1. Introduction to QoS and QoE and service performance

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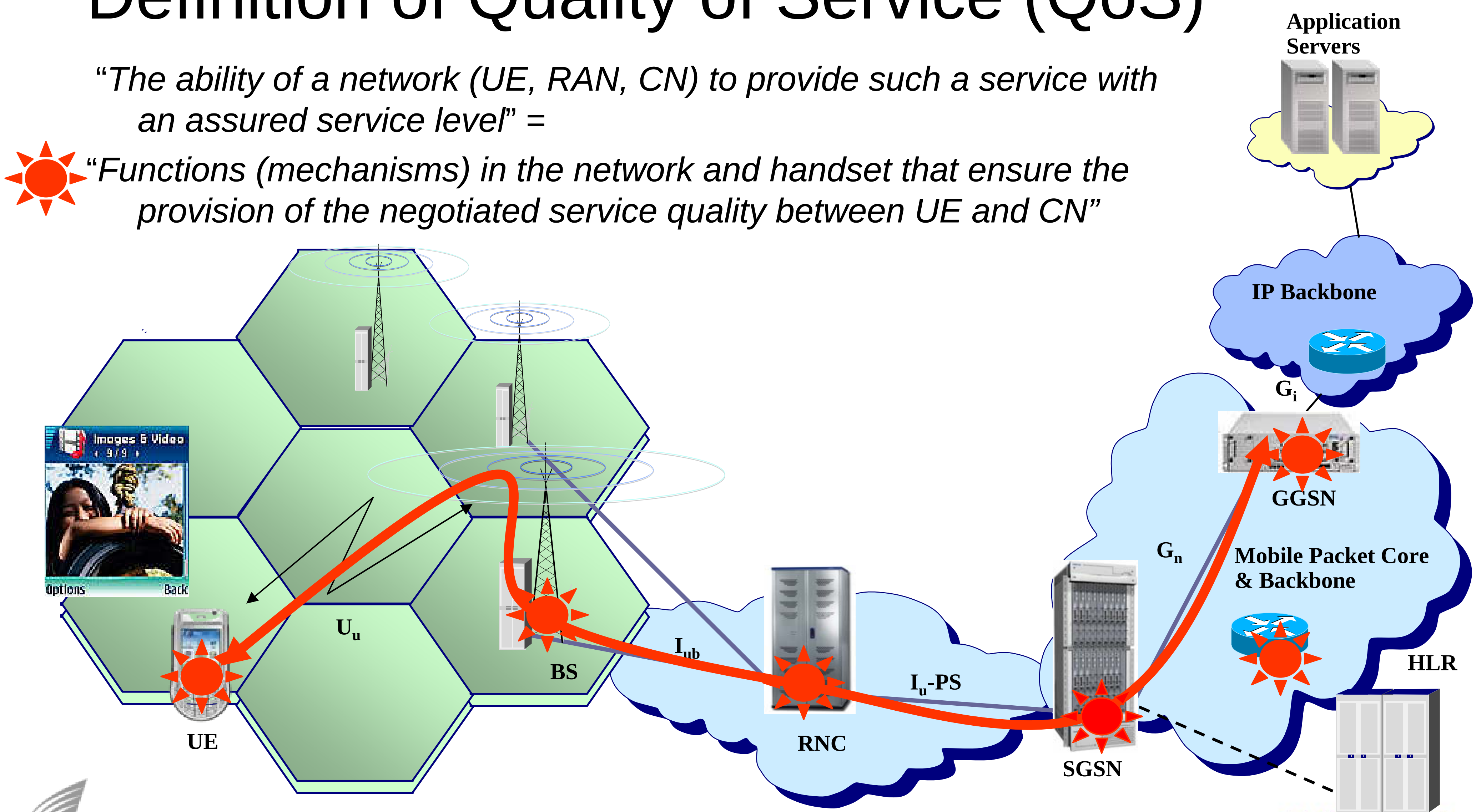
Contents

Definition of Quality of Service (QoS)

"The ability of a network (UE, RAN, CN) to provide such a service with an assured service level" =

"Functions (mechanisms) in the network and handset that ensure the provision of the negotiated service quality between UE and CN"



Definition of Quality of Experience (QoE)

"*What the user really perceives, i.e. how satisfied he or she is with the service, in terms of usability, accessibility, retainability and integrity of the service*"

"*QoE reflects the collective effect of service performances that determines the degree of satisfaction of the end user*"

Ultimate goals

- The aim of the network and services should be to achieve the maximum user rating (QoE)
- Network quality (QoS) is the main building block for reaching that goal effectively

Factors (aspects) affecting QoE

- This course will only deal with the technical aspects of QoE in detail

Technical factors (mainly QoS):
- E2E network quality
- Network/service coverage
- Handset functionality

Non-technical (subjective) factors:
- Ease of service set-up
- Service content
- Pricing
- Customer support

↓ ↓

Quality of end-user Experience (QoE)

QoE value chain

- QoE depends on how well the operator orchestrates the entire value chain as seen by the user

End user device and application software

Enable the end user to experience the content

Service and network providers

Provide <u>infrastructure</u> and enable the content transport from Content Providers to the mobile users

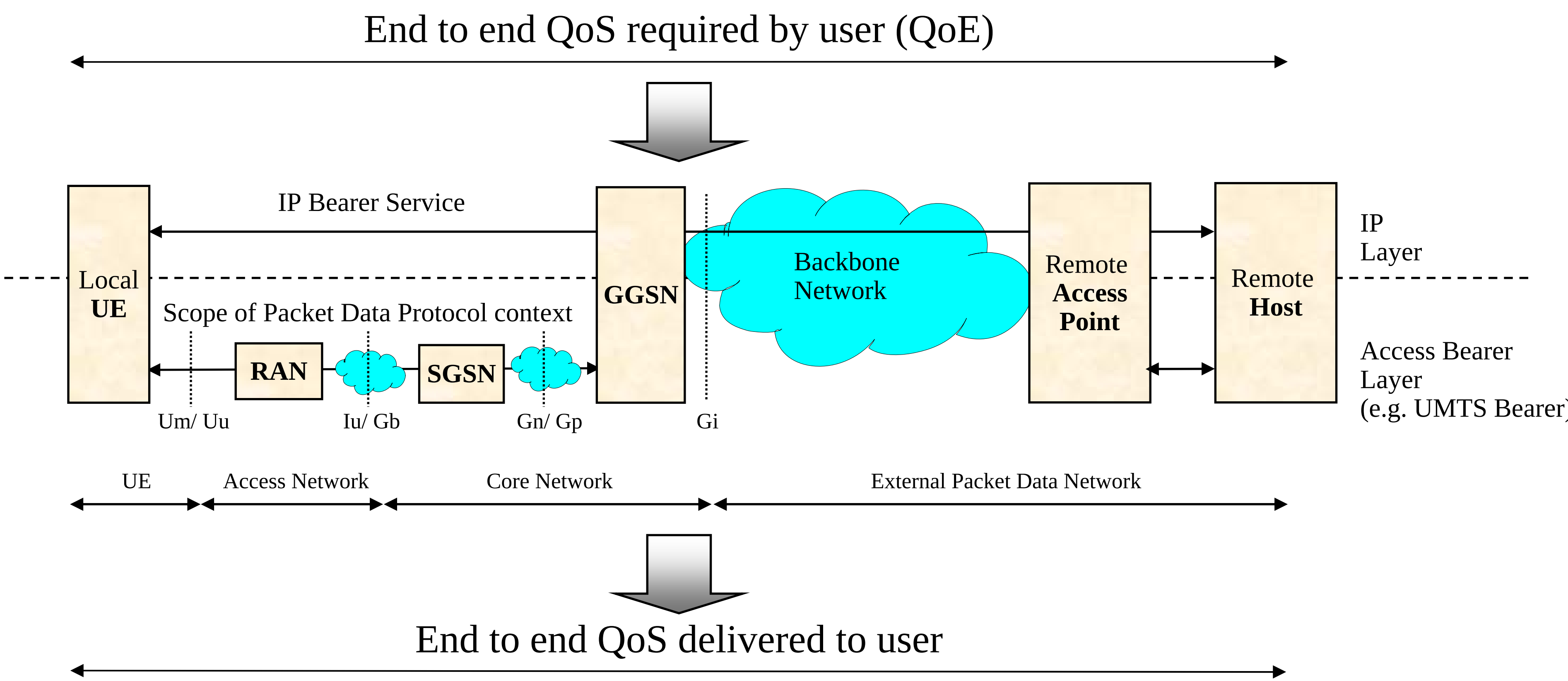
Mobile Content providers and contents

Mobile Content Providers and originators, websites, WAP sites, Games, Video, Audio, portals etc…

