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"Investigations on TCP traffic in mobile networks"

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Outlook

- Small Introduction on TCP
- TCP in Wireless Networks
    - Problems
    - Literature Proposed Solutions with Remarks
- TCP in Mobile Networks
    - Special Aspects
    - Network Scenario and Application Scenario
- Simulation Studies
    - Simulation scenarios
    - Simulation Results and Analytical Validation
    - Design Guidelines
- Further Studies

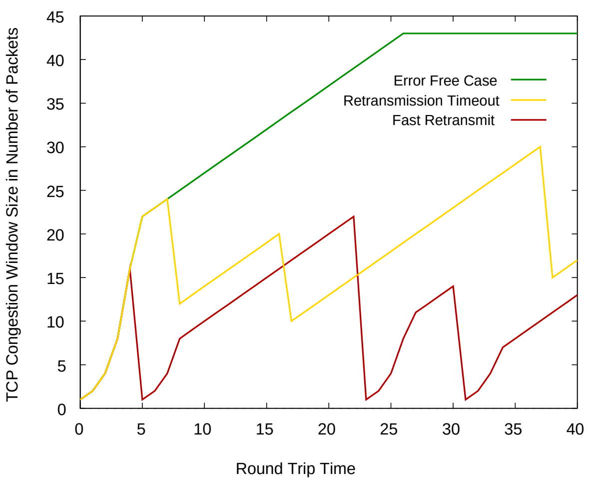
TCP Importance

- **TCP stands for Transmission Control Protocol**
- specified in RFC 793
- it is THE standard for reliable, best effort data transmission in today's Internet
- any internet compatible device supports the TCP/IP suite
- 70% − 80% of the internet traffic today is based on TCP
- applications using TCP
  - http (Web browsing)
  - smtp (mail transfer)
  - ftp (file transfer)

TCP Functionality

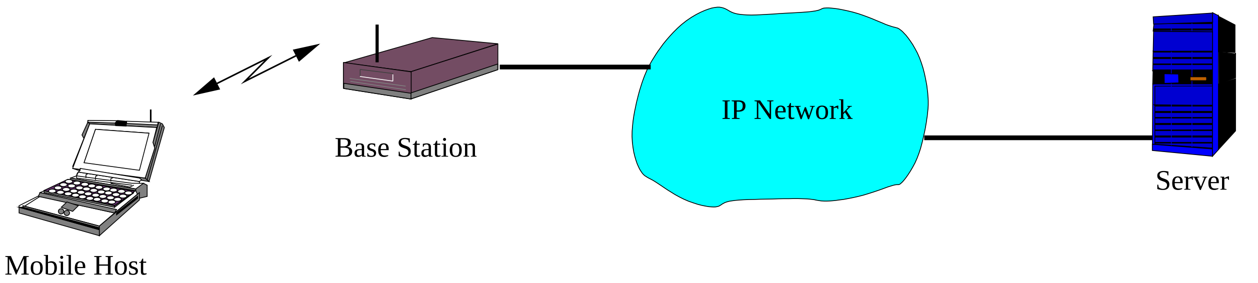
- reliable end−to−end transport
  - based on a mix of "go back n" and "selective repeat" retransmission strategy
  - only dealing with positive acknowledgments
  - timer based packet loss detection (Retransmission Timeout)
  - and acknowledgment indicated loss detection (Fast Retransmit)

- end system based traffic control functions
  - to avoid network congestion and end system overload
  - sending an adaptive data rate using a window mechanism
  - based on the returning acknowledgments
  - **assumes network congestion to be the primary cause for packet losses !**
  - reacts on packet loss with rapid slow down of the sending data rate



TCP in Networks with Wireless Links

- much higher bit error rate on the wireless link than on the wireline links
- sporadic additional packet delays due to handovers and layer 2 disconnections

