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Quality of telecommunication services: concepts, models, objectives and dependability planning – Terms and definitions related to the quality of telecommunication services

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QoS aspects for popular services in mobile networks

Recommendation ITU-T E.804

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Recommendation ITU-T E.804

QoS aspects for popular services in mobile networks

Summary

Recommendation ITU-T E.804 provides sets of quality of service (QoS) parameters from an end-user's perspective for the operational aspects of mobile communication. As services per se are not standardized, it focuses on popular services, which means commonly or widely used services.

This does not preclude applying the definitions in this Recommendation for other (not widely used) services, if feasible.

History

Edition	Recommendation	Approval	Study Group	Unique ID*
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Table of Contents

Page

Page

Introduction

This Recommendation provides quality of service (QoS) parameters definitions for mobile services and related trigger points. Furthermore, it discusses all aspects of practical application thereof, including field testing and statistical considerations.

Although this Recommendation has been produced under the umbrella of the SG12 RG-AFR, readers from other regions are expected to find it useful and applicable in their own regional or local context.

Insofar as technical content in this Recommendation has been adopted from ETSI TS 102 250, *Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in GSM and 3G networks*, ITU acknowledges the copyright of ETSI.

Recommendation ITU-T E.804

QoS aspects for popular services in mobile networks

1 Scope

This Recommendation covers the following quality of service (QoS) aspects for popular services in mobile networks:

– Assessment of QoS
– Definition of QoS parameters and their computation
– Typical procedures for QoS measurement equipment
– Requirements for QoS measurement equipment
– Definition of typical measurement profiles
– Post processing and statistical methods
– Network based QoS measurements.

The Recommendation summarizes the basics of QoS, always seen from the user's perspective. Differences to quality of experience (QoE) are also discussed. In extension to generic definitions, specific definitions for this Recommendation are stated here. Furthermore, it gives guidance to assure that QoS assessments can be conducted in a meaningful way and proposes a corresponding process.

It defines QoS parameters and their computation for popular services in mobile networks. The parameter definition is split into several parts. It contains an abstract definition which gives a generic description of the parameter, an abstract equation and the corresponding user and technical trigger points. The harmonized definitions are considered as prerequisites for the comparison of QoS measurements and measurement results.

The Recommendation also describes the measurement procedures needed to perform the measurements of QoS parameters.

It defines the minimum requirements of QoS measurement equipment for mobile networks in the way that the values and trigger points needed to compute the QoS parameter can be measured following the procedures defined. Test equipment fulfilling the specified minimum requirements will allow performing the proposed measurements in a reliable and reproducible way.

Furthermore, the Recommendation specifies typical measurement profiles that are required to enable benchmarking of different mobile networks both within and outside national boundaries.

In addition, it describes procedures to be used for statistical calculations in the field of QoS measurement of mobile networks using probing systems.

Finally, the Recommendation describes how QoS measurements should be done inside the network without direct access to the end point terminal.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[ITU-T E.800] Recommendation ITU-T E.800 (2008), *Definitions of terms related to quality of service.*

[ITU-T E.802] Recommendation ITU-T E.802 (2007), *Framework and methodologies for the determination and application of QoS parameters.*

[ITU-T G.109] Recommendation ITU-T G.109 (1999), *Definition of categories of speech transmission quality.*

[ITU-T G.1000] Recommendation ITU-T G.1000 (2001), *Communications Quality of Service: A framework and definitions.*

[ITU-T I.350] Recommendation ITU-T I.350 (1993), *General aspects of quality of service and network performance in digital networks, including ISDNs.*

[ITU-T P.10] Recommendation ITU-T P.10/G.100 (2006), *Vocabulary for performance and quality of service*; plus Amendment 2 (2008), *New definitions for inclusion in Recommendation ITU-T P.10/G.100.*

[ITU-T P.56] Recommendation ITU-T P.56 (2011), *Objective measurement of active speech level.*

[ITU-T P.564] Recommendation ITU-T P.564 (2007), *Conformance testing for voice over IP transmission quality assessment models.*

[ITU-T P.800] Recommendation ITU-T P.800 (1996), *Methods for subjective determination of transmission quality.*