Network Vendors and system integrators

Although not seen by the end user, they enable the above three entities of the value chain

Top-down approach / end-to-end definition

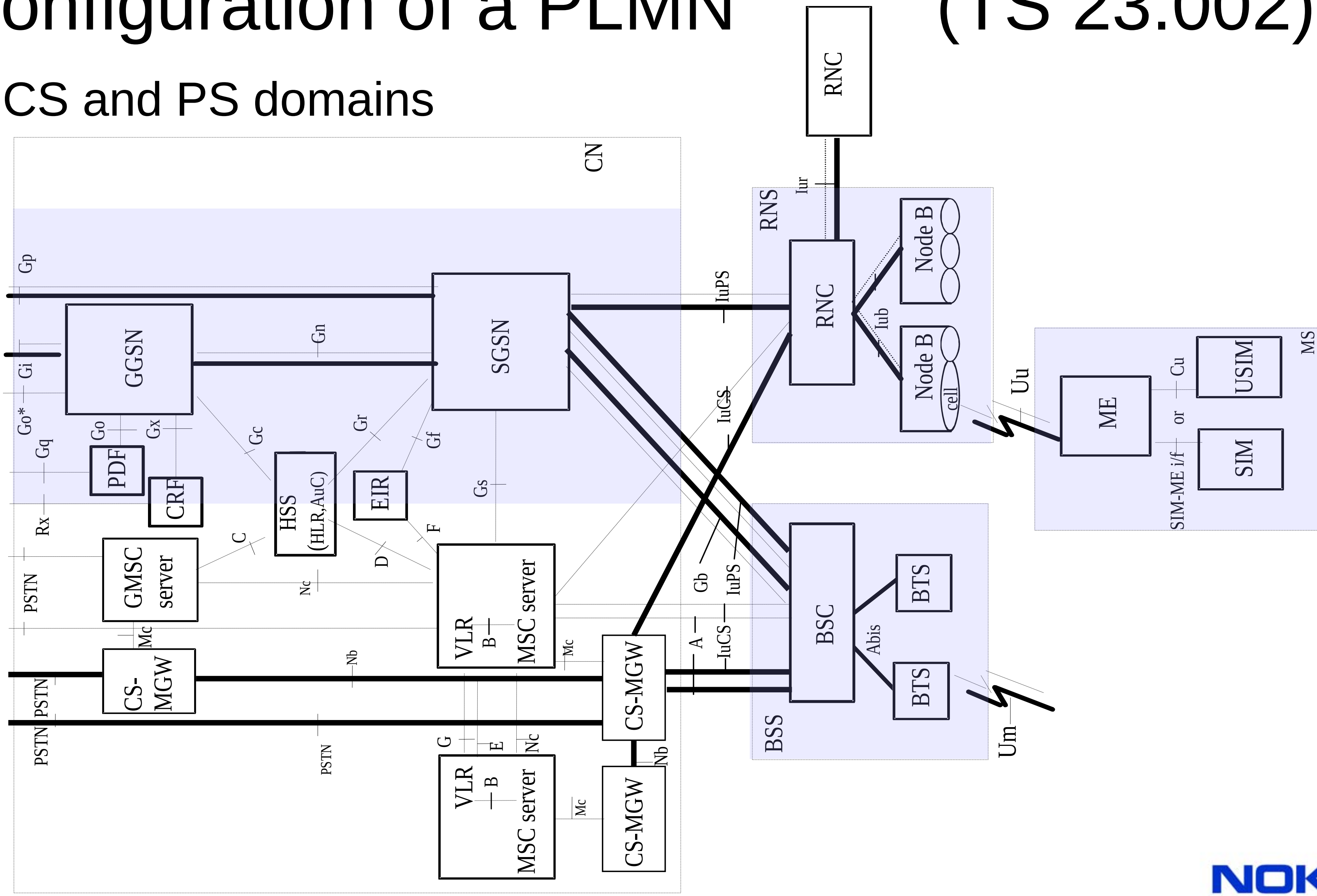


QoS and QoE Management

- Network planning (design)
  - Network dimensioning and detailed network planning
- QoS provisioning (configuration mechanisms)
  - Radio, core and transport QoS configuration
  - Mapping of services onto QoS profiles
  - Application QoS specific information to terminal
- QoE and QoS monitoring (and data analysis)
  - Service level approach using statistical samples
  - Network management system approach using QoS parameters
- Optimizations (performance improvement)
  - Performance measurements
  - Analysis of measurement results
  - Updates of the network/service configuration and parameters

Configuration of a PLMN (TS 23.002)

- CS and PS domains

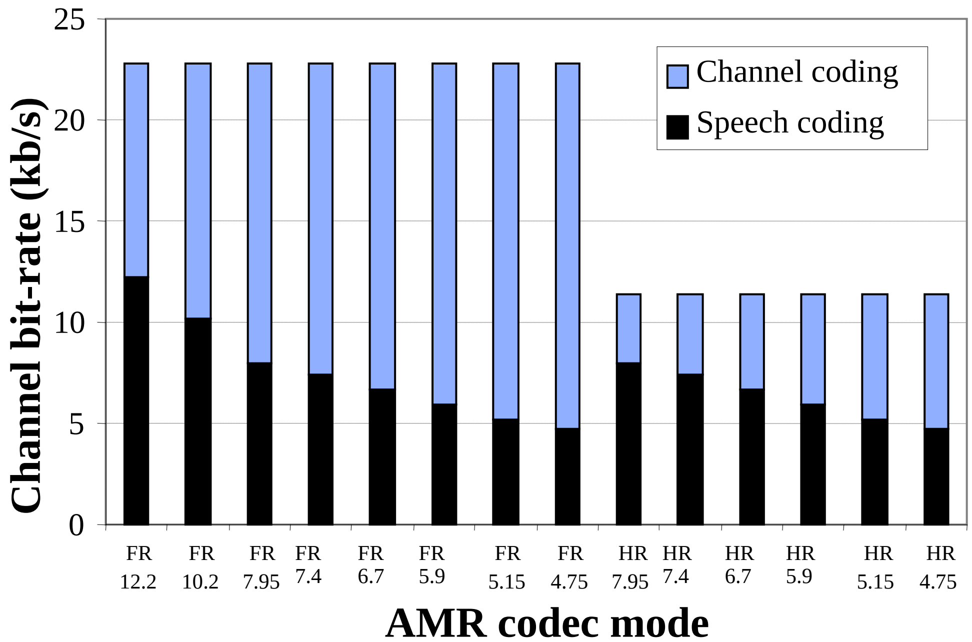


Circuit switched (CS) service applications

- Resources are allocated at service session setup and reserved during the entire session duration
- Examples of CS service applications
  - Emergency calls
  - Short Message Service (SMS)
  - Telephony with Adaptive Multi-Rate (AMR)
  - Multimedia

CS: Telephony with AMR

- **GSM**: full-rate (FR), half-rate (HR) and enhanced full-rate (EFR)
- **3GPP R9**: adaptation of HR or FR and error protection level to radio channel and traffic conditions controlled by operator on a cell-by-cell basis
- **3GPP R5**: Wideband AMR (AMR-WB) with speech quality enhancements, suitable for high-quality audio requirements (50-7000 Hz)



CS: Multimedia

- Based on ITU H.324 terminal
  - Mobile-originating and mobile-terminating call against Integrated Services Digital Network (ISDN) or Public Switched Telephone Network (PSTN) call party
  - Single and multiple numbering
  - In-call modification: from speech to multimedia call (and vice versa) during the call
  - End-to-end user rate negotiation
  - H.324 and H.323 (for PS multimedia) interworking