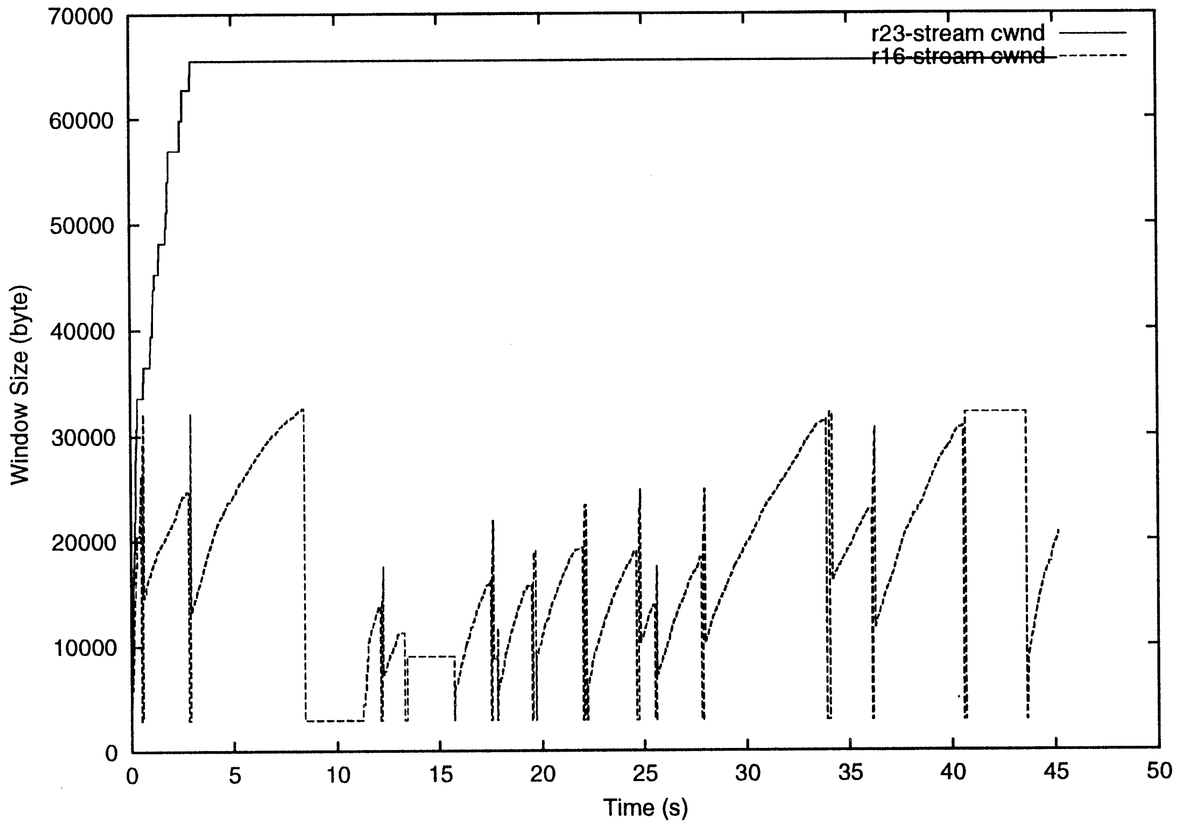


- ➢ non−congestion packet losses
- ➢ very high packet delay variations

Effects

- The TCP congestion control always assumes network congestion in case of packet errors/losses. TCP reacts with unnecessary decrease of the offered (sending) data rate.
- The TCP congestion control mechanism assumes network congestion in case of spontaneous additional delays (like handover delays). TCP reacts with unnecessary decrease of the offered data rate and spurious retransmissions.
- TCP increases the offered data rate slowly. Meanwhile, the potentially bandwidth can not fully be used.
- TCP uses an exponential backoff strategy for the TCP retransmission timeout timer. In case of layer 2 disconnection or longer handover delays this behavior can lead in transmission lacks.
- Bursty traffic nature because of the TCP window mechanism.

TCP Behaviour



Dynamics of the TCP window size [M. Mathes et al. “Verkehrsmessungen in Mobile −IP Netzen”]

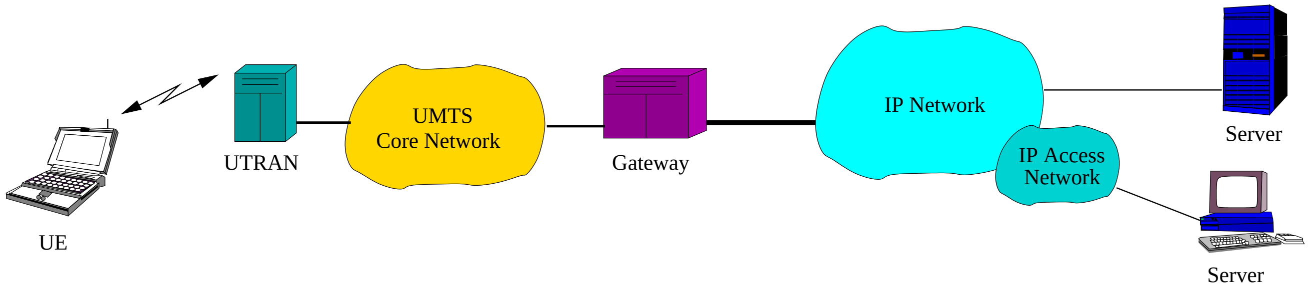
Literature Approaches

- **Radio Link Layer schemes**
  - Foreward Error Correction FEC
  - appropriate link layer ARQ schemes
  - TCP aware Link Layer Protocols
  - Snoop protocol
- **TCP add ons**
  - TCP Selective Acknowledgments
  - Explicit Loss Notification (ELN)
  - TCP Eifel Algorithm
  - inter−protocol communication [Fikouras et al.]
- **TCP split connection approaches**
  - indirect TCP, M−TCP, W−TCP
  - Remote Socket Architecture
  - WAP concept