[ITU-T P.800.1] Recommendation ITU-T P.800.1 (2006), *Mean Opinion Score (MOS) terminology.*

[ITU-T P.862] Recommendation ITU-T P.862 (2001), *Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs.*

[ITU-T P.862.1] Recommendation ITU-T P.862.1 (2003), *Mapping function for transforming P.862 raw result scores to MOS-LQO.*

[ITU-T P.862.2] Recommendation ITU-T P.862.2 (2007), *Wideband extension to Recommendation P.862 for the assessment of wideband telephone networks and speech codecs.*

[ITU-T P.862.3] Recommendation ITU-T P.862.3 (2007), *Application guide for objective quality measurement based on Recommendations P.862, P.862.1 and P.862.2.*

[ITU-T P.863] Recommendation ITU-T P.863 (2011), *Perceptual objective listening quality assessment.*

[ITU-T X.745] Recommendation ITU-T X.745 (1993), *Information technology – Open Systems Interconnection – Systems Management: Test management function.*

[ITU-R BS.1387-1] Recommendation ITU-R BS.1387-1 (2001), *Method for objective measurements of perceived audio quality.*

[ITU-R BT.500-13] Recommendation ITU-R BT.500-13 (2012), *Methodology for the subjective assessment of the quality of television pictures.*

[ITU-R BT.1359-1] Recommendation ITU-R BT.1359-1 (1998), *Relative timing of sound and vision for broadcasting.*

[ETSI EN 300 392-2] ETSI EN 300 392-2 V3.4.1 (2010), *Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 2: Air Interface (AI).*

[ETSI EN 300 392-5] ETSI EN 300 392-5 V2.2.0 (2010), *Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D) and Direct Mode Operation (DMO); Part 5: Peripheral Equipment Interface (PEI).*

[ETSI TR 102 505] ETSI TR 102 505 V1.3.1 (2012), *Speech and multimedia Transmission Quality (STQ); Development of a Reference Web page*.

[ETSI TR 102 493] ETSI TR 102 493 V1.2.1 (2009), *Speech and multimedia Transmission Quality (STQ); Guidelines for the use of Video Quality Algorithms for Mobile Applications.*

[ETSI TR 102 581] ETSI TR 102 581 V1.1.1 (2007), *Speech Processing, Transmission and Quality Aspects (STQ); A Study on the Minimum Additional Required Attenuation on the Antenna Path of the Field Test Equipment.*

[ETSI TS 100 910] ETSI TS 100 910 V8.20.0 (2005), *Digital cellular telecommunications system (Phase 2+); Radio Transmission and Reception (3GPP TS 05.05 version 8.20.0 Release 1999).*

[ETSI TS 123 228] ETSI TS 123 228 V11.10.0 (2013), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Subsystem (IMS); Stage 2 (3GPP TS 23.228* version 11.10.0 Release 11*).*

[ETSI TS 124 008] ETSI TS 124 008 V8.19.0 (2013), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Mobile radio interface Layer 3 specification; Core network protocols; Stage 3 (3GPP TS 24.008* version 8.19.0 Release 8*).*

[ETSI TS 125 304] ETSI TS 125 304 V10.7.0 (2013), *Universal Mobile Telecommunications System (UMTS); User Equipment (UE) procedures in idle mode and procedures for cell reselection in connected mode (3GPP TS 25.304 version 10.7.0 Release 10).*

[ETSI TS 127 005] ETSI TS 127 005 V9.0.1(2011), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Use of Data Terminal Equipment – Data Circuit terminating Equipment (DTE-DCE) interface for Short Message Service (SMS) and Cell Broadcast Service (CBS) (3GPP TS 27.005 version 9.0.1 Release 9).*

[ETSI TS 127 007] ETSI TS 127 007 V.9.9.0 (2013), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; AT command set for User Equipment (UE) (3GPP TS 27.007 Version 9.9.0 Release 9).*

[ETSI TS 129 002] ETSI TS 129 002 V9.12.0 (2013), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Mobile Application Part (MAP) specification (3GPP TS 29.002 version 9.12.0 Release 9).*