- Small residual BER (e.g., 10^{-5}) for good quality of experience

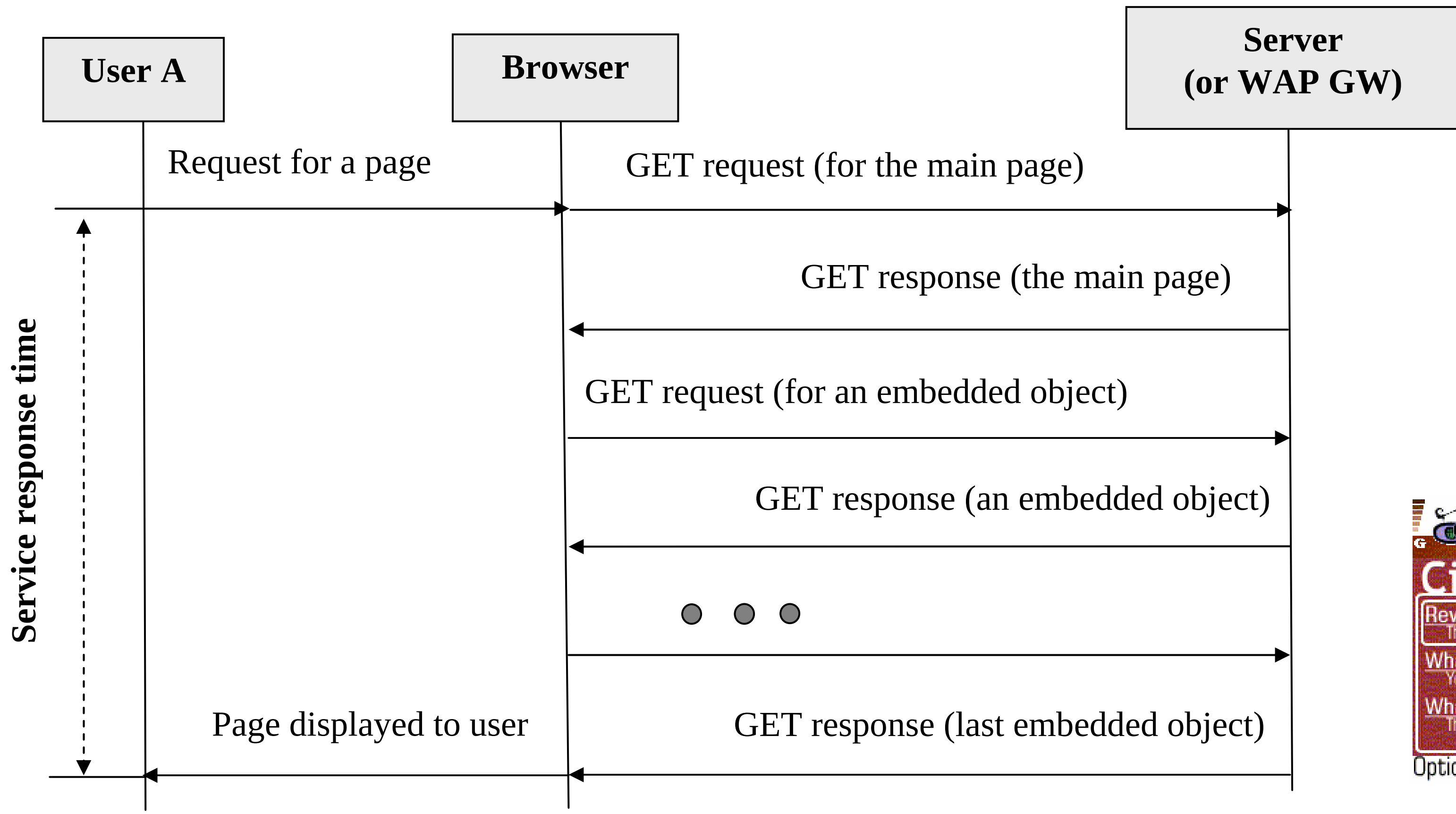
Packet switched service applications

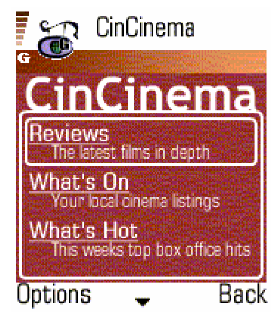
- Resources are dynamically allocated on a need basis for bursty traffic with long idle periods
- Examples of PS service applications
  - Session Initiation Protocol (SIP)
  - Web browsing
  - Multimedia Messaging Service (MMS)
  - Content download
  - Streaming
  - Gaming
  - Business connectivity
  - Push To Talk over Cellular (PoC/PTT)
  - Video Sharing (VS)
  - Voice over IP (VoIP), Presence and Instant Messaging (IM)

PS: Web browsing (1/2)

- Open Mobile Alliance (OMA) browsing enabler is based on Wireless Application Protocol (WAP) standards from the WAP Forum and is migrating towards Internet protocols
- A mobile phone may use:
  - Hyper Text Transfer Protocol (HTTP) 1.1 to communicate directly with a web server
  - Wireless Profile HTTP to communicate with a WAP 2.0 gateway that in turn contacts a Web server, or
  - Wireless Session Protocol (WSP) to communicate with a WAP 1.0 or 2.0 gateway, which in turn contacts a web server
- All three protocols are based on HTTP 1.1 request and response paradigm

PS: Web browsing (2/2)

