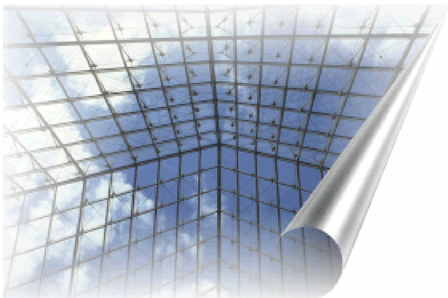
University of Würzburg
Informatik III (Distributed Systems)
Prof. Dr. P. Tran-Gia

Skype over UMTS

Tobias Hoßfeld

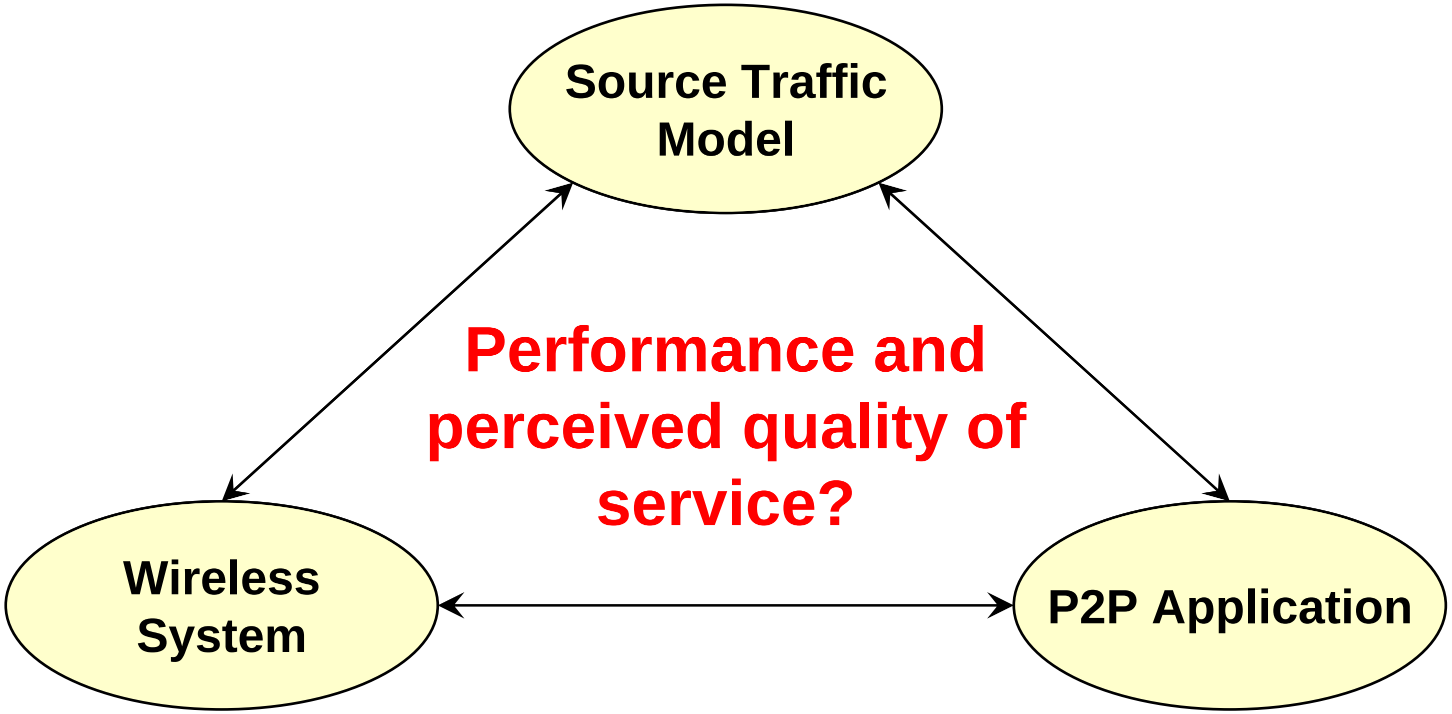
www3.informatik.uni-wuerzburg.de



Talk (25+5min)
ITG Fachgruppe 5.2.4
“VoIP over Wireless”
15th May 2006, Würzburg

P2P Applications Across Mobile Networks

When Do We Need Rate Control
for Dedicated Channels in UMTS?



Measurements in a Laboratory UMTS Network with time-varying Loads

Measurement and Analysis of Skype VoIP Traffic in 3G UMTS Systems

Measurement and Analysis of Skype VoIP Traffic in UMTS

Tobias Hoßfeld, Andreas Binzenhöfer, Kurt Tutschku
University of Würzburg, Institute of Computer Science,
Department of Distributed Systems.
Würzburg, Germany.
{hossfeld,binzenhoefer,tutschku}@informatik.uni-wuerzburg.de

Markus Fiedler
Blekinge Institute of Technology,
Department of Telecommunication Systems.
Karlskrona, Sweden.
markus.fiedler@bth.se

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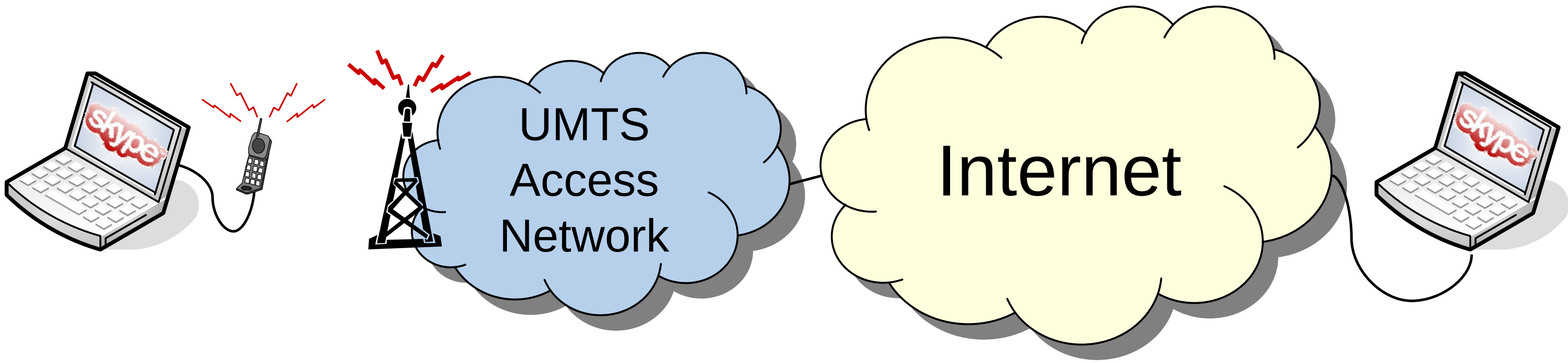
Lake Como, Italy, 2005