Approach Remarks

- most of the approaches focus on the packet loss problem in a LAN environment
- only a few deal with the packet delay variation problem (Eifel algorithm and [Fikarous et. Al])
- TCP aware link layer approaches (e.g. Snoop) do not work with IP Sec or IP tunneling
- TCP ad on solutions are not practicable, since they can only be realized with software updates on servers and mobile end systems
- split connection schemes break the end−to−end semantic of TCP

Mobile Network Scenario

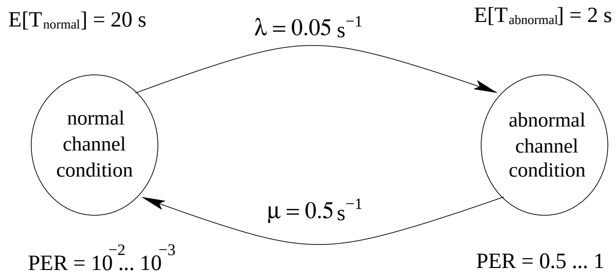


- the mobile wireless link already includes a strong link layer protection (FEC + ARQ)
- therefore, a bad quality of the wireless channel primary result in additional packet delay due to local ARQ retransmissions or link layer disconnections
- in addition with spontaneous delays due to handover

➢ **TCP has to deal with a significant packet delay variation**

Mobile Network Environment

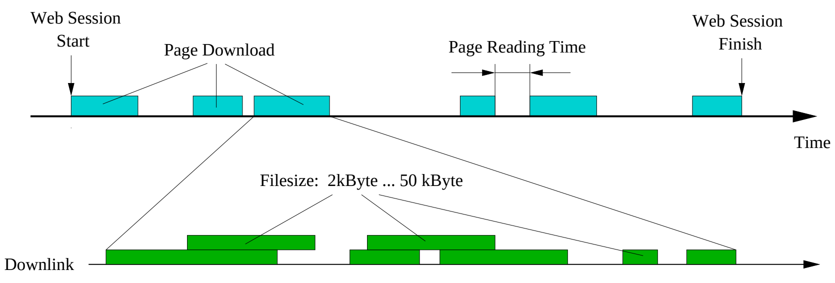
- Wireless Link
  - wireless link is assumed to be the bottleneck of the network
  - **shared capacity is less than 170 kbit/s (GPRS) or 384kbit/s / 2 Mbit/s (UMTS)**
  - downlink is the limiting resource, due to asymmetric nature of web traffic
- Channel
  - **ARQ + FEC schemes lead to a much smaller packet error rate (PER) than commonly assumed for other wireless systems**
  - UMTS supports the setting of a target IP loss rate (1%) realized by appropriate settings for UMTS power control and error correction schemes

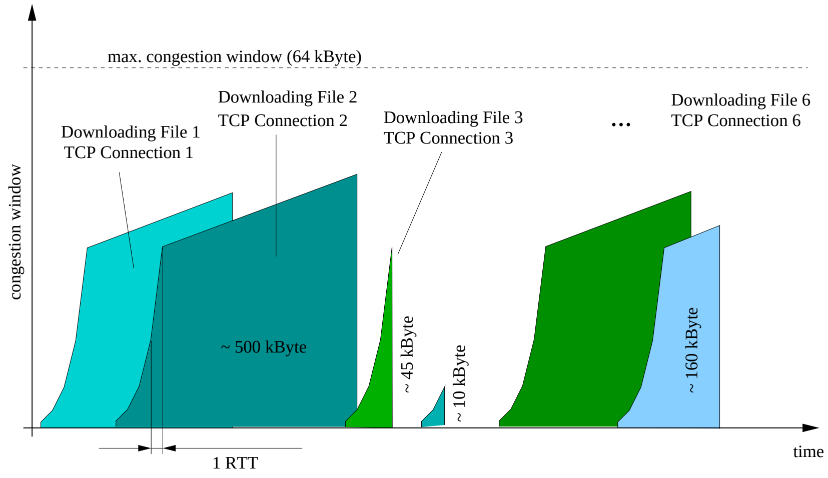


- Users
  - only a few users will be active (running one/more TCP connections) within one cell
  - **aggregated flow behavior can not be considered**