[ETSI TS 145 008] ETSI TS 145 008 V6.22.0 (2013), *Digital cellular telecommunications system (Phase 2+); Radio subsystem link control (3GPP TS 45.008 version 6.22.0 Release 6).*

[IETF RFC 1939] IETF RFC 1939 (1996), *Post Office Protocol – Version 3.*

[IETF RFC 2177] IETF RFC 2177 (1997), *IMAP4 IDLE command.*

[IETF RFC 2326] IETF RFC 2326 (1998), *Real Time Streaming Protocol (RTSP).*

[IETF RFC 2821] IETF RFC 2821 (2001), *Simple Mail Transfer Protocol.*

[IETF RFC 3481] IETF RFC 3481 (2003), *TCP over Second (2.5G) and Third (3G) Generation Wireless Networks.*

[IETF RFC 3903] IETF RFC 3903 (2004), *Session Initiation Protocol (SIP) Extension for Event State Publication*.

[OMA-1] OMA-AD-PoC-V2_0-20110802.pdf (2011), *Push to talk over Cellular (PoC) – Architecture.*

[OMA-2] OMA-TS-PoC_UserPlane-V2_0-20110802-A.pdf (2011), *PoC User Plane.*

[OMA-3] OMA-TS-PoC_ControlPlane-V2_0-20110802-A.pdf (2011), *OMA PoC Control Plane.*

[OMA-4] OMA-ERELD-POC-V2_0-20110802-A.pdf (2011), *Enabler Release Definition for Push-to-Talk over Cellular*.

3 Definitions

3.1 Terms defined elsewhere

This Recommendation uses terms defined in [ITU-T E.800], [ITU-T E.802], [ITU-T G.1000] and [ITU-T P.10].

3.2 Terms defined in this Recommendation

This Recommendation defines the following terms:

3.2.1 1-1 PoC session: Feature enabling a push to talk over cellular (PoC) user to establish a PoC session with another PoC user.

3.2.2 1 kbyte: 1 024 byte.

3.2.3 1 Mbyte: 1 024 byte.

3.2.4 A-Party: Initiating part of a connection (also: Mobile Originating, MO) or in direct transactions, the party initiating the transaction (calling party).

NOTE – In store-and-forward transactions, the party sending the content.

3.2.5 access point name: Is used to identify a specific IP network and a point of interconnection to that network.

3.2.6 ad hoc poc group session: Push to talk over cellular (PoC) session for multiple PoC users that does not involve the use or definition of a pre-arranged or chat PoC group.

3.2.7 AT interface: Interface within a user equipment (UE) between a terminal equipment (TE), which can be an external measurement equipment, and a mobile termination (MT) used for sending Attention (AT) commands from the TE to the MT and receiving responses or indications from the MT at the TE.

NOTE 1 – The AT interface is commonly referred to as R reference point.

NOTE 2 – In TETRA, the AT interface is referred to as peripheral equipment interface (PEI), see [ETSI EN 300 392-5].

3.2.8 automatic answer: Accepting the invitation automatically by a terminal if resources are available.

3.2.9 B-Party: In direct transactions, the termination or counterpart of a transaction.

NOTE – In store-and-forward transactions, the party receiving the content.

3.2.10 bearer: Resource in the broadcast transport system that allows the transmission of data to the terminal or from the terminal.

NOTE – In this Recommendation, there is a distinction between broadcast bearer and mobile network bearer. The latter one is synonymously referred to as interactivity channel.

3.2.11 benchmark: Evaluation of performance value(s) of a parameter or set of parameters for the purpose of establishing value(s) as the norm against which future performance achievements may be compared or assessed.

3.2.12 bootstrapping: Mechanism where the broadcast signal is accessed for the first time within a service usage.

NOTE – Parts of this procedure are the synchronization to the signal and its decoding so that afterwards a list of available channels is accessible and presented to the user.

3.2.13 bootstrapping bearer: Bearer on which the bootstrapping procedure is executed.

3.2.14 broadcast: Information transfer from one transmitting entity to many receiving entities.

3.2.15 broadcast bearer: Bearer supporting the broadcast service (e.g., DVB-H, MBMS, etc.).