Skype over UMTS

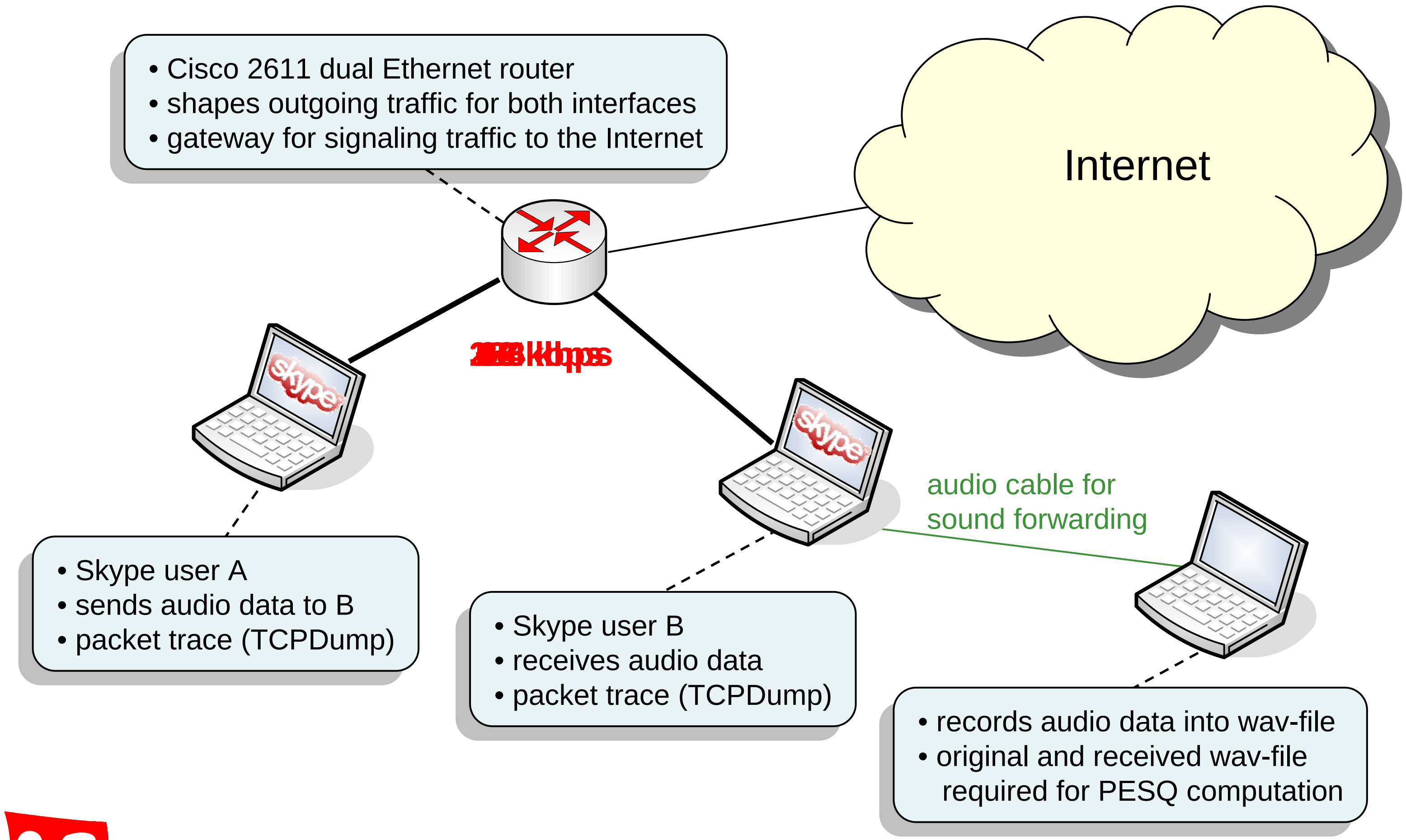
- **Mobile VoIP** as inexpensive alternative for voice calls
- Skype VoIP application is **very popular**
- 5.332.691 users online on Thursday, 15.05.2006, 11:27



- VoIP user **not interested** in **network performance** indicators
- **Perceived quality** of service is measured (PESQ values)
  - in a test-bed emulating rate-controlled dedicated channels
  - and dynamic changes of the (emulated) network
  - in a public German UMTS network

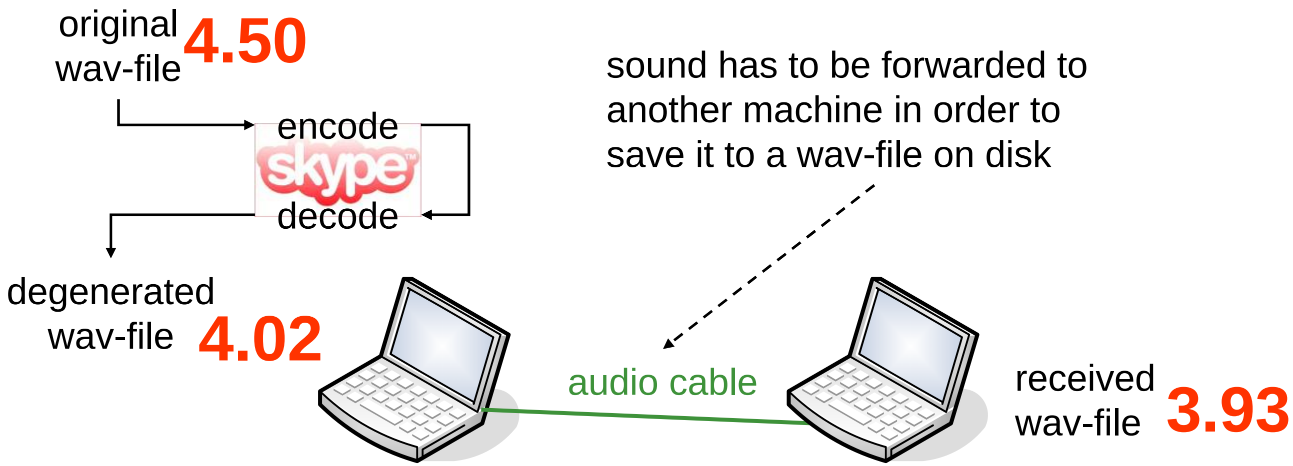
Bottleneck LAN: Emulated Rate-Controlled DCH in UMTS

- Bottleneck LAN scenario using a traffic shaping router
- We start a voice call with 16 kbps and increase up to 384 kbps



Reference PESQ Value w/o Influence of Network

- PESQ is not linear and very sensitive at its upper value
- 4.5 is best, -0.5 is worst quality
- Measurement setup **does not falsify** PESQ value measurements



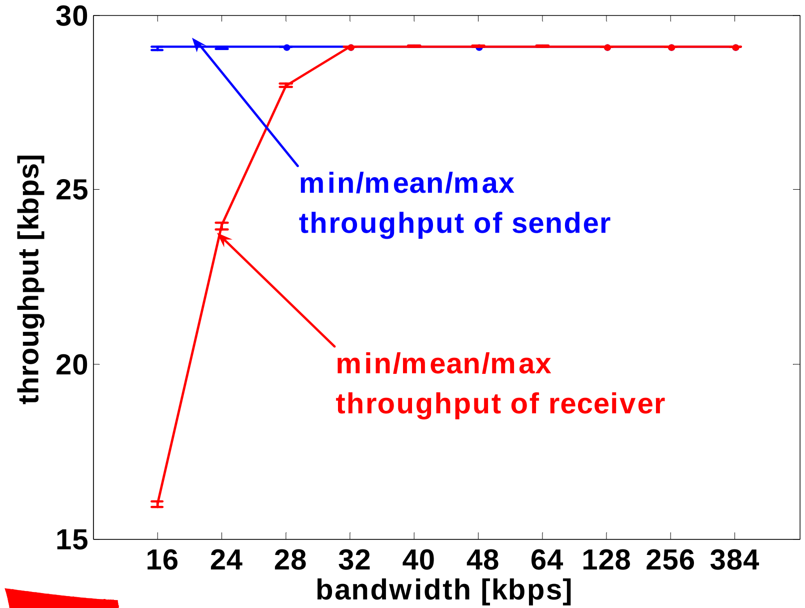
Remarks:

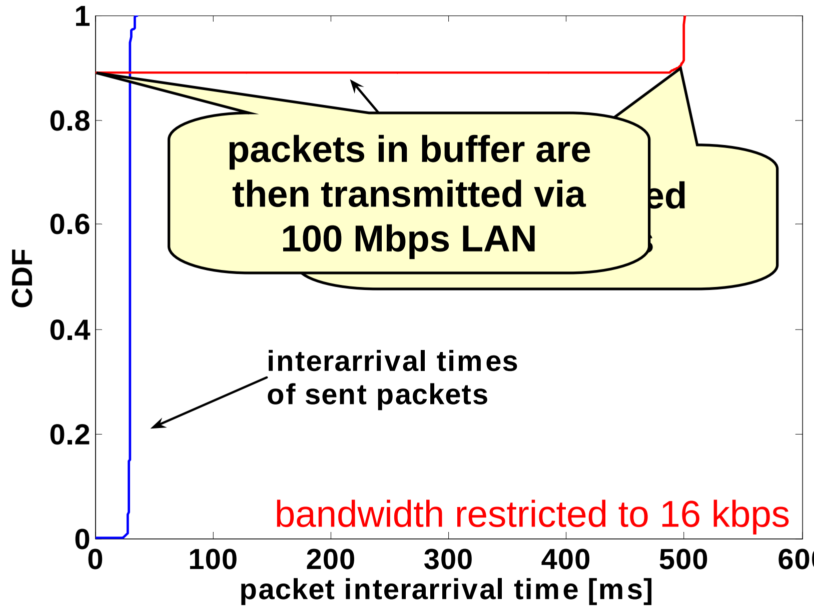
- **iLBC codec** (instead of ISAC) used due to **500 MHz CPU**

Bottleneck LAN: Throughput and Packet Interarrival Times

- Throughput includes 67 Bytes payload, 28 Bytes UDP/IP, 14 Bytes Ethernet header
- Total size of Skype packet 872 bit
- Packets are sent every 30 ms
- → Throughput of sender is about 29 kbps

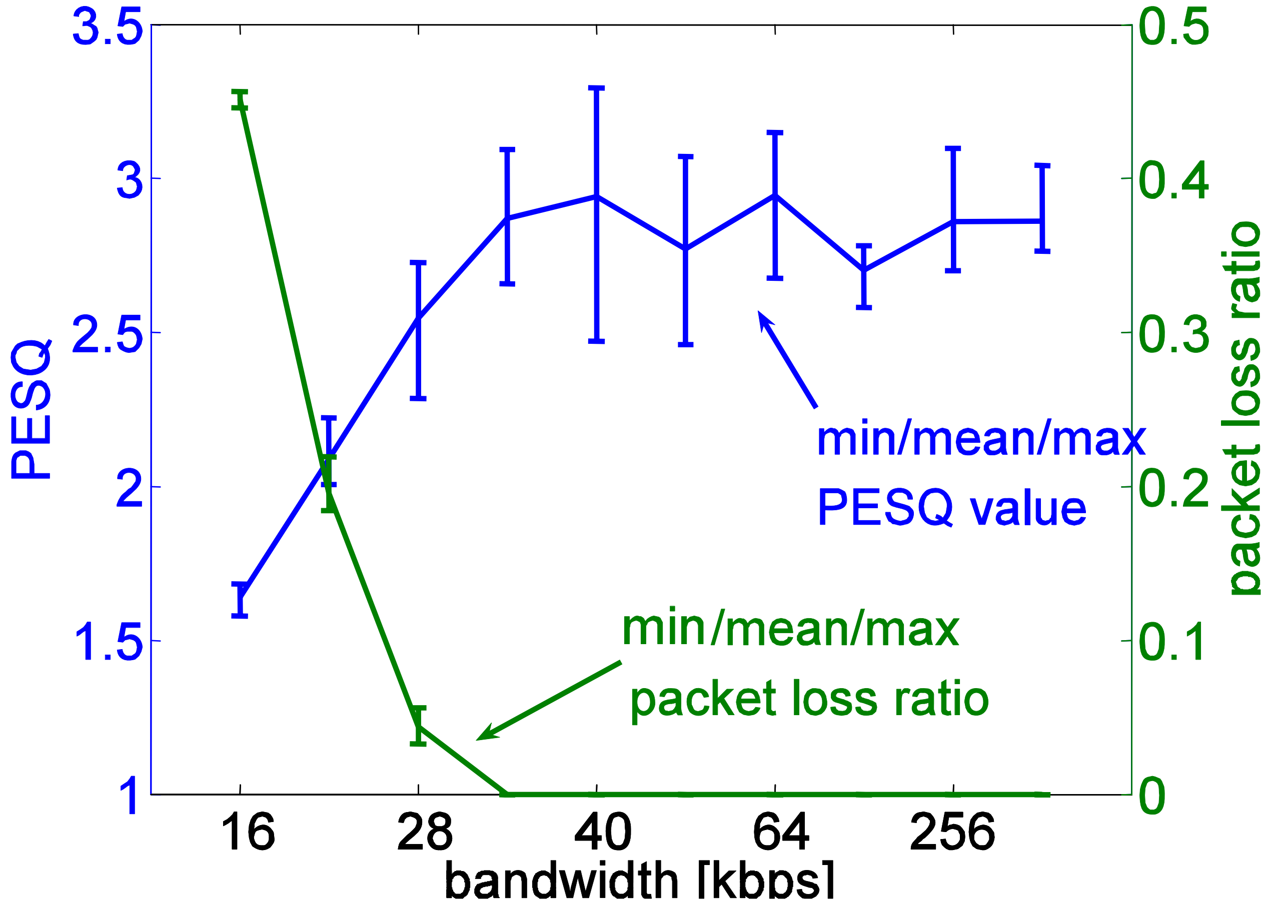
- Router has a buffer of 8 kbit
- 9 packets fit into buffer
- Bottleneck link of 16 kbps
- Buffer is emptied every 500 ms
- → 500 ms / 30 ms ~ 17 packets, i.e. 8 lost packets or 46% loss ratio