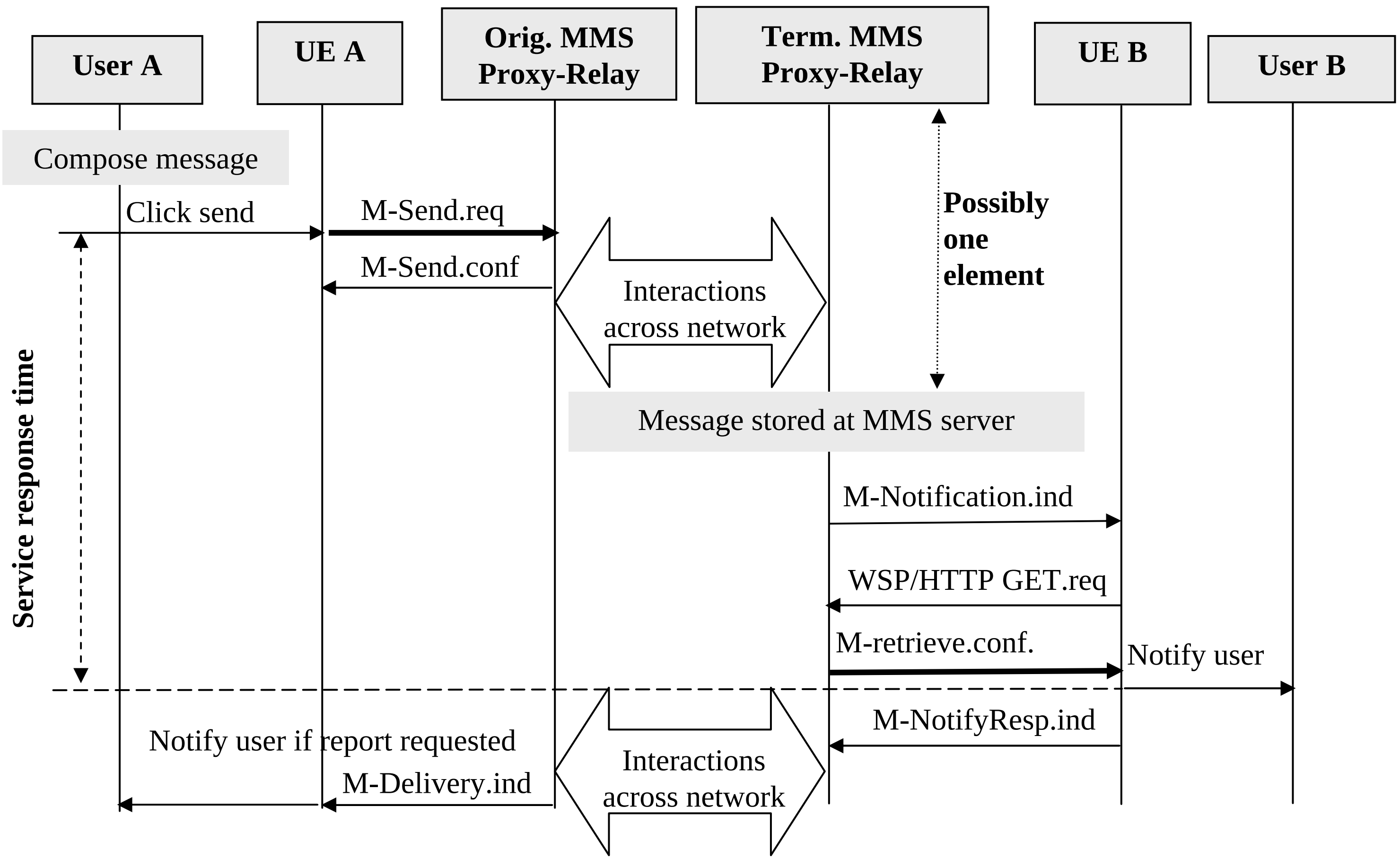


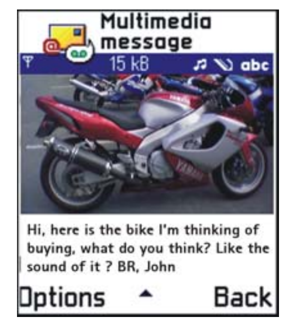


PS: Multimedia Messaging Service (1/2)

- Messaging with rich set of media contents (e.g. image, video) and interoperating with other systems (e.g. Internet email)
- MMS proxy-relay
  - Interacts with MMS clients to provide MMS services
  - Provides access to an MMS server that stores messages
  - Serves as a gateway when interacting with other messaging systems
- Client retrieval
  - Immediate (as soon as a new message notification arrives)
  - Deferred (e.g. when the user asks to read the message)
- Client delivery report (not guaranteed)

PS: Multimedia Messaging Service (2/2)

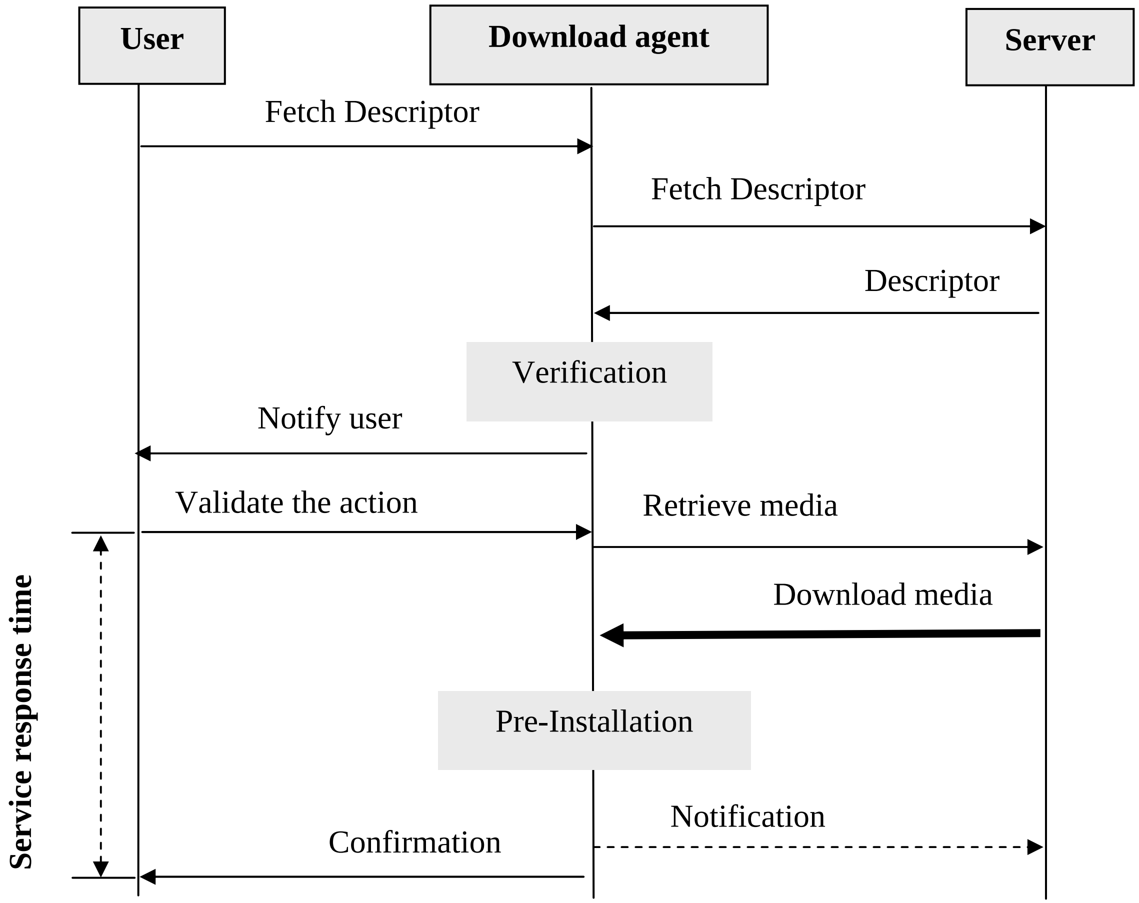


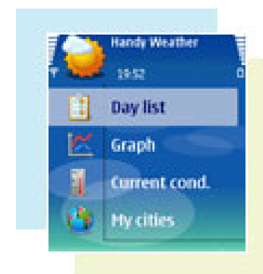


PS: Content download (1/2)

- OMA specification for over-the-air generic content download
  - Download agent: software function in the device responsible for downloading a media object
  - Download descriptor: information about the media object and instructions to the download agent about how to download it
- Two possible scenarios (with notification of transaction status)
  - Separate delivery of download descriptor and media object
  - Co-delivery of download descriptor and media object
- The transfer mechanism or protocol may be HTTP or secure HTTP (HTTPS) but can also be through MMS, email or some instant messaging protocol

PS: Content download (2/2)

