VoIP over Wireless
Opportunities and Challenges

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Voice-over-IP over Wireless (VoIPoW)

- G723.1
- QoE
- IMS
- UDP
- Skype
- HSDPA
- SIP
- G729
- WiFi
- Play-out buffer
- WMAC
- IP
- G723
- Header Compression
- UMTS
- Codec
- LTE
- Packet Loss Concealment
- AMR
- RTP
- PTT
- QoS
- VoIPoW
- VoIP over Wireless
- Opportunities and Challenges

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Distributed Systems
VoIP over Wireless – Opportunities and Challenges
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?

flexibility of IP

spectral efficiency

future VoIPoW

circuit-switched voice

QoS
VoIP and VolPoW

- 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

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

must fit to each other
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

- 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

- 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

- iLBC codec: 108 Byte voice packet with every 60 ms

**Uplink**

- DSL receiver with 1024 kbps
- UMTS sender with 64 kbps

**Downlink**

- UMTS receiver with 384 kbps
- DSL sender with 128 kbps

- 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 160 bytes 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

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<th>MAX</th>
<th>PF</th>
<th>DEDF</th>
<th>CH-EDD</th>
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<td>&lt; 1 %</td>
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Conclusion

- **Situation today**
  - Circuit-switched voice is optimized for QoS and spectral efficiency
  - Little/no support for VoIP in cellular networks
  - VolPoW is VoIP over WLAN

- **Drivers for VolPoW 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