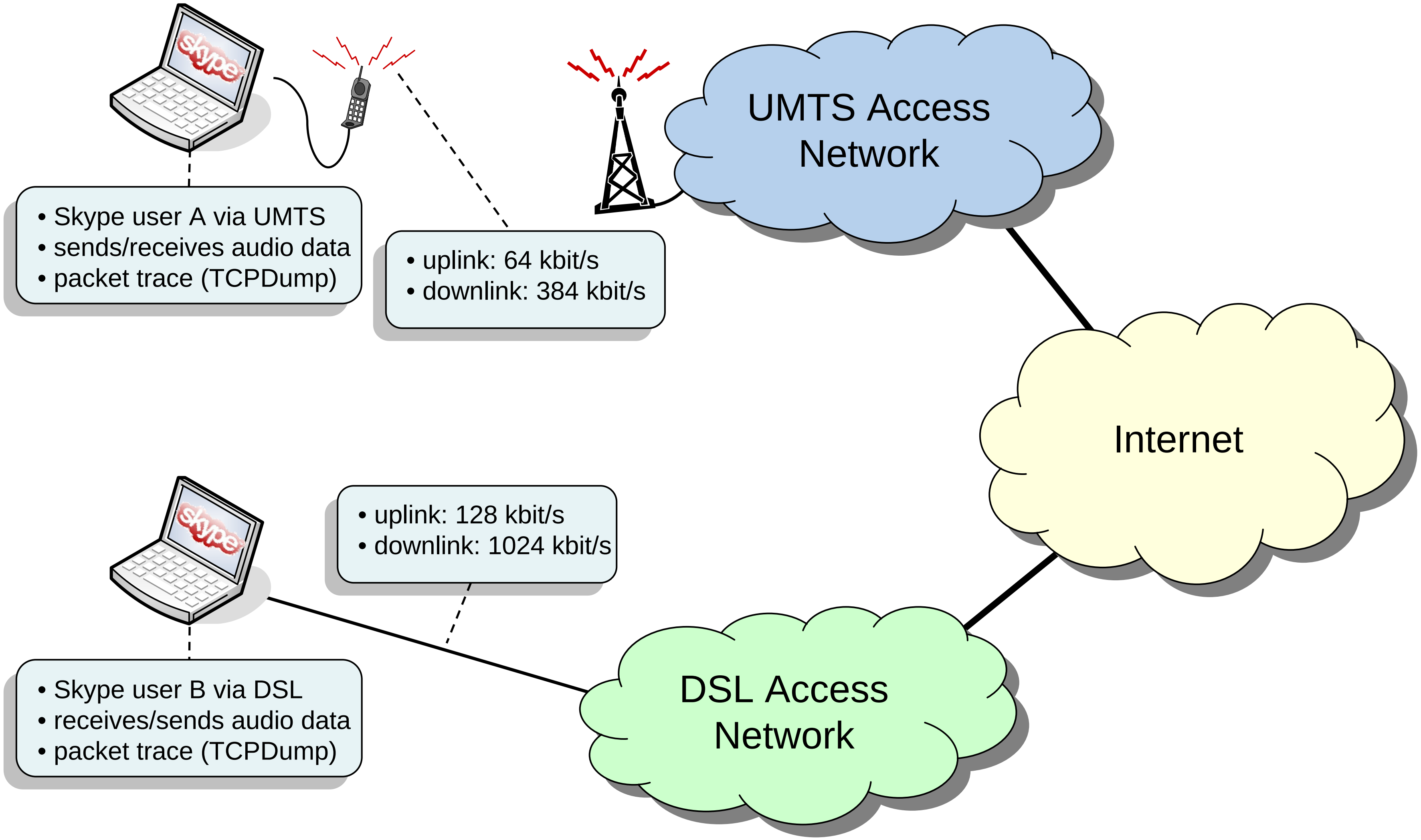


Bottleneck LAN: PESQ and Packet Loss Ratio

- Linear relationship between packet loss and PESQ for bottleneck links below 32 kbps

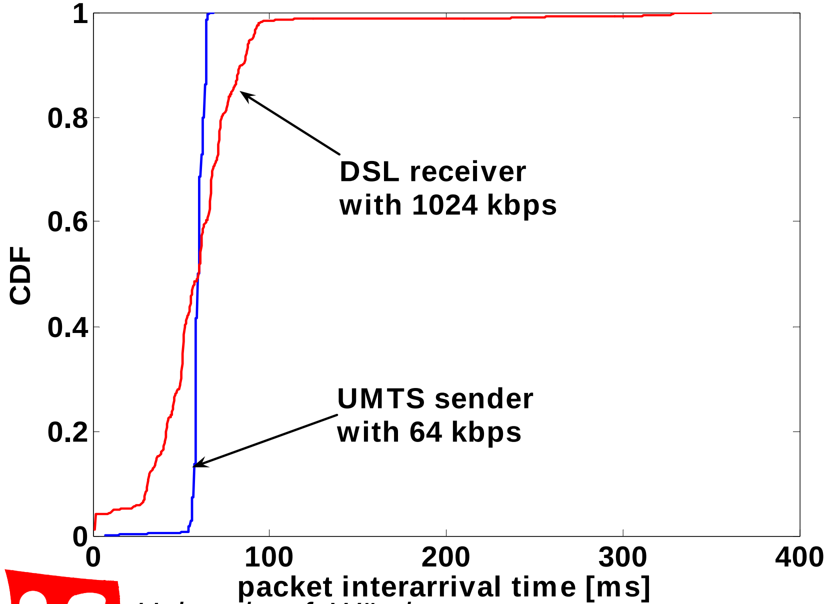


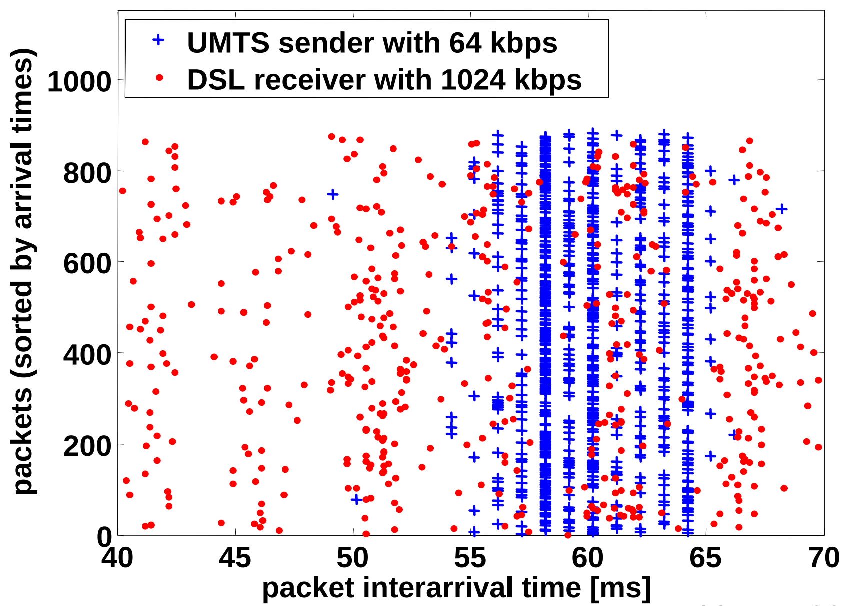
Measurement Setup in a Public UMTS Network