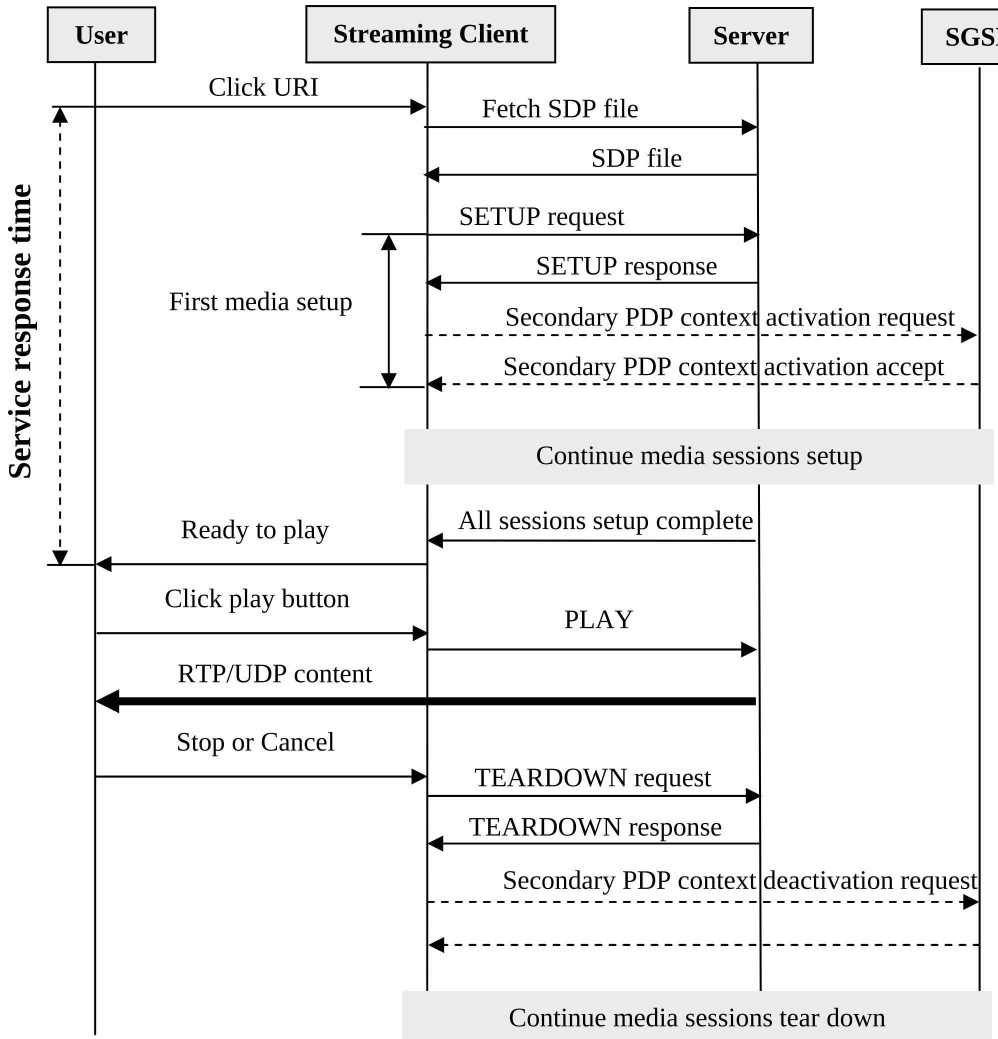




PS: Streaming (1/2)

- Set of one or more streams presented to a user as a complete media feed
- The content is transported using Real time Transport Protocol (RTP) over User Datagram Protocol (UDP)
- Control for session setup and for playing of media (PLAY, PAUSE) is via the Real Time Streaming Protocol (RTSP)
- Actions in the streaming client
  - Obtain a presentation (media streams) description using e.g. MMS, RTSP signaling or Session Description Protocol (SDP)
  - Establish a session for each media (e.g. secondary PDP contexts)

PS: Streaming (2/2)



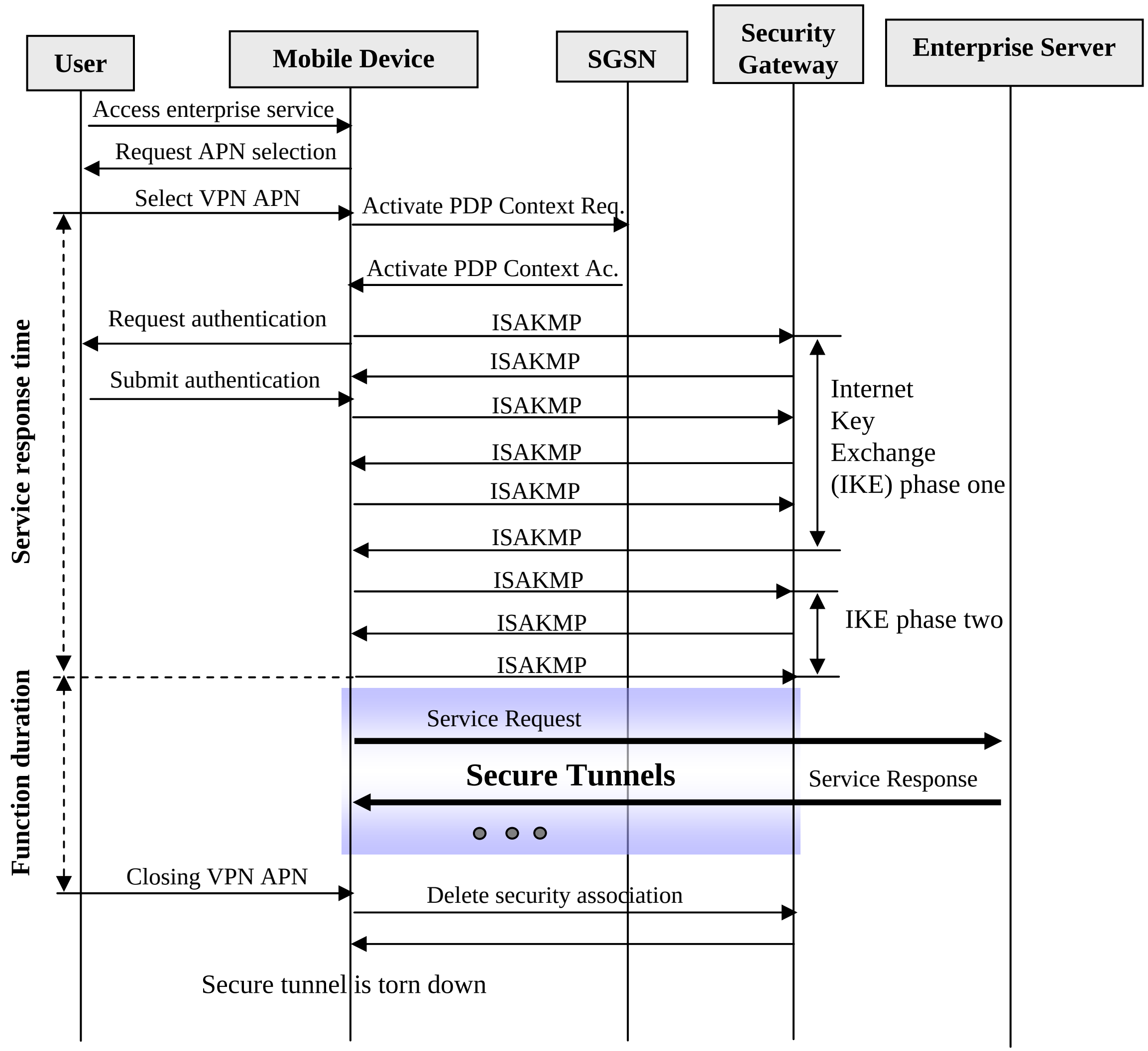
PS: Gaming (1/1)

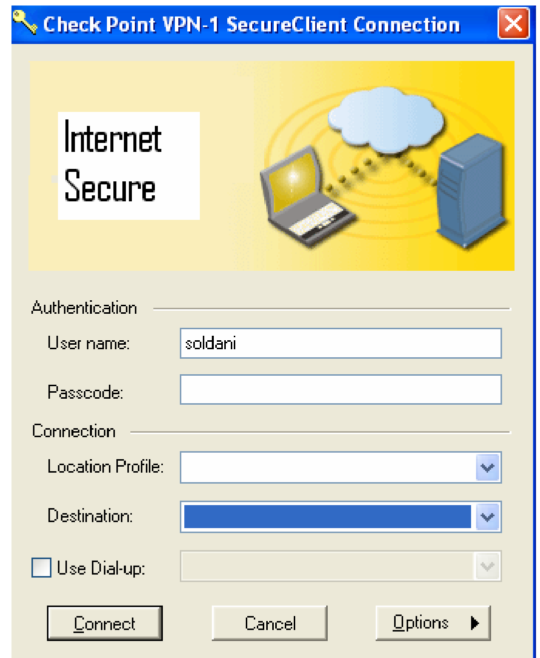
- Scenarios with different performance requirements
  - **Solo game**: a single game player interacts with a game server
  - **Multiplayer game**: multiple players with game rooms in a lobby
- Gaming services
  - **Person-to-person game**: two or more players interact with each other without the intervention of a game server
  - **Server-based game**: server responsible for game synchronization between players, updating the game status to all players, etc.
- Game applications may run on top of different transport protocols: HTTP, TCP, UDP, SMS, WAP push, etc.
- OMA gaming service standardization: gaming architecture, server framework and a client/server
- See www.s60.com

PS: Business connectivity (1/2)

- Enabling end-users to access corporate Intranet or Internet services from a wireless device, in a secure manner, through e.g. EGPRS, WCDMA or WLAN
- Security is ensured with a virtual private network (VPN)
  - **End-to-end security**: encryption between client – enterprise GW
  - **Internet security**: encryption between the mobile operator's domain and enterprise's domain
- IP security (IPsec) protocols protects IP packets by offering
  - **Packet confidentiality** – packets are encrypted before being sent
  - **Packet integrity** – packets are protected so that any alterations can be detected
  - **Packet origin authentication** – packets are protected to ensure that they are indeed from the claimed sender

PS: Business connectivity (2/2)