Application Protocols

- major application: web browsing (http)
- web pages consist of several small objects/files (2...50 kByte)
- TCP sends only a few packets (1 ... 30) before finishing the connection and the application protocol establishes the next TCP connection.
- **bulk data transfer at the application layer can not be assumed**
- depending on the http version there can be parallel TCP connections with or without persistent connections per page download [T. Inrich]



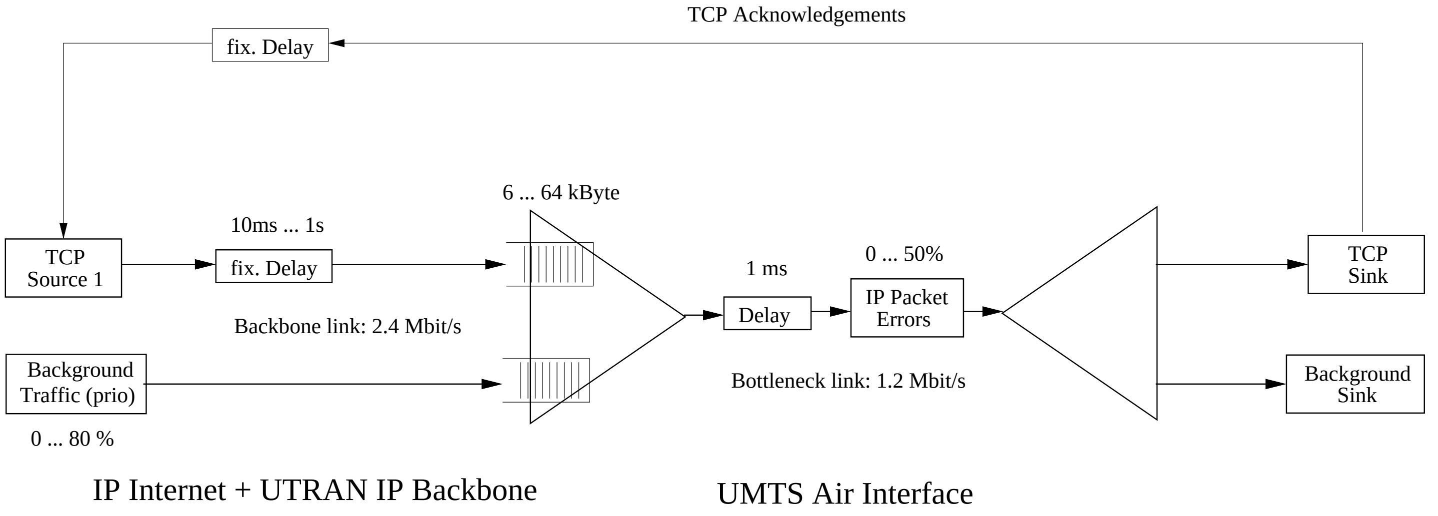


Simulation Studies

- Mobile Network Environment
  - Influence of the buffer size at the interconnection between wireline and wireless links
  - Influence of the packet error rate on the wireless channel
  - Influence of packet delay variations due to link quality variations and handover
- Application Protocol (HTTP)
  - http traffic characteristic
  - mean number of active users within the radio cell

Simulation Model for MNE

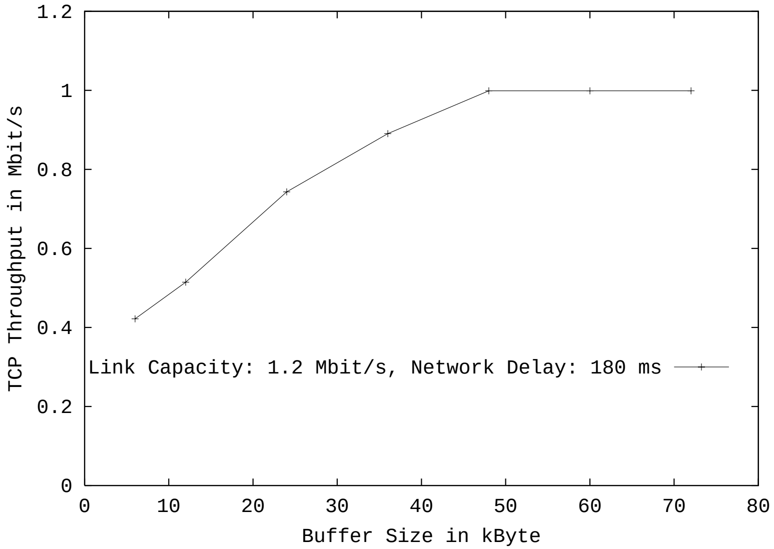
- YATS simulation tool
- investigation on a single TCP flow
- bulk data transmission assumed (ftp download)
- background On/Off source with neg. exp. phase duration

