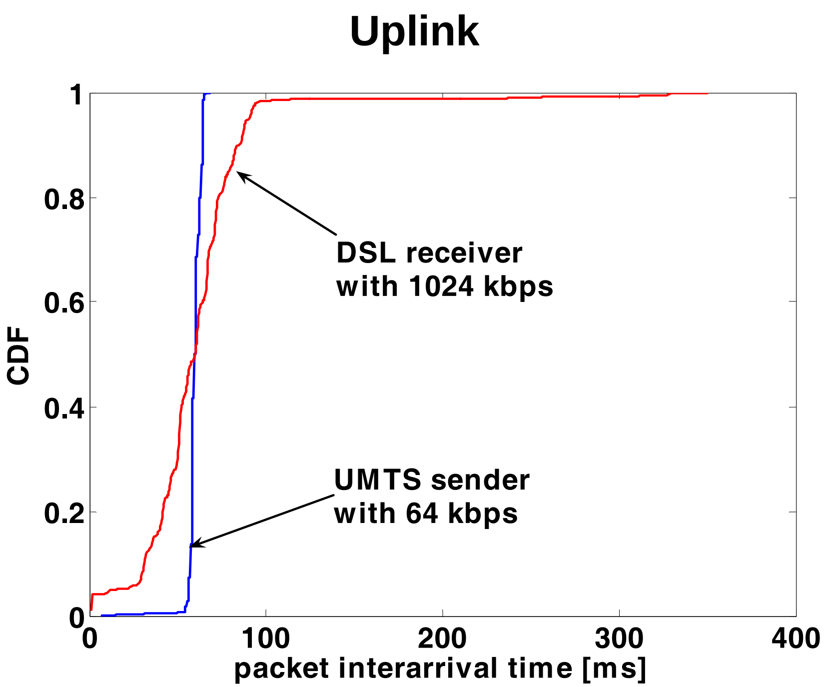


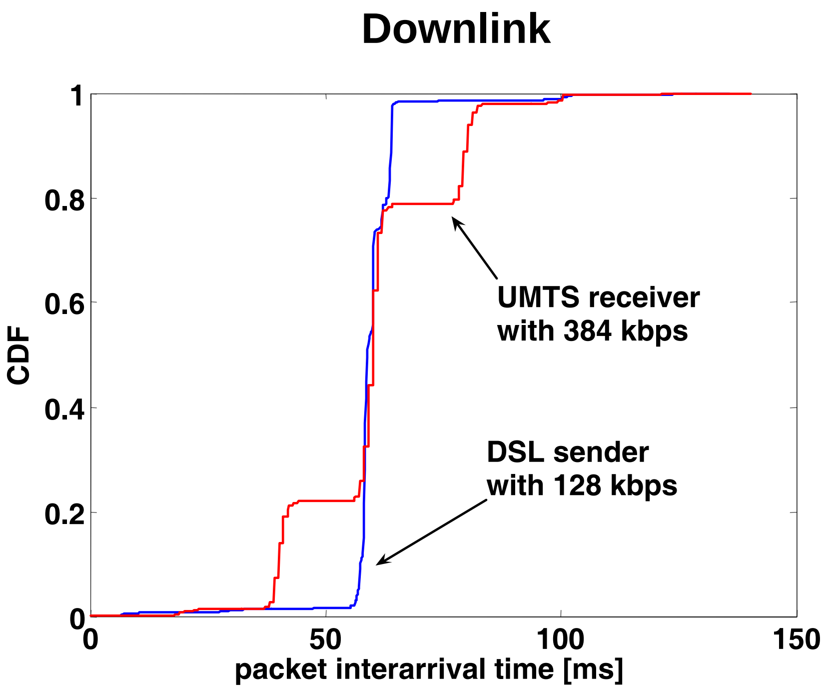
VoIP over UMTS

- Today:
  - Typical: circuit-switched voice over dedicated channels using AMR codec (Adaptive Multi-Rate)
  - VoIP transmission as "normal" data traffic on DCH/HSDPA
    - typically no service differentiation
- Future:
  - IMS, special dedicated channels for VoIP
  - Special support for VoIP over HSDPA/HSUPA? Scheduling disciplines
  - VoIP in UTRA LTE
    - enhanced VoIP capacity by enhanced transmission techniques?
- CDMA2000 1x EV-DO Rev A
  - similar to HSDPA/HSUPA
  - special support for VoIP