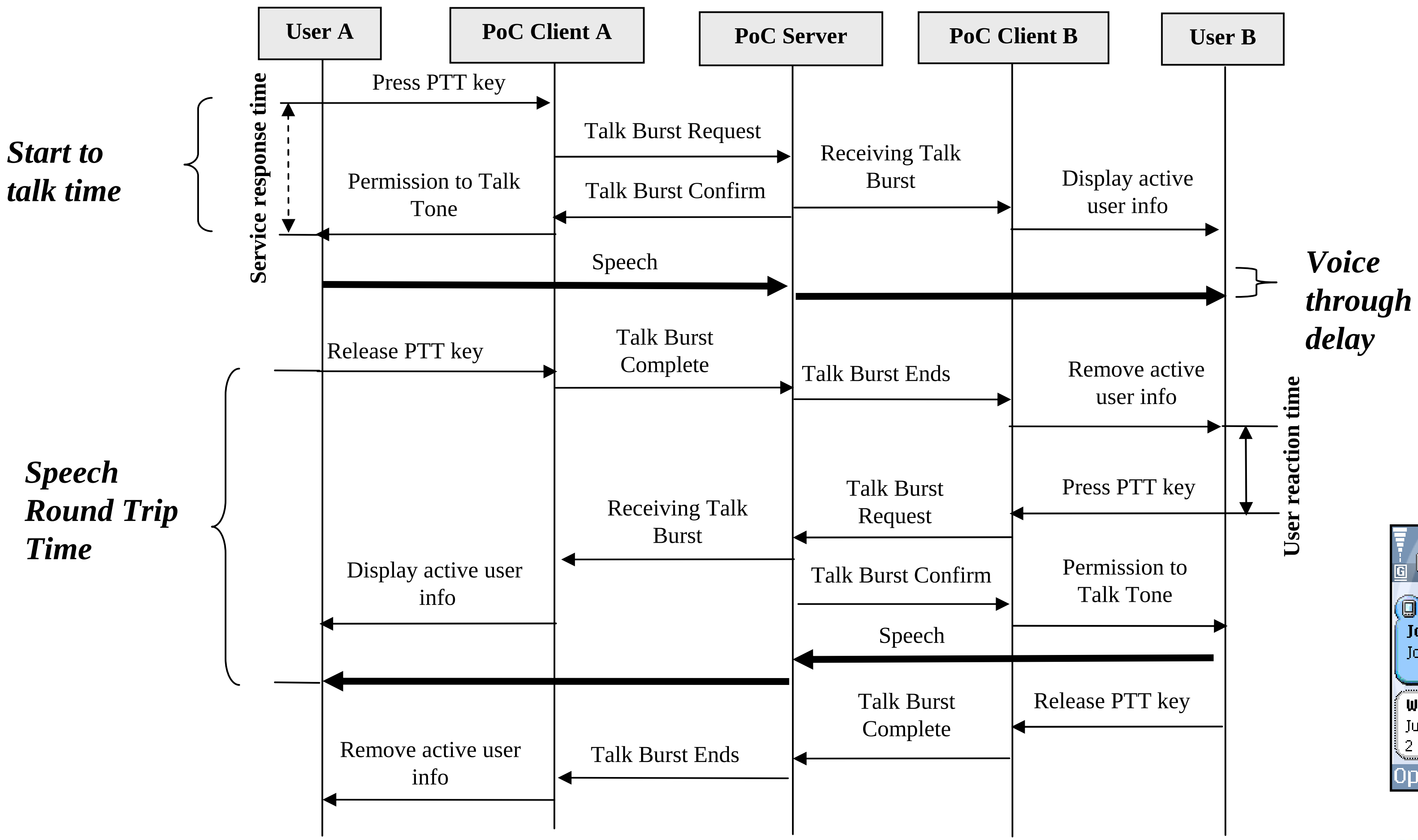




PS: PoC/PTT (1/2)

- Real time one-to-one and one-to-many voice communication service
- OMA specifications
- PoC calls are **one-way communication**: while one person speaks, the other(s) only listens
- PoC server orchestrates the communications
  - Grants floor to clients
  - Queues or rejects permission to send talk bursts
  - Revokes permissions to talk

PS: PoC/PTT (2/2)



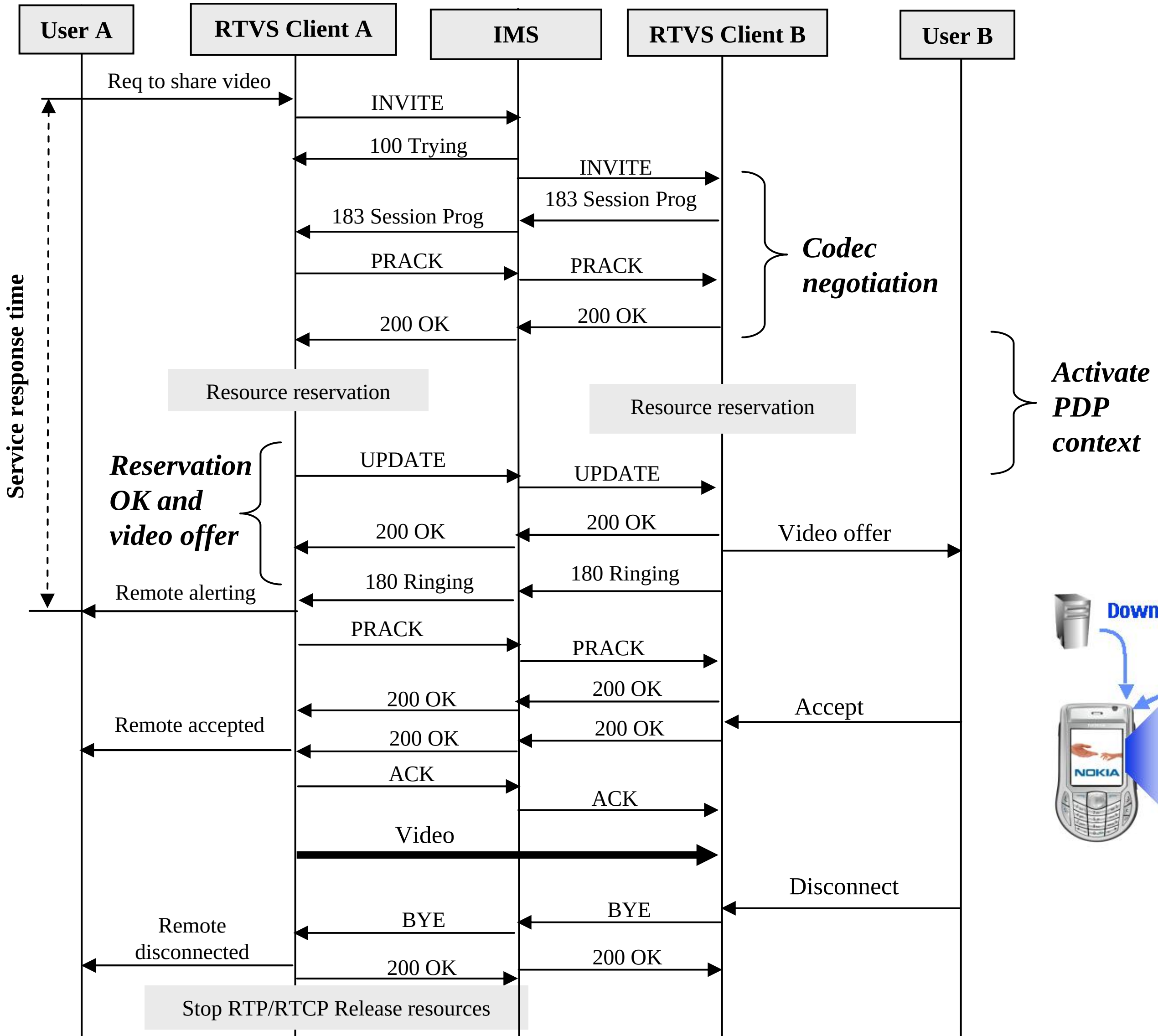
User A
PoC Client A
PoC Server
PoC Client B
User B
Start to talk time
Service response time
Press PTT key
Talk Burst Request
Receiving Talk Burst
Display active user info
Permission to Talk Tone
Talk Burst Confirm
Speech
Voice through delay
Release PTT key
Talk Burst Complete
Talk Burst Ends
Remove active user info
User reaction time
Speech Round Trip Time
Press PTT key
Talk Burst Request
Receiving Talk Burst
Display active user info
Talk Burst Confirm
Permission to Talk Tone
Speech
Release PTT key
Talk Burst Complete
Remove active user info
Talk Burst Ends

Push to Talk
John
John talking
Workmates
Julie
2 min ago
Options
Disconn...