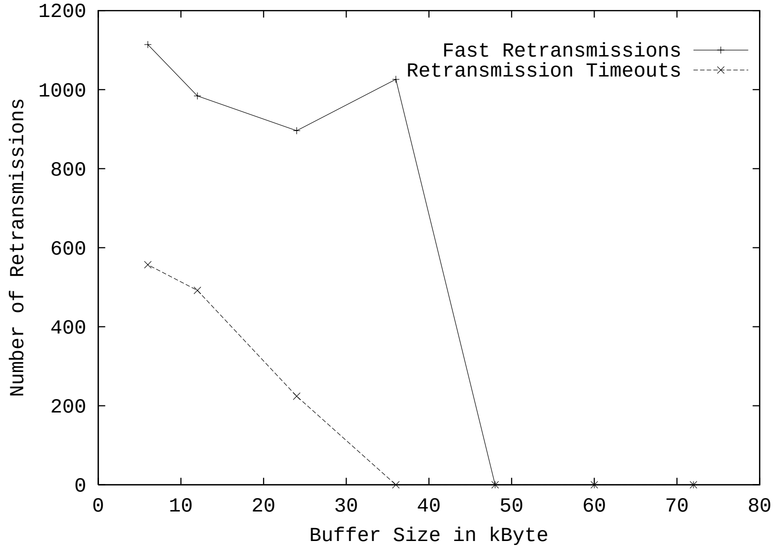


Parameters

Network Scenario	
Wireless Link Capacity	1.2 Mbit/s
Backbone Link Capacity	2.4 Mbit/s
Network delay	180 ms; (vari)
Buffer Size	70 kByte; (vari)
Packet Loss Probability	0.0 %; (vari)
Background On/Off Source	
Mean On phase duration	0s; (vari)
TCP Parameters	
Sender buffer size	65535 Byte
Receiver buffer size	65535 Byte
Maximum congestion window	65535 Byte
Maximum Segment Size (MSS)	1460 Byte
Maximum Transmission Unit (MTU)	1500 Byte
Fast Retransmit	on
Nagle Algorithm	off
Clock Tick	500 ms
Minimum Retransmission Timeout	500 ms
Initial Retransmission Timeout	3 s
Initial sstresh value	32768 Byte
Simulation	
Realtime Simulation	~ 1h