Skype over UMTS

- T. Hoßfeld, A. Binzenhöfer, M. Fiedler, K. Tutschku, *Measurement and Analysis of Skype VoIP Traffic in 3G UMTS Systems*, IPS-MoMe 2006
- iLBC codec: 108 Byte voice packet with every 60 ms





- considerable jitter
- PESQ ~2.2 instead of ~3 in bottleneck LAN with 64 kbps

- packet inter-arrival time deterministic
- PESQ ~2.5 instead of ~3 in bottleneck LAN with 128 kbps

Outlook to the Future

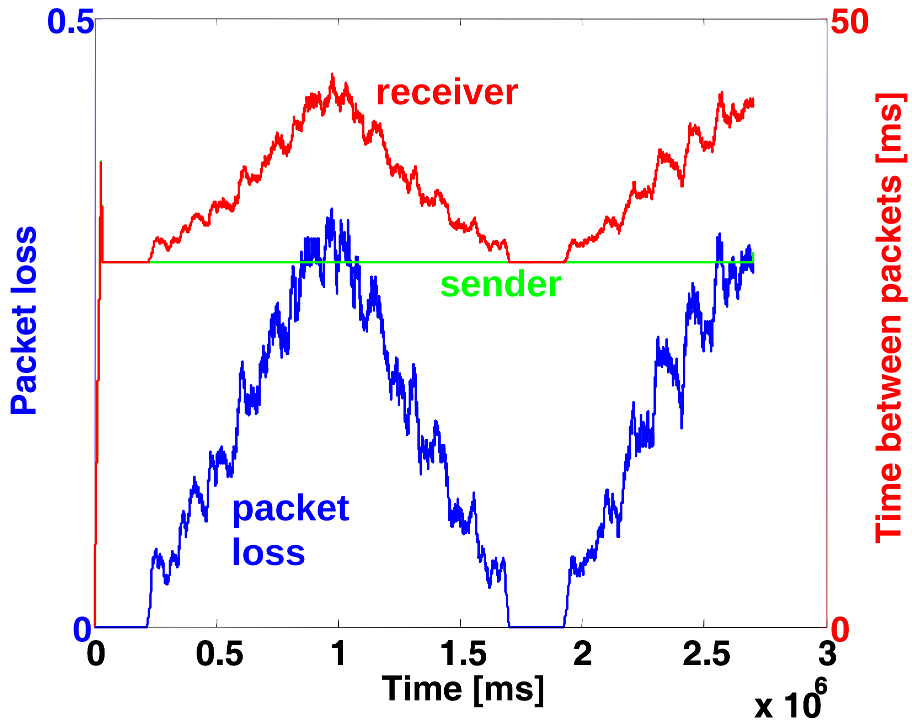
- ▷ Development of VoIPoW
  - current codec optimized for circuit-switched data
  - development of special codecs for VoIPoW
  - differentiated packet dropping