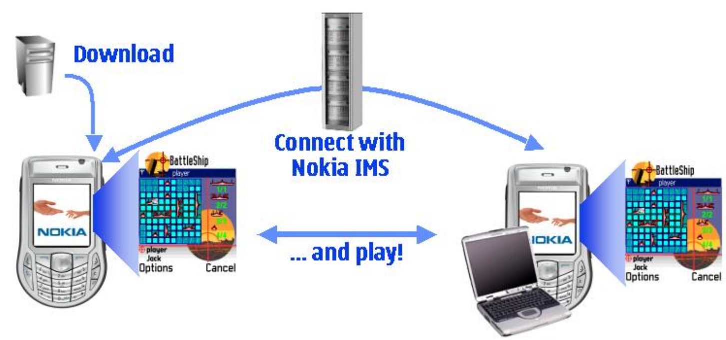
PS: VS (1/2)

- Peer-to-peer, **unidirectional**, multimedia streaming service where at least one of the actors is using a mobile device
- The multimedia data (live video or stored multimedia file) are streamed from one device to the other and are consumed in **real time**, creating the experience of ‘sharing the moment’
- One use case for VS is to enrich a CS voice call by sharing live video or pre-recorded video clips during the voice call
- Not standardized, IMS implementation possible
- Video media are carried by RTP, and **RTCP is used to provide video performance feedbacks** in order to adjust media delivery according to network conditions

PS: VS (2/2)



User A
RTVS Client A
IMS
RTVS Client B
User B
Req to share video
INVITE
100 Trying
INVITE
183 Session Prog
183 Session Prog
PRACK
PRACK
200 OK
200 OK
Codec negotiation
Resource reservation
Resource reservation
Activate PDP context
Service response time
Reservation OK and video offer
UPDATE
UPDATE
200 OK
200 OK
Video offer
180 Ringing
180 Ringing
Remote alerting
PRACK
PRACK
200 OK
200 OK
Accept
Remote accepted
200 OK
200 OK
ACK
ACK
Video
Disconnect
BYE
BYE
Remote disconnected
200 OK
200 OK
Stop RTP/RTCP Release resources



Download
Connect with Nokia IMS
... and play!
NOKIA
BattleShip
Options
Cancel

PS: VoIP (1/1)

- Used over different networks such as fixed broadband (DSL/cable), WLAN (IEEE 802.11) and cellular 3G
- IETF, 3GPP/3GPP2 standard systems use SIP, while other systems use different, non-interoperable protocols
- With 3G networks and handsets, **conversational full-duplex** VoIP services become feasible
- VoIP is not mandatory for conversational-rich communication – Rich Call – services in a cellular network environment
- VoIP service setup in cellular may be similar to VS, in case the session setup uses SIP

PS: Presence (1/1)

- The ability and willingness to be reached for communication is defined by items of information known as ‘presence information’
- Some examples of profile are:
  - Personal status (available, busy, on holiday, in a meeting)
  - Terminal status (switched off, out of coverage, in a videoconference)
  - Terminal capabilities (supports chat and instant messaging)
  - Location (in the office, at home, on-the-move)
  - Personal data (name, address, telephone number, email address)
  - Mood (happy, frustrated, angry, sad)
  - List of content to be shared (games, etc…)

PS: Instant Messaging (1/1)

- IM is defined as the exchange of content between a set of participants in real time
- There are several different messaging schemes
  - One-shot messaging (e.g. MMS) and conversational messaging (e.g. Chat)
  - Session-based messaging in a separate SIP session
- R6 defines even tighter integration of the MMS with the IMS especially for addressing and using SIP as a way to notify the UE of the MMS received

Three aspects for satisfactory QoE

How to efficiently deliver the services in a network with significant load without an overdimensioned network?

Application performance

How do applications work in the network at light load? Are new network features needed? When is the end-user satisfied with the service?

Network quality of service and capacity

Dimensioning, performance monitoring and service management

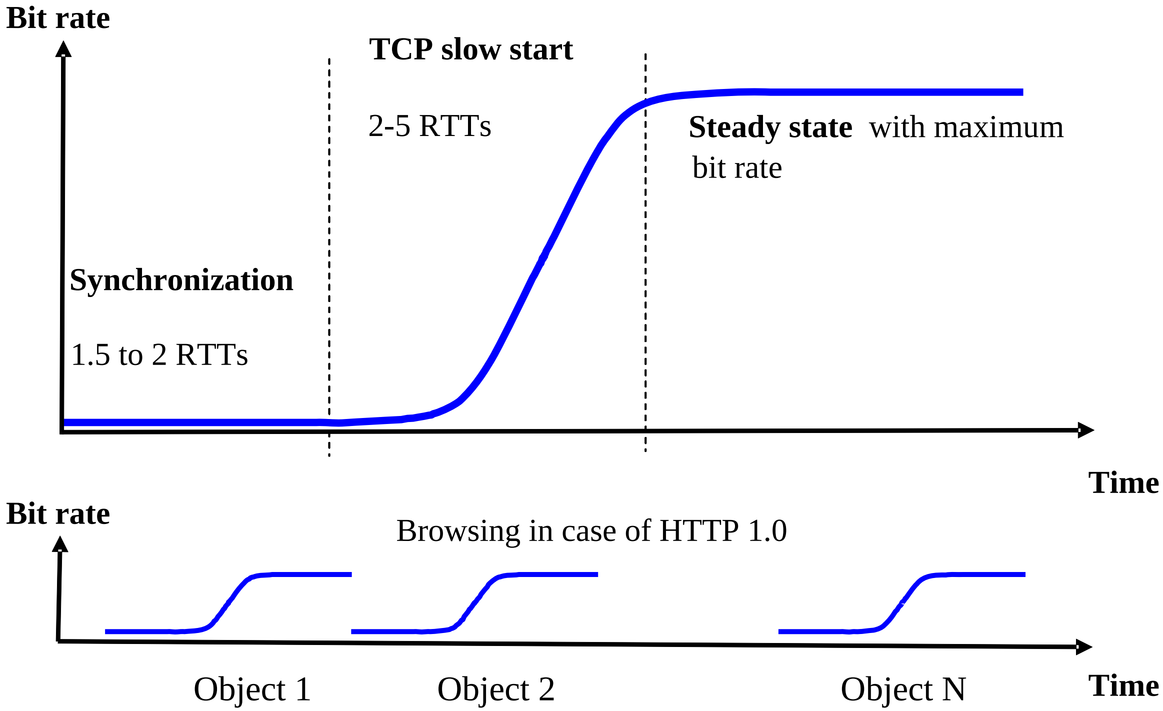
Does the network perform as expected when the original dimensioning was done? Is it possible to differentiate monitoring by services? Are the criteria set for the end-user experience met?