Packet Interarrival Times (PITs) for UMTS Uplink

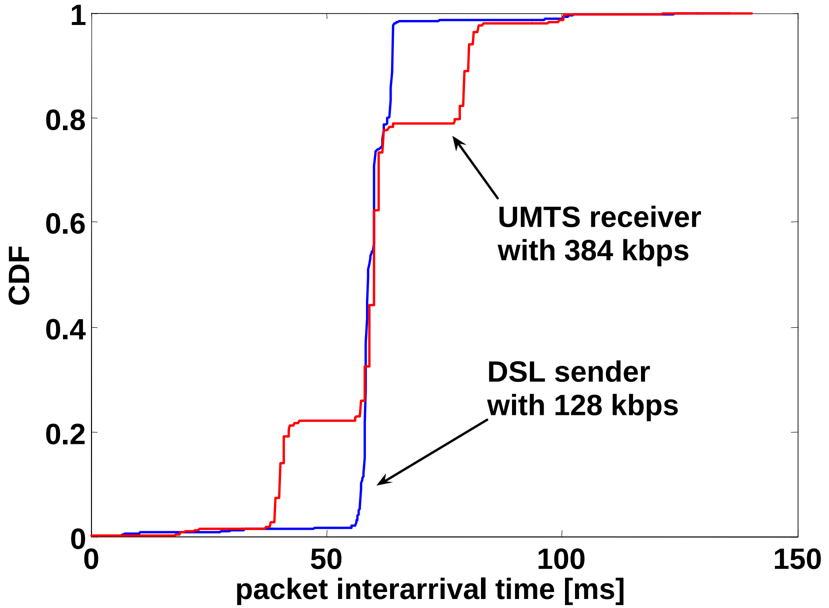
- UMTS client constantly sends a voice packet every 60 ms of 108 Byte payload
- → Another iLBC codec is used
- Due to jitter PITs at DSL receiver spread around mean
- PESQ 2.24 instead of 2.95 in bottleneck LAN with 64 kbps
- UMTS packets are sent at a discrete resolution of 1 ms
- Discretization happens at sender, probably by the PCMCIA UMTS card
- deviation of sent and received throughput differs about 1200bps

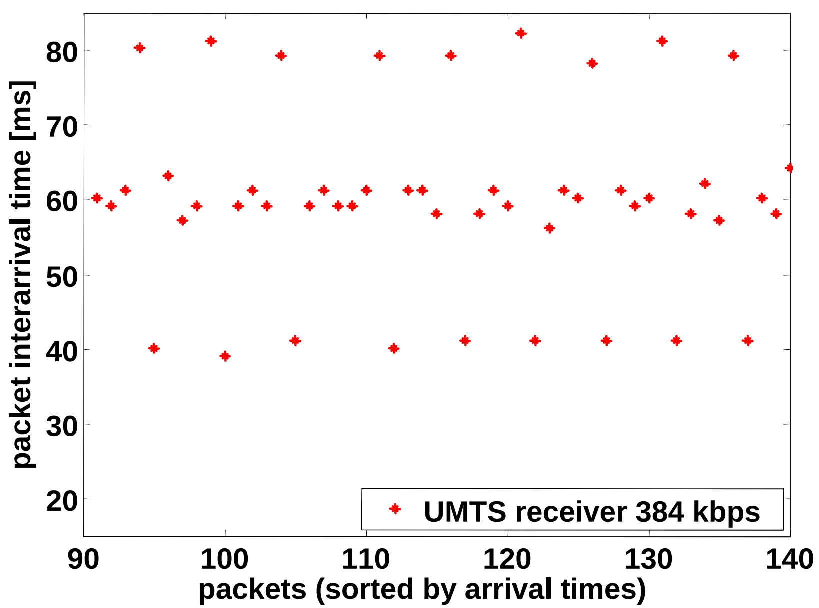




Packet Interarrival Times (PITs) for UMTS Downlink

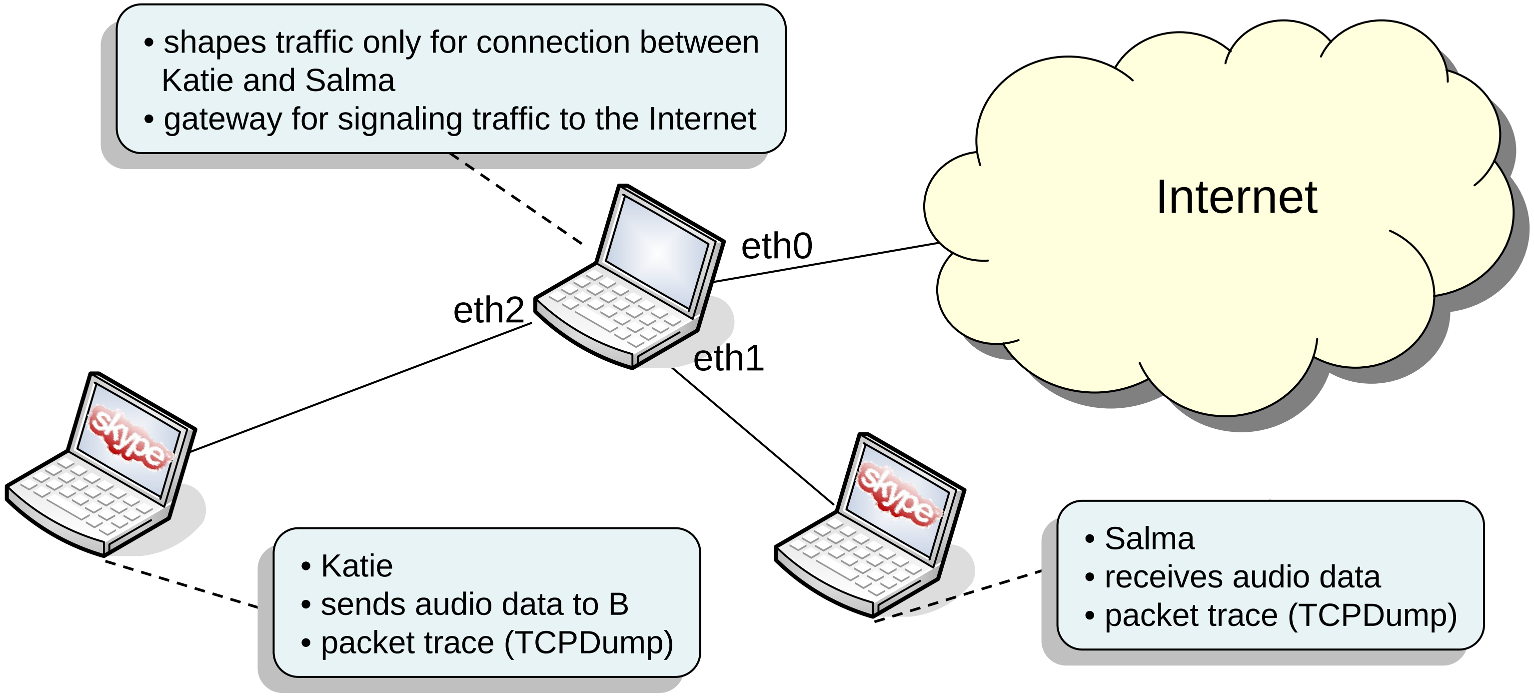
- Packets of 108 Byte are sent by DSL sender every 60 ms
- Interarrival times of packets arriving at UMTS receiver are discrete at 40, 60, and 80 ms
- UMTS transmission time interval (TTI) has a value of 10 ms
- Every 5th packet is retransmitted (FER adaptation of outer loop?)
- Deviation of sent and received throughput differs about 300 bps