Influence of Buffer Size

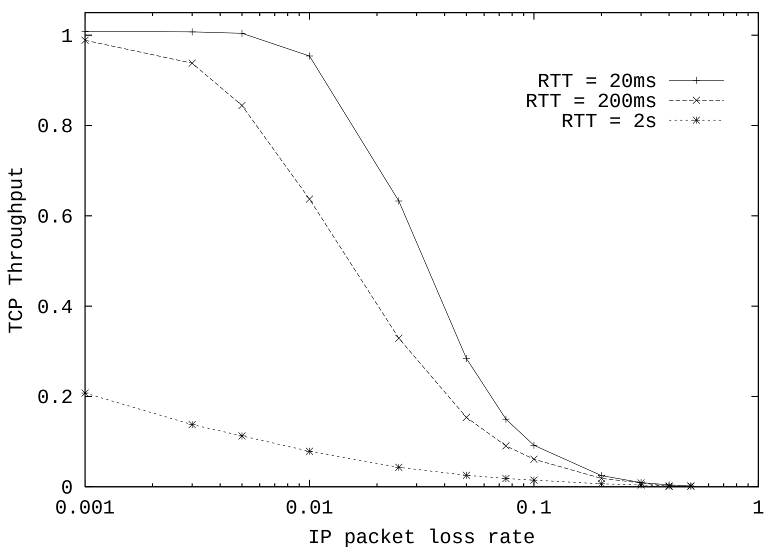


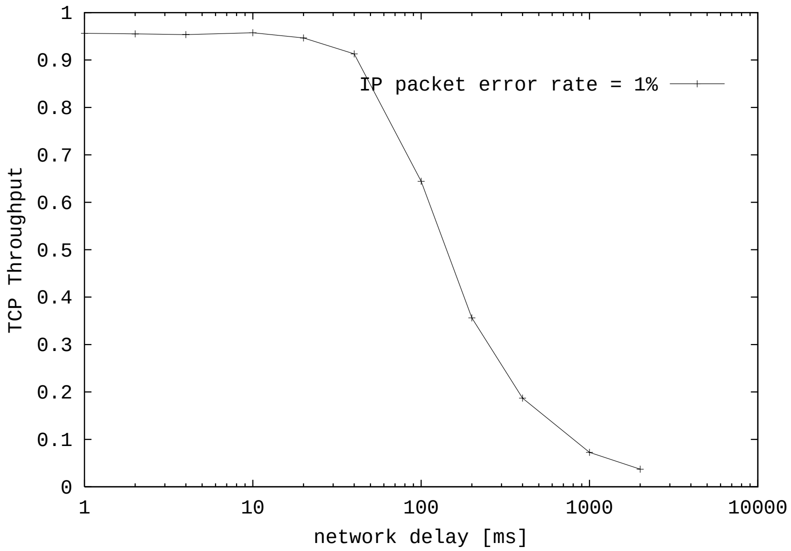


TCP goodput and number of retransmissions over buffer size

$$\textit{Current Flow Queue Lenght} = \textit{current Congestion Window} - \frac{\textit{Network Delay} \cdot \textit{Bottleneck Link Capacity}}{\textit{Number of active TCP sources}}$$

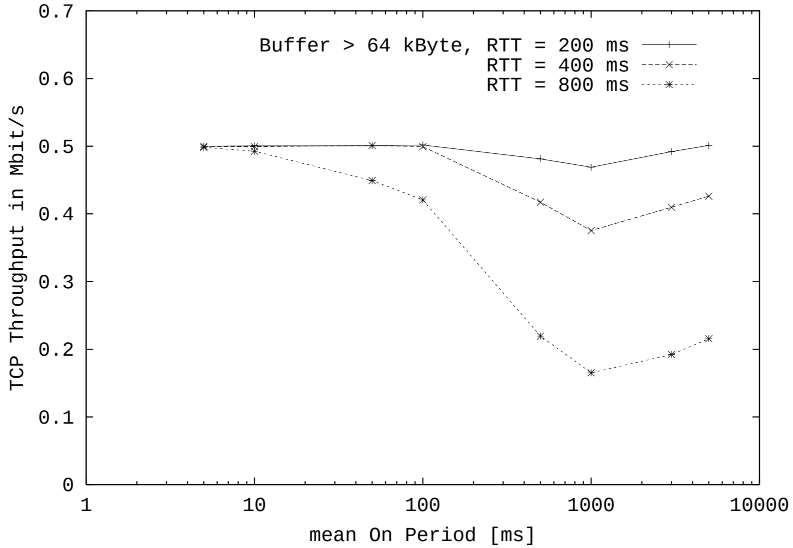
Influence of Bit Errors

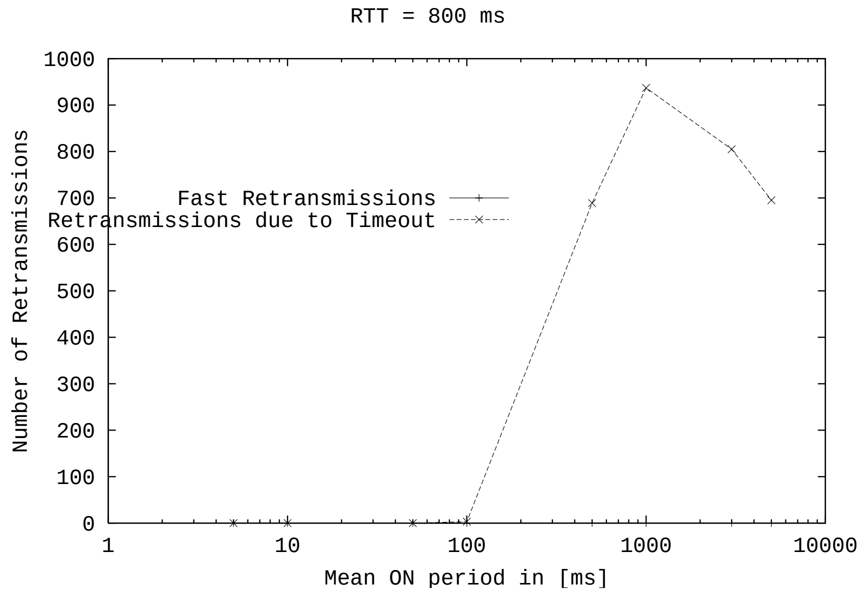




Influence of packet error rate for different network delays (RTT)

Link Capacity Variations





Influence of varying link capacity for different network delays (RTT)