NOTE – The broadcast signal is transmitted via this bearer.

3.2.16 chat PoC group: Persistent group in which each member individually joins the push to talk over cellular (PoC) session, i.e., the establishment of a PoC session to a chat PoC group does not result in other members of the chat PoC group being invited.

3.2.17 chat PoC group session: Push to talk over cellular (PoC) session established to a chat PoC group.

3.2.18 confirmed indication: Signalling message returned by the push to talk over cellular (PoC) server to confirm that the PoC server, all other network elements intermediary to the PoC server and a terminating terminal are able and willing to receive media.

3.2.19 content: Entirety of information transferred within a transaction, seen from the user's perspective.

NOTE – In case of services requiring entrance procedures (e.g., server login with FTP), information flow to achieve the state of being able to transfer actual user data is not counted as content.

3.2.20 cut-off: Unintended termination of a communication session.

3.2.21 data service: Telecommunications service involving the transport of data via the network termination point (PTN) such that any user can use equipment connected to a network termination point to exchange data with another user of the equipment connected to another termination point.

3.2.22 direct service: Service which makes use of direct communications between a client entity and a server entity without persistent storage of transferred data in interconnected network elements.

3.2.23 direct transaction: Real time transaction between two entities.

3.2.24 download: Transfer of data or programs from a server or host computer to one's own computer or device.

3.2.25 drive test tool: End-point test tool which is designed to be moved around, i.e., by walking or driving a car.

3.2.26 e-mail: Messages automatically passed from one computer user to another, often through computer networks and/or via modems over telephone lines.

3.2.27 end-point test tool: Typically especially designed mobile which uses active test calls to collect measurements.

3.2.28 end-to-end quality: Quality related to the performance of a communication system, including all terminal equipment. For voice services it is equivalent to mouth-to-ear quality.

3.2.29 ESG retrieval bearer: Bearer that is used to retrieve the electronic service guide (ESG) information.

3.2.30 event: In this Recommendation, an event is understood as a change of condition (the corresponding point of time is considered in addition).

3.2.31 host: An entity that provides client stations with access to files and printers as shared resources to a computer network.

3.2.32 idle mode: A communication device is in this state when it is powered-on but not transmitting a signal.

3.2.33 IP service access: Basic access to the generic packet-data transfer capabilities the service is based upon.

3.2.34 landing page: The first website that appears in the Internet browser when a user tries to browse the Internet. It is often used to allow the user to make some specific settings for the following Internet session.

3.2.35 last data packet: Packet that is needed to complete the transmission of the content on the receiving side.

NOTE – For FTP download, the last data packet contains a set TCP FIN flag bit.

3.2.36 manual answer: Push to talk over cellular (PoC) user accepts the invitation manually.

3.2.37 maximum expected delivery time: For store-and-forward services, this defines the time span within which a message shall be received by the B-party to rate the transaction successful from the user's perspective.

3.2.38 mean data rate: Average data rate of a data transmission, calculated by dividing the number of transmitted bits by the duration of the transmission.

3.2.39 mobile broadcast service: End-to-end system for delivery of any types of digital content and services towards a mobile terminal using IP-based mechanisms.

3.2.40 mobile network bearer: Bearer provided by a mobile network operator (e.g., GSM, GPRS, UMTS, etc.) to establish interactivity within the mobile broadcast service.

3.2.41 network access: Access to the network under test.

3.2.42 network accessibility: The probability that the user of a service after a request (to a network) receives the proceed-to-select signal within specified conditions.

3.2.43 network availability: Probability of success of network functions performed by a network over a specified time interval.

3.2.44 network operator: Organization that provides a network for the provision of a public telecommunication service.

3.2.45 on-demand session: Push to talk over cellular (PoC) session set-up mechanism in which all media parameters are negotiated at PoC session establishment.

NOTE – The on-demand sessions are defined by the OMA PoC specification [OMA-1, OMA-2, OMA-3, OMA-4] as mandatory for PoC enabled user equipment, whereas pre-established sessions are defined as optional.

3.2.46 PoC session: Established connection between push to talk over cellular (PoC) users where the users can communicate using speech one at a time.