Emulate Dynamic Changes in UMTS

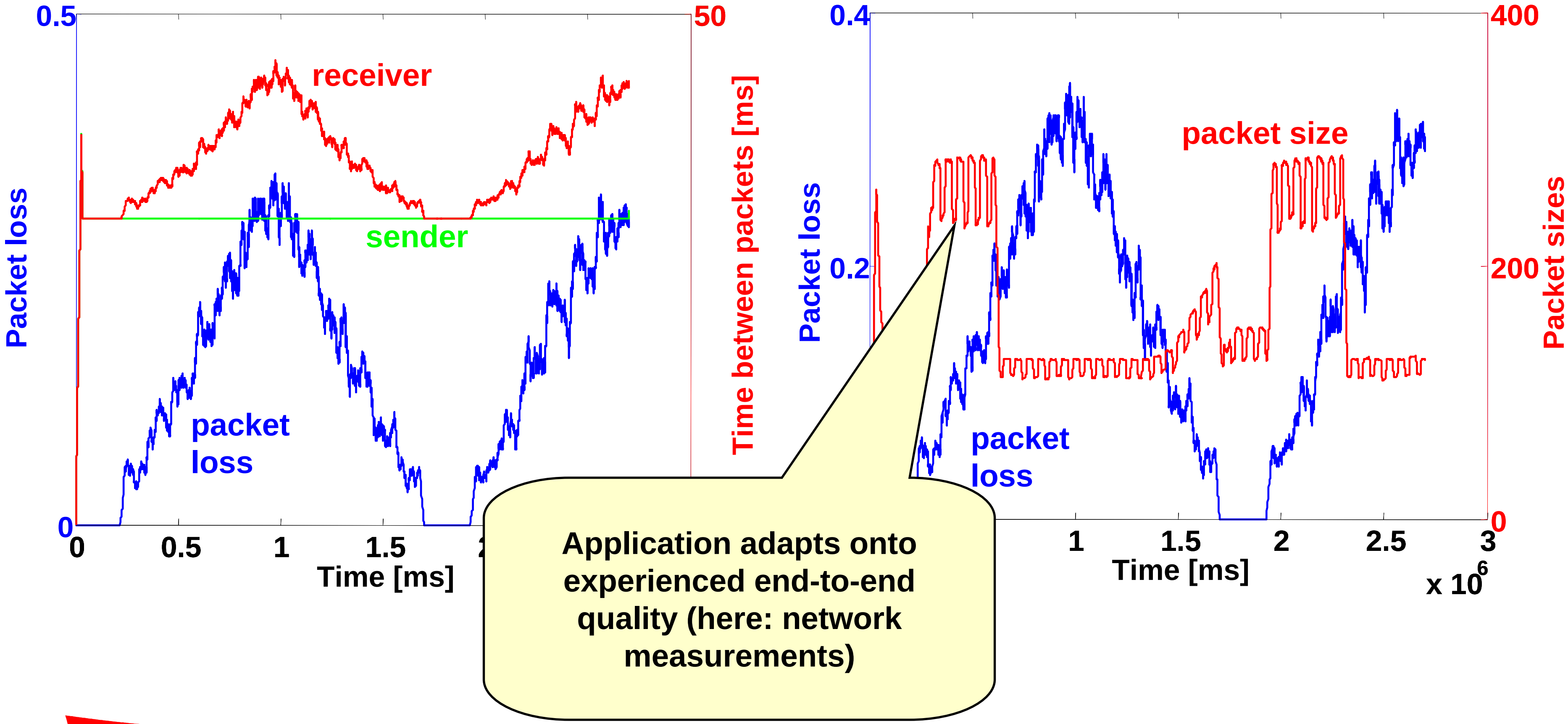
- Dynamic scenario using a software tool: NistNet
- CPU power of machines 1.2GHz ➡ **ISAC codec used**



- Audio file (51s) is repeated with a pause of 5s in between
- PESQ value computed for intervals of 56s
- Network characteristics evaluated using moving average (5min)

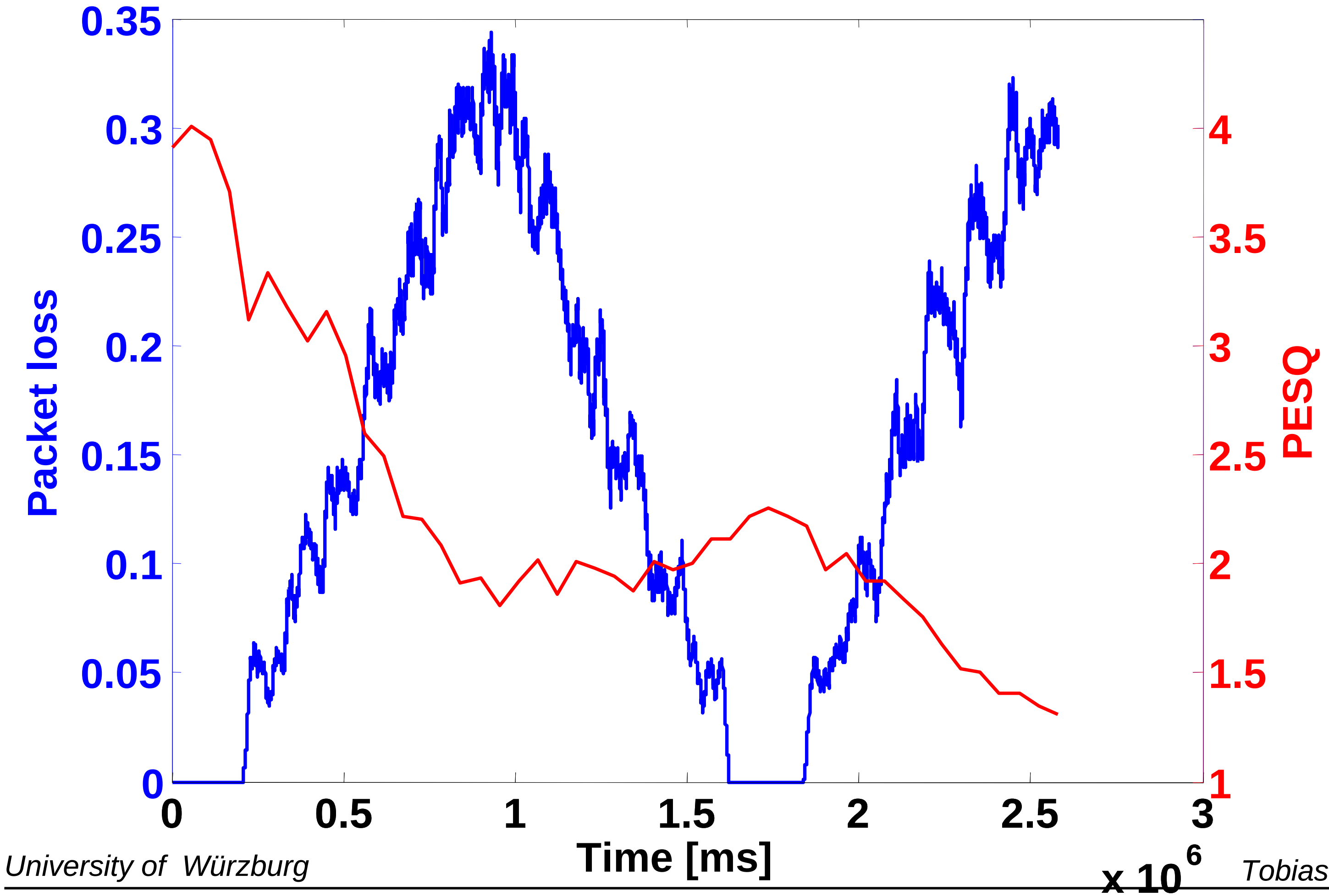
ISAC Codec Reacts on Dynamic Changes

- Packet sent times depend on codec, independent on packet loss
- Variable bit rate by increasing packet size, i.e. more audio data