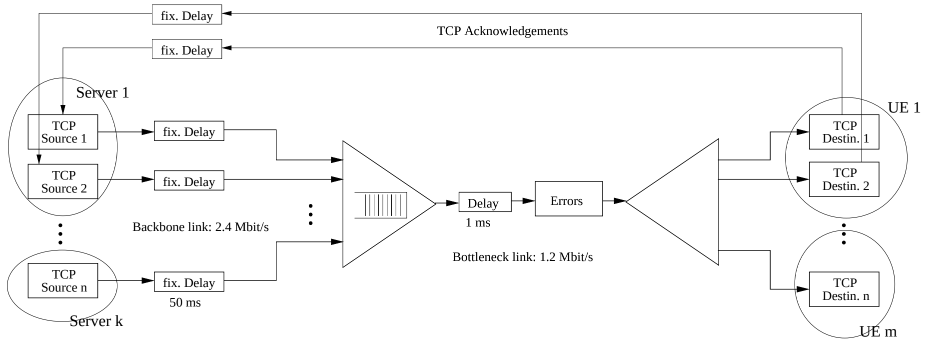
$$RTO_i = A_i + 4 \cdot D_i \qquad A_i = A_{i-1} + \frac{1}{8} \cdot (RTT_i - A_{i-1}) \qquad D_i = D_{i-1} + \frac{1}{4} \cdot (|RTT_i - A_{i-1}| - D_{i-1})$$

Design Guidelines

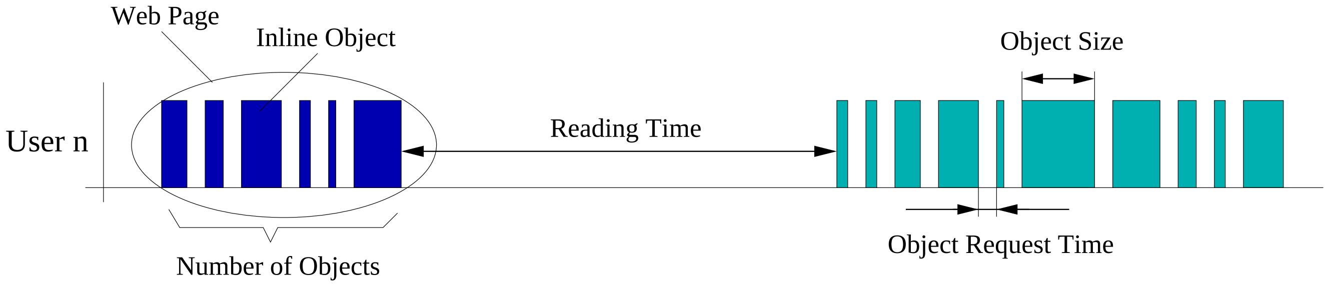
- avoid buffer overflows by providing **sufficient buffer space** at the interconnection between high bit rate links and low bit rate links (Internet <−> UTRAN)
- the additional **network delay** caused by the UMTS network should be as **small** as possible
- a **constant available bandwidth** guarantees best TCP behavior
- avoid phases there TCP does **not get any link capacity** of more than **100ms**
- try to keep **handover delays** and **layer 2 connection setup** as **fast** as possible
- target on an **IP packet loss rate** smaller than **0.2%**

Simulation Model for HTTP Invest.

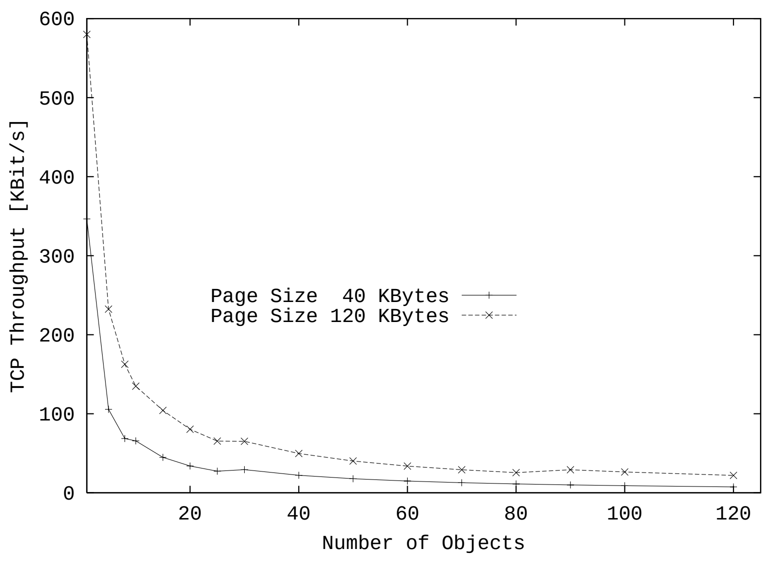
- YATS simulation tool
- focus on downlink traffic
- modeling HTTP 1.0 without parallel connections nor keep alive