3.2.47 PoC user: User of the push to talk over cellular (PoC) service.

3.2.48 pre-arranged PoC group session: Persistent push to talk over cellular (PoC) session that has an associated set of PoC members.

NOTE – The establishment of a PoC session to a pre-arranged PoC group results in inviting all members of the defined group.

3.2.49 pre-established session: SIP session established between the terminal and the push to talk over cellular (PoC) server that performs the participating PoC function.

NOTE – The terminal establishes the pre-established session prior to making requests for PoC sessions to other PoC users.

3.2.50 probing attempt: Trial to examine if the service under test works as expected.

3.2.51 QoS indicator: A characteristic that is used to determine the quality of service (QoS).

3.2.52 rate: Change of amount of a quantity divided by the portion of time during which it has been changed.

NOTE – The denominator's unit is related to time.

3.2.53 ratio: Measurement result which represents a subgroup of all single measurements is related to the total number of executed single measurements.

NOTE – Usually, nominator and denominator share the same unit, namely a counter for measurements (subgroup/all).

3.2.54 reliability: The probability that an item can perform a required function under stated conditions for a given time interval.

3.2.55 retrieval: Transport of content from network to B-party, initiated by the B-party.

3.2.56 service access: A set of functions offered to a user by an organization constitutes a service.

3.2.57 service family: Group of services having main characteristics in common.

Example: Speech and video telephony, as well as short message service (SMS) and multimedia messaging service (MMS), are assumed to form a service family.

3.2.58 service integrity: The degree to which a service is provided without excessive impairments, once obtained.

3.2.59 service retainability: Service retainability describes the termination of services (in accordance with or against the will of the user).

3.2.60 session: Continuous usage of a given service, e.g., a speech call or a data session.

3.2.61 session time: Duration of a session.

3.2.62 set-up: The period starting when the address information required for setting up a call is received by the network (recognized on the calling user's access line) and finishing when the called party busy tone, or ringing tone or answer signal is received by the calling party (i.e., recognized on the calling user's access line). Local, national and service calls should be included, but calls to other licensed operators (OLOs) should not, as a given operator cannot control the QoS delivered by another network.

3.2.63 stationary tool: End-point test tool that is installed in a fixed location.

3.2.64 store-and-forward: Store-and-forward services are services where content is stored in the network and delivered to the recipient at a later point in time.

3.2.65 store-and-forward transaction: Transaction where information is sent from party A to party B using an entity C to store information sent from A and attempting to deliver it to B.

3.2.66 streaming: Multimedia data (usually combinations of voice, text, video and audio) transferred in a stream of packets that are interpreted and rendered, by a software application as the packets arrive.

3.2.67 talk burst: Flow of media, e.g., some seconds of speech, from a terminal while that has the permission to send media.

3.2.68 talk burst control: Control mechanism that arbitrates requests from the terminals, for the right to send media.

3.2.69 TBCP talk burst granted: Used by the push to talk over cellular (PoC) server to notify the terminal that it has been granted permission to send a talk burst.

3.2.70 TBCP talk burst idle: Used by the push to talk over cellular (PoC) server to notify all terminals that no one has the permission to send a talk burst at the moment and that it may accept the "TBCP Talk Burst Request" message.

3.2.71 TBCP talk burst request: Used by the terminal to request permission from the push to talk over cellular (PoC) server to send a talk burst.

3.2.72 terminal: A push to talk over cellular (PoC) enabled user equipment implementing a PoC client.

3.2.73 test case: Consists of a number of single identical transactions.

3.2.74 timeout: Specified period of time that will be allowed to elapse in a system (e.g., inactivity) before a specified event is to take place.

3.2.75 transaction: Single, complete, typical usage of a particular service.

3.2.76 transaction result: Set (list) of possible outcomes for a particular transaction.

NOTE – Services belonging to the same service family share the same set of transaction results.

3.2.77 trigger: Always defined with respect to a particular transaction – An event adopting one of the trigger roles for a particular transaction.

3.2.78 trigger point: Absolute time (also called "point in time") of occurrence of a trigger event.