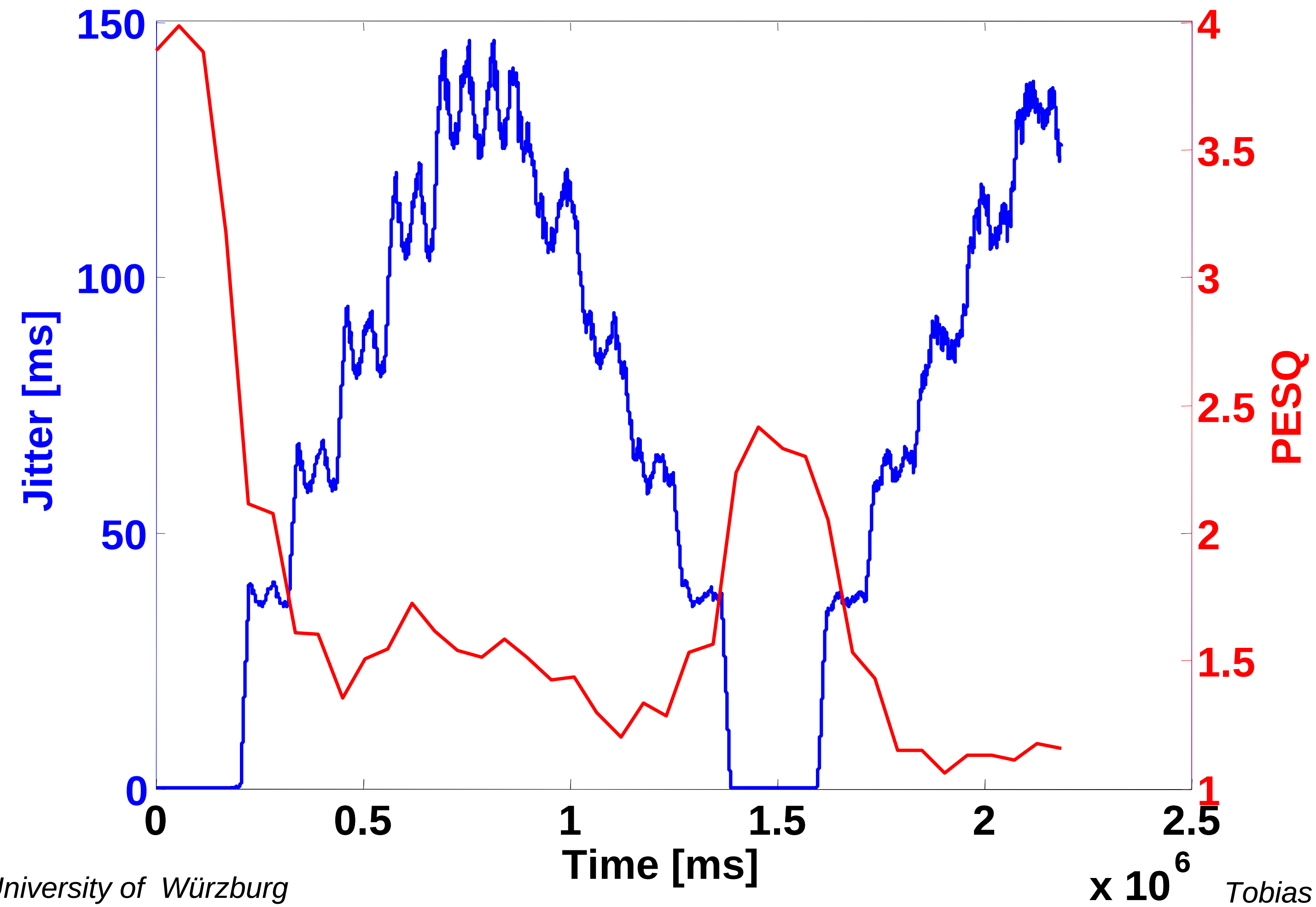
Dynamic Scenario: PESQ and Packet Loss

- PESQ computation: 51s traces within 50min (synchronization)



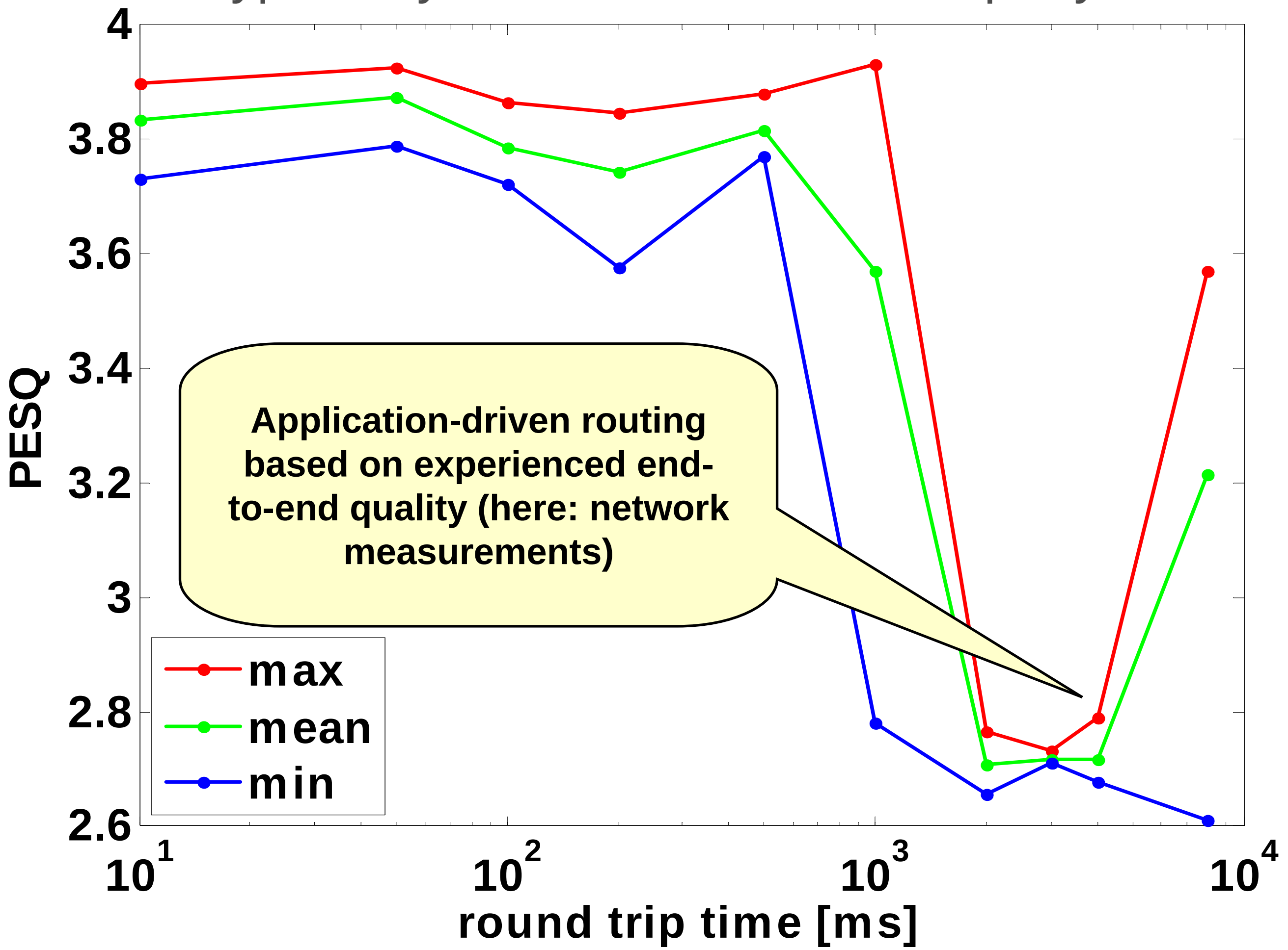
Dynamic Scenario: PESQ and Jitter

- Jitter is defined as standard deviation of end-to-end delay
- Mean end-to-end delay is set to 60 ms