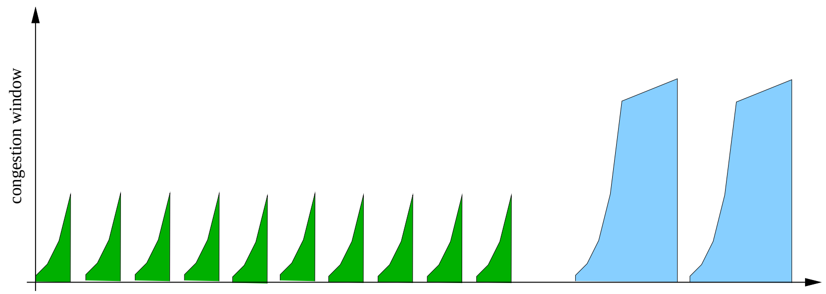


Parameter	Distribution	Mean Value
Reading Time	Neg. exponential	20 s
Number of Inline Objects	Equal distributed	20
Object Size	Equal distributed	4 kByte / (vari)
Object Request Time	Fixed	100 ms



Offered Load incl. TCP Influence





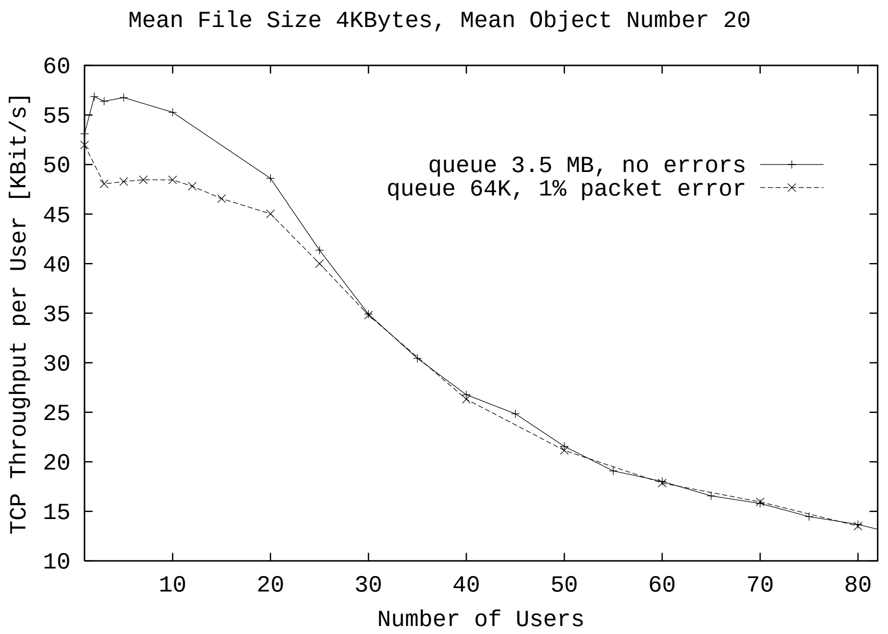
Offered Page Download Bitrate of the HTTP 1.0 model including the influence of TCP congestion control

$$Offered\ Page\ Download\ Bitrate = \frac{ObjectSize}{Cycles \cdot Round\ Trip\ Time + Object\ Request\ Time}$$

$$Round\ Trip\ Time = Network\ Delay + \frac{Bottleneck\ Link\ Capacity}{PacketSize\,(MTU)} + Buffer\ Delay$$

$$Cycles\,[TCP\ Slow\ Start\ Phase] = \left\lceil \log_2\left(\left\lceil \frac{ObjectSize}{MSS} \right\rceil + 1\right) \right\rceil$$

Page Download Throughput



Mean over all reached throughputs while downloading a Web Page in relation to the number of active users within the cell

$$Offered\,Load < 95\,kbit/s$$

$$Fair\ Share = \frac{Bottleneck\ Link\ Capacity}{Number\ of\ Users}$$

Further Studies

- Simulation Enhancements
  - Mobile Network Environment
    - Gilbert Channel Model for IP packet losses
    - Influence of different link layer ARQ schemes
    - Detailed investigations on handover scenarios (hard, soft, intra system, inter system, mobile IP) according to the resulting delay and packet losses
  - Application Protocols
    - HTTP with parallel TCP connections and keep alive
    - Detailed view on single requests, Fairness
    - Combination with Mobile Network Environment assumptions