Service performance

Application	KPI	Requirements
Mobile station browsing (Content-to-person)	Click-to-content	Click-to-content delivery time < 4s – 10s. High bit rate, short initial connection setup time and packet round trip time (RTT) < 200 ms
Laptop browsing (Terminal used as modem)	Click-to-content	High bit rates (uplink and downlink), indoor coverage, and packet round trip times. Downlink bit rates ~ 200 – 400 kb/s and packet round trip times < 200 – 300 ms
Downloading (Content-to-person)	Click-to-content	Click-to-content delivery time < 2 minutes
Audio and video streaming (Content-to-person)	Click-to-content Number of breaks during the service delivery Picture/audio quality	Bit rates 64 kb/s – 128 kb/s video streaming 3GPP codec. Content bit rate adaptation improves quality. Breaks in the connection due to mobility < 3s – 5s and small bit rate variations
Push-to-Talk (Person-to-person)	Start-to-talk time Voice-through delay Speech-round-trip time Voice quality	Stable minimum bit rate of around 8 kb/s, start-to-talk time < 1s – 2s, speech round trip delay < 4s. Short initial and subsequent bearer setup times, fast mobility procedures and minimum bit rate guaranteed: always on PDP context
VoIP (Person-to-person)	Mouth-to-ear delay Mean opinion score for the voice quality Call setup time	Mouth-to-ear delays < 200 – 300 ms $\Rightarrow$ packet RTT ~ 150 to 250 ms. Bit rates ~ 16 – 64 kb/s depending on compression and codecs. Call setup time comparable to CS domain of < 7 s, always on PDP contexts
Gaming	Response times and bit rates	Strategy games require packet RTT ~ 500 ms, while action based games require RTT ~ 70 – 200 ms

TPC/IP connection states

- RTT: time it takes to send a small packet from a computer to a server and back again

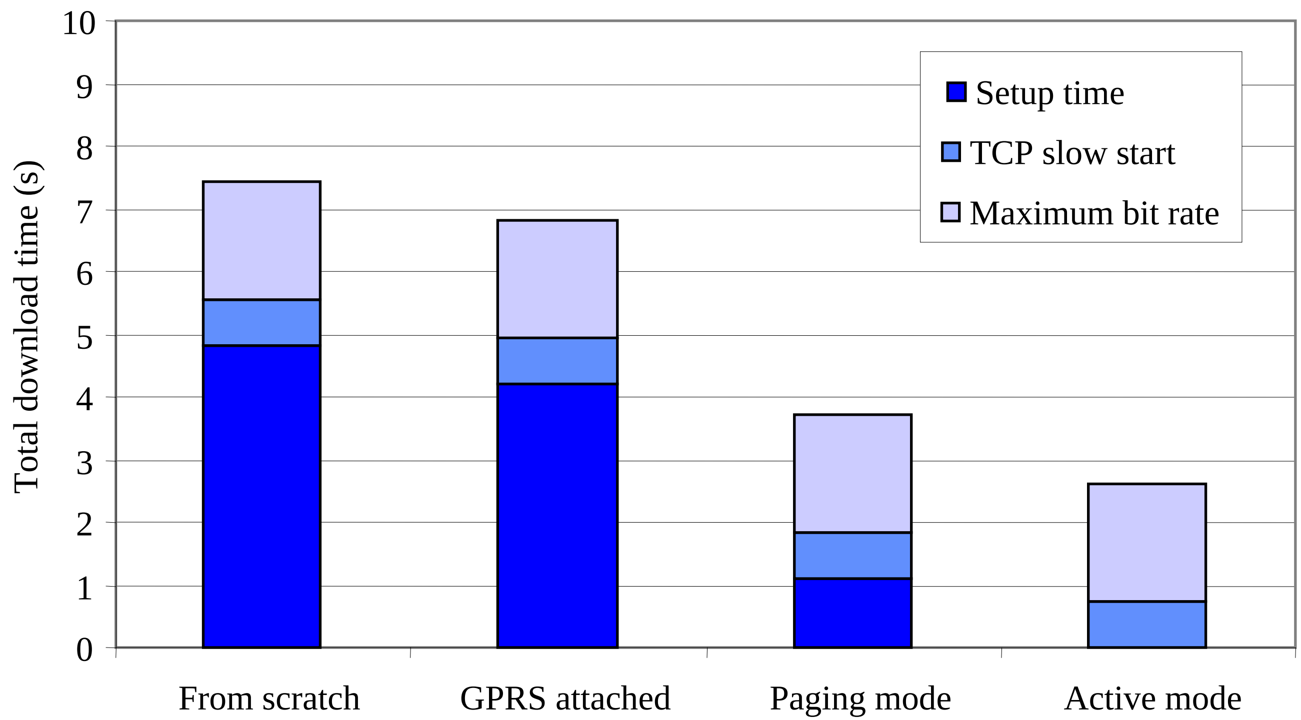


The maximum bit rate provided by TCP is limited by the transmitter/receiver buffer size divided by the RTT

Performance improves with HTTP1.1, where multiple objects may be transferred within the same TCP session

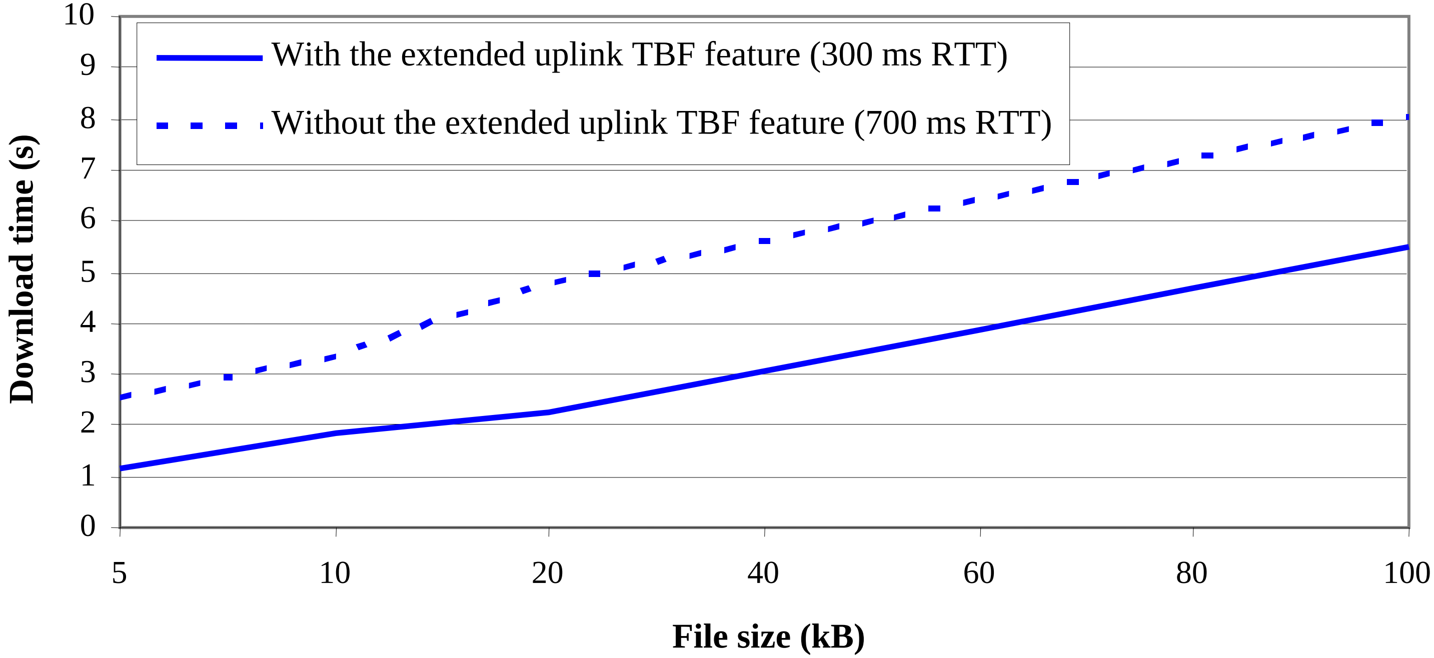
Example of download times in UTRA FDD

- File: 100 kB, DL BR: ≤ 384 kb/s, UL BR: 64–128 kb/s
- RTT < 200 ms, seamless mobility
- Even better performance with HSPA



Example of download times in GERA

- EDGE DL BR: ~ 200 kb/s, UL BR: << 200 kb/s
- RTT < 300-500 ms with “extended UL TBF mode” (MS-BSS), 2x otherwise
- Connection breaks during cell reselection
  - 2-3 s (no routing/location area updates)
  - 0.5 s with Network Assisted Cell Change (NACC)



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  - http://eu.wiley.com/WileyCDA/WileyTitle/productCd-0470016396.html
  - http://www.connecting.nokia.com/NOKIA/nns.nsf/a/78786C61AB5A7C5AC225718F0026BAA3

    (Contact Mr. Geoff Farrell @ Wiley gfarrell@wiley.co.uk)

See also:

  - http://lib.tkk.fi/Diss/2005/isbn9512278340/