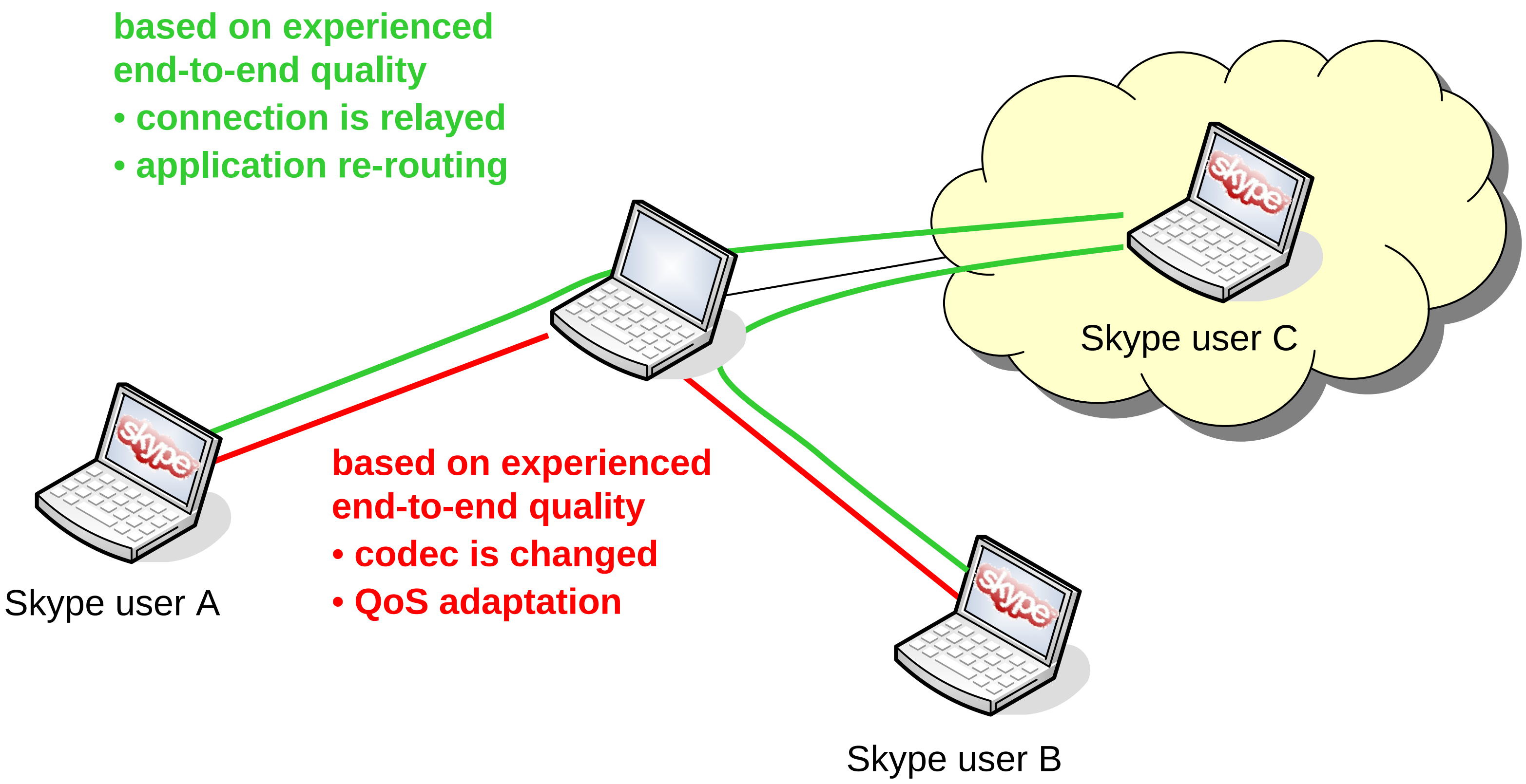


Dynamic Scenario: PESQ and RTT

- RTT>500ms results in strong PESQ degradation
- If RTT>4s Skype relays connection over third party machine



Application-driven QoS Adaptation and Routing



Conclusion and Outlook

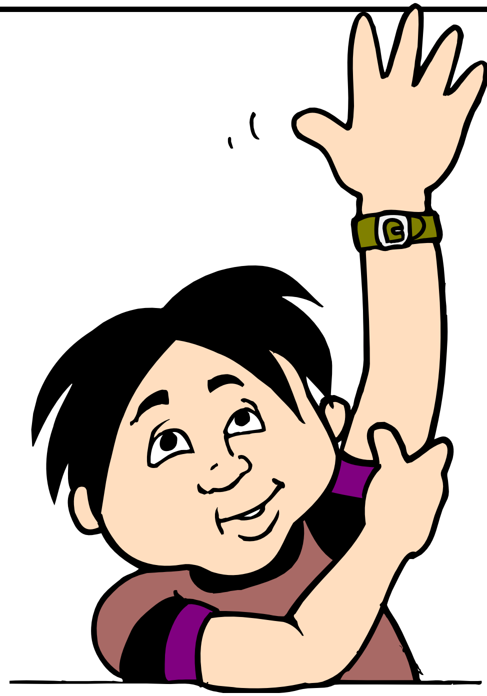
- Measurement of Skype considering the perceived voice quality
- UMTS sufficient to make **mobile VoIP** calls with Skype possible
- Variable bit rate ISAC codec used if CPU power above 600 MHz
- Connection relayed if ...
  - packet loss too high (>25%)
  - round trip time too high (>4s)
- Based on **experienced end-to-end quality** Skype implements
  - dynamic QoS adaptation onto environment
  - application-driven re-routing
  - **intelligence is moved to the edge**

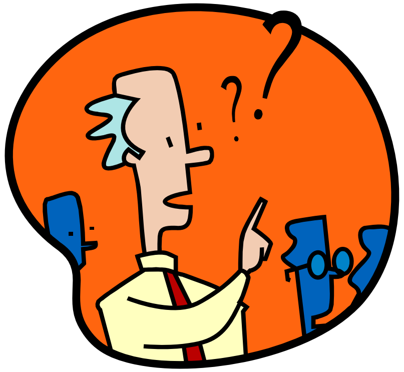
Outlook

- Impact of noise traffic / network congestion on the PESQ
- Characterization of Skype traffic patterns

Thank you!

- Any questions?





Answers…

- Melanie Brotzeller
- Michael Duelli
- Tommy Zinner
- Frank Bennewitz