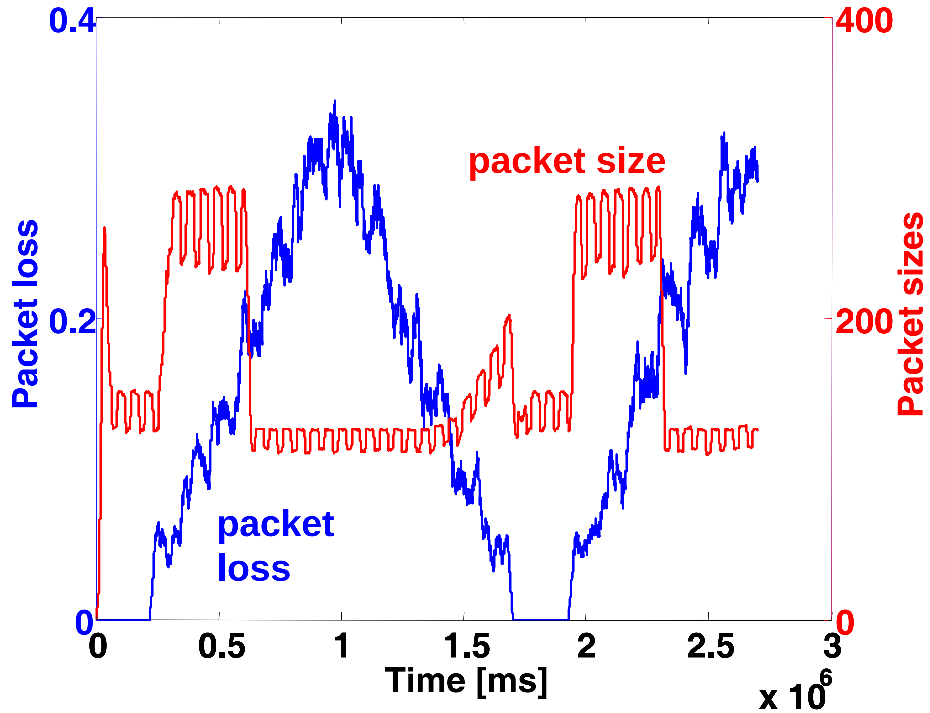
- ▷ Challenges and opportunities for VoIPoW
  - adaptive modulation and coding
  - channel-aware scheduling
  - frequency-selective scheduling
  - enhanced antenna techniques

  - multi-hop networks
  - heterogeneous networks

Skype: Adaptive Codec

- ISAC codec with artificial time-variant packet loss
- Packetization independent of packet loss
- Variable bit rate by increasing packet size, i.e. more audio data





Scenario: VoIP over HSDPA

- G.711 codec: 64 kbps 160bytes per 20 ms
- Performance of different schedulers

Maximum CQI Scheduler
- optimizes throughput
- channel-aware
- starvation, unfairness

Proportional Fair Scheduler
- optimizes throughput considering
- long-term throughput fairness
- channel-aware

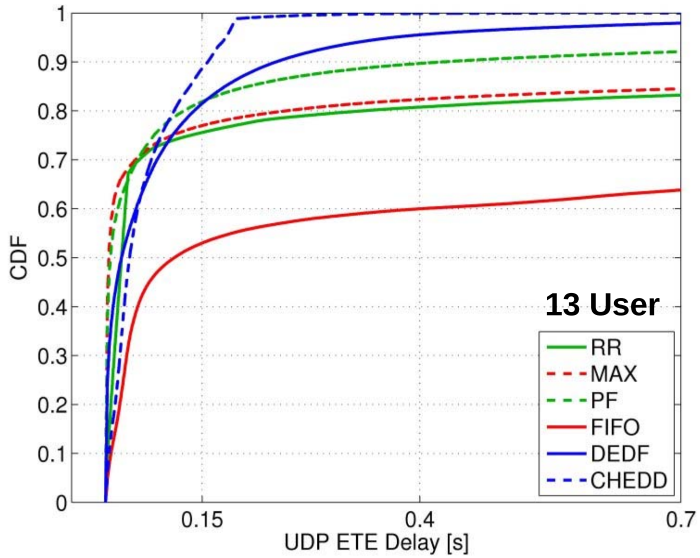
Round Robin
- optimal short-term time fairness
- channel-unaware

FIFO
- First In First Out
- common buffer
- channel-unaware

DEDF Scheduler
- Dynamic-Earliest-Deadline-First
- considers buffering time
- channel-aware
- optimizes delay