NOTE – This term is widely used in a less restrictive manner, meaning either the event or its time of occurrence, respectively, depending on the context. A specific event adopting one of the trigger roles for a particular transaction is often referred to as "technical trigger point" (for this transaction). It occurs/is measured at a specific point of control and observation (PCO).

3.2.79 unconfirmed indication: Indication returned by the push to talk over cellular (PoC) server to confirm that it is able to receive media and believes the terminal is able to accept media.

NOTE – The PoC server sends the unconfirmed indication prior to determining that all egress elements are ready or even able to receive media.

3.2.80 user equipment: Technical device in user's possession, used for communication purposes.

3.2.81 video: A signal that contains timing/synchronization information as well as luminance (intensity) and chrominance (colour) information that when displayed on an appropriate device gives a visual representation of the original image sequence.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

3G	Third-generation
AAL2	Asynchronous transfer mode Adaptation Layer type 2
ACK	Acknowledgement
ACM	Address Complete message
ALCAP	Access Link Control Application Protocol
AM	Acknowledged Mode

AMR	Adaptive Multi-Rate
AOV	Angles Of View
AP	Access Point
APN	Access Point Name
ARMA	Auto-Regressive Moving Average
AT	Attention Command
ATA	Attention Answer
ATD	Attention Dial
ATH	Attention Hang-up
AVGn	Averaging Operator (regarding n days)
BH	Busy Hour
BSC	Base Station Controller
CC	Call Control
CD	Call Duration
CCCH	Common Control Channel
CDF	Cumulative Distribution Function
CI	Cell Identity
CLI	Calling Line Identity
CLIP	Calling Line Identity Presentation
CMCE	Circuit Mode Control Entity
CPN	Calling Party Number
CRLF	Carriage Return Line Feed
CS	Circuit Switched
CSCF	Call Session Control Function
CSD	Circuit Switched Data
CTR	Controller
CUSUM	Cumulated SUM
CUT	PoC session Cut-off
DCCH	Dedicated Control Channel
DCE	Data Circuit-terminating Equipment
DCH	Data Channel
DCH-FP	Data Channel Frame Protocol
DELAY	talk burst Delay (PoC)
DeREG	PoC Deregistration (PoC)
DL	Down Link
DLDT	Downlink Direct Transfer
DNS	Domain Name Service

DP	Detection Point
DRM	Digital Rights Management
DROP	talk burst Drop (PoC)
DT	Direct Transfer
DTE	Data Terminal Equipment
DVB-H	Digital Video Broadcasting – Handheld
EPG	Electronic Program Guide
ESG	Electronic Service Guide
EWMA	Exponentially Weighted Moving Average
FACH	Forward Access Channel
FEC	Forward Error Correction
FIN	TCP Finish flag
FQDN	Fully Qualified Domain Name
FQT	Fixed QoS Test equipment
FR	Failure Ratio
FTP	File Transfer Protocol
GC	Group Call
GGSN	Gateway GPRS Support Node
GMM	GPRS Mobility Management
GMSC	Gateway Mobile Switching Centre
Gn	Interface between GSN nodes
GPRS	General Packet Radio Service
GPS	General Positioning System
GR	GPRS Register
GSM	Global System for Mobile communications
GSN	GPRS Support Node
GW	Gateway
HLR	Home Location Register
HSPDA	High Speed Downlink Packet Access
HSS	Home Subscriber Server
HSUPA	High Speed Uplink Packet Access
HTML	Hypertext Markup Language
HTTP	Hypertext Transfer Protocol
IAM	Initial Address message
ICE	Interactive Connectivity Establishment
I-CSCF	Interrogating-CSCF
ICMP	Internet Control Message Protocol