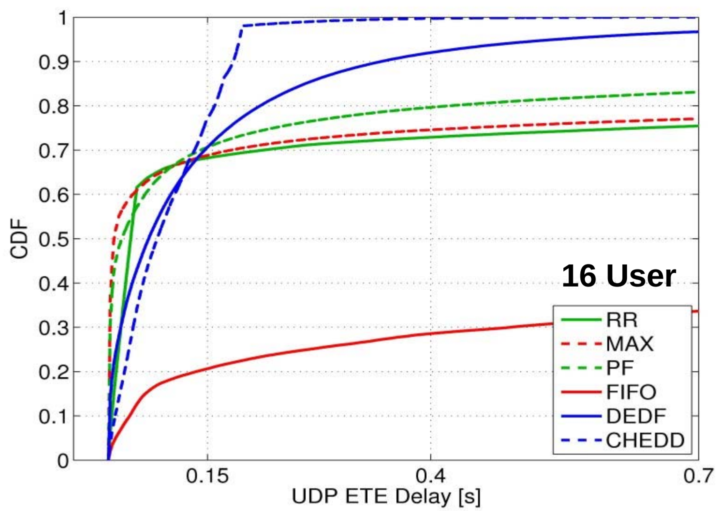
CH-EDD Scheduler
- Channel-Dependent-Earliest-Due-Date
- considers buffering time
- channel-aware
- drops packet after deadline

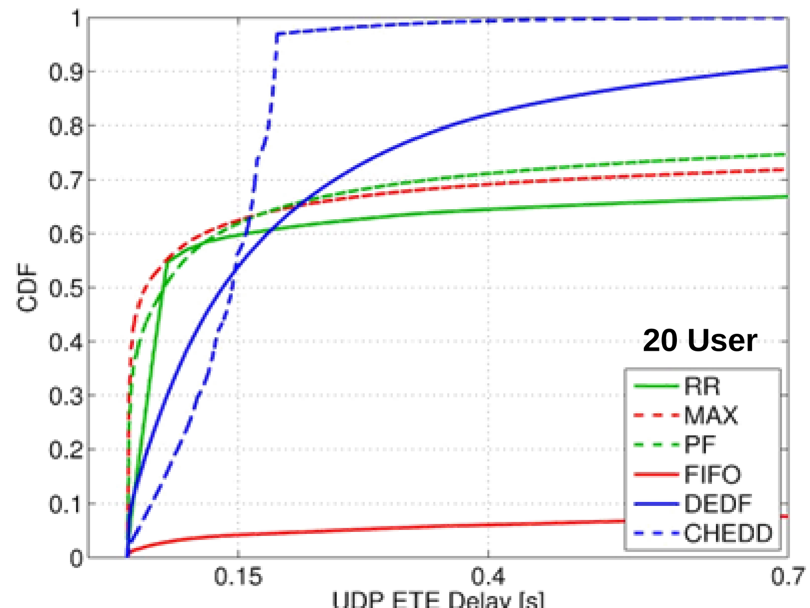
VoIP over HSDPA



packet dropping probability

User	MAX	PF	DEDF	CH-EDD
13	**1.81 %**	**< 1 %**	**< 1 %**	**2.09 %**
16	**6.75 %**	**1.90 %**	**< 1 %**	**3.99 %**
20	**14.06 %**	**7.48 %**	**< 1 %**	**7.76 %**





Conclusion

- Situation today
  - circuit-switched voice is optimized for QoS and spectral efficiency
  - little/no support for VoIP in cellular networks
  - VoIPoW is VoIP over WLAN

- Drivers for VoIPoW in cellular networks are
  - all-IP infrastructure, IMS
  - vertical handover
  - possibilities of packet-switched radio transmission

- VoIP over Wireless will replace circuit-switched voice in the future

- Future challenges and opportunities
  - enhanced packet-switched radio transmission
  - multi-hop
  - development of VoIP optimized codecs
  - charging