IE	Information Element
IMAP	Internet Messaging Access Protocol
IMEI	International Mobile Equipment Identity
IMS	IP Multimedia Subsystem
IMSI	International Mobile Subscriber Identity
INIT	PoC session Initiation
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ISUP	ISDN User Part
Iub	Interface between RNC and Node B
IWMSC	Interworking Mobile-services Switching Centre
JPG	Joint Photographic experts Group
KPI	Key Performance Indicator
L1	Layer 1
LAC	Location Area Code
LC	Local Control
LEAVE	PoC session Leaving
LLC	Logical Link Control
LSL	Lower Specification Level
MAC	Media Access Control
MAWD	Monthly Average Working Day
MBMS	Multimedia Broadcast/Multicast Service
MCC	Mobile Country Code
MD5	Message-Digest algorithm 5
MEDT	Maximum Expected Delivery Time
MGW	Media Gateway
MIME	Multipurpose Internet Mail Extensions
mle	mobile link entity
MM	Mobility Management
MMI	Man Machine Interface
MMQ-Plot	Median-Mean-Quantile Plot
MMS	Multimedia Messaging Service
MMSC	Multimedia Messaging Service Centre
MNC	Mobile Network Code
MO	Mobile Originating
MOC	Mobile Originated Call

MOF	Mobile Originated to Fixed
MOM	Mobile Originated to Mobile
MOS LQO	Mean Opinion Score Listening speech Quality Objective
MOS	Mean Opinion Score
MQT	Mobile QoS Test equipment
MQT-LC	Mobile QoS Test equipment Local Control
MQT-RC	Mobile QoS Test equipment Remote Control
MS	Mobile Station
MSC	Mobile Switching Centre
MSISDN	Mobile Station ISDN number
MSN	Mobile Station Number
MSRP	Message Session Relay Protocol
MT	Mobile Termination
MTAS	Multimedia Telephony Application Server
MTM	Mobile Terminated, originator is also a Mobile unit
MTSI	Multimedia Telephony Service for IMS
MTU	Maximum Transmission Unit
NAT	Network Address Translation
NBAP	Node B Application Part
NE	Network Element
NP	Network Performance
OLO	Other Licensed Operator
OS	Operating system
OMA	Open Mobile Alliance
PC	Personal Computer
P-CSCF	Proxy-CSCF
PCO	Point of Control and Observation
PDF	Probability Density Function
PDN	Packet Data Network
PDP	Packet Data Protocol
PDU	Protocol Data Unit
PEI	Peripheral Equipment Interface
PEP	Performance Enhancement Proxy
PLMN	Public Land Mobile Network
PMN	Public Mobile Network
PoC	Push to talk over Cellular
POP3	Post Office Protocol Version 3

POR	Point Of Recording
PROC	Processor
PS	Packet Switched
PSD	Packet Switched Data
PSS	Packet-switched Streaming Service
PSTN	Public Switched Telephone Network
PTN	Network Termination Point
PtS	Push to Speech
PTT	Push To Talk
PUB	PoC Publish
PWR	Power Supply
QoE	Quality of Experience
QoS	Quality of Service
QoSD	Quality of Service Delivered
QoSE	Quality of Service Experienced
QoSO	Quality of Service Offered
QoSR	Quality of Service Required
QQ-Plot	Quantile-Quantile Plot
QSI	Quality Sequence Indicator
RAB	Radio Access Bearer
RACH	Random Access Channel
RAN	Radio Access Network
RANAP	Radio Access Network Application Protocol
RAS	Remote Access Service
RC	Remote Control
REG long	PoC Registration and publish
REG	PoC Registration
REL	Release message
RF	Radio Frequency
RLC	Radio Link Control
RNC	Radio Network Controller
RR	Radio Resources
RRC	Radio Resource Control
RTCP	Real-Time Control Protocol
RTP	Real=time Transport Protocol
RTSP	Real-Time Streaming Protocol
RX	Reception

SBC	Session Border Controller
SC	Service Centre
S-CSCF	Serving-CSCF
SCCP	Signalling Connection Control Part
SDCCH	Stand-alone Dedicated Control Channel
SDP	Session Description Protocol
SDS	Short Data Service
SDSC	Short Data Service Centre
SDS-TL	Short Data Service Transport Layer
SGSN	Serving GPRS Support Node
SIM	Subscriber Identity Module
SIP	Session Initiation Protocol
SLA	Service Level Agreement
SM	Session Management
SMS	Short Message Service
SMS-MO	SMS Mobile Originating
SMS-MT	SMS Mobile Termination
SMSC	Short Message Service Centre
SMTP	Simple Mail Transfer Protocol
SNDCP	Subnetwork Dependent Convergence Protocol
SpQ	Speech Quality
SQL	Structured Query Language
SRTP	Secure Real time Transfer Protocol
SSID	Service Set Identifier
STUN	Session Traversal Utilities for NAT
SwMI	Switching and Management Infrastructure
SYN	TCP Synchronize flag
TBCP	Talk Burst Control Protocol
TBF	Temporary Block Flow
TCP	Transmission Control Protocol
TCP/IP	Transmission Control Protocol/Internet Protocol
TCP-HS	Transmission Control Protocol Handshake
TE	Terminal Equipment
TETRA	Terrestrial Trunked Radio
TS	Timeslot
TX	Transmission
UDP	User Datagram Protocol

UE	User Equipment
UL	Uplink
ULDT	Uplink Direct Transfer
UM	Unacknowledged Mode
UMTS	Universal Mobile Telecommunications System
UNI	User Network Interface
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
USL	Upper Specification Level
UTC	Coordinated Universal Time
VLR	Visitor Location Register
VT	Video Telephony
WAE	Wireless Application Environment
WAP	Wireless Application Protocol
WCDMA	Wideband Code Division Multiple Access
WGR	WAP Get Request
WGS-84	World Geodetic System 1984
WSL	Wireless Session Layer
WSP	Wireless Session Protocol
WTLS	Wireless Transport Layer Security
WTP	Wireless Transport Protocol
XHTML	extensible Hypertext Markup Language
XML	Extensible Markup Language

5 Conventions

For the purposes of this Recommendation, the following symbols apply:

$E(x) = \mu$	Expected value of random variable x
$Var(x)=\sigma^2$	Variance of random variable x
σ	Standard deviation of random variable x
$f(x)$	Probability density function (PDF) of random variable x
$F(x)$	Cumulative distribution function (CDF) of random variable x
$S, x\in S$	Set of discrete values or interval of values the random variable x may take
IR	Set of real numbers
s, s^2	Empirical standard deviation/variance, analogous to σ and σ^2 (theoretical)
q_α	α-Quantile
u_α	α-Quantile of standard normal distribution
$x_{(i)}, x_{(1)}, x_{(n)}$	i-th ordered value, minimum and maximum of a given data set x_i, i =1,...,n
{X|Y}	Denotes one of X or Y

{X|Y|Z} Denotes one of X, Y or Z

6 Assessment of quality of service

6.1 QoS background

Recommendation [ITU-T E.800] provides the basic definition of QoS and highlights operational aspects of providing networks and services. In doing so, [ITU-T E.800] already gives a QoS definition and a framework for QoS implementation.

The definition given in [ITU-T E.800] is as follows:

– *Totality of characteristics of a telecommunications service that bear on its ability to satisfy stated and implied needs of the user of the service*.

Thus, in general QoS is focused on the service from the user's viewpoint being a complete end-to-end view. However, since the QoS consists of the collective effect of numerous single performances, any QoS analysis will have to deal also with sub-parts, e.g., network and terminal performance that can be analysed separately and independently from another. Therefore, there are a lot of standards and concepts dealing with QoS that are focusing on specific details and aspects of QoS.

The perceived quality of a service is the result of the combined influence of the performance of networks and terminals as well as the perception and expectation of the user. Thus QoS should also take into account both the user's and the service provider's point of view; it should always be judged from these different perspectives. There is an interrelation between user's requirements and its perception of the delivered quality on the one hand and the service/QoS planned and achieved by the service provider on the other hand.

A comprehensive view on QoS should take into account all aspects and perspectives including the numerous standards dealing with specific sub-parts of QoS. In the following clauses basic issues that need to be considered are discussed in more detail.

6.1.1 End-to-end QoS

As already indicated, QoS covers the whole end-to-end view of a telecommunications service and can be subdivided in separate parts that all have an influence on the resulting QoS. The degree of QoS depends on the collective effect of all sub-parts. This is illustrated in Figure 6-1 which is based on [ITU-T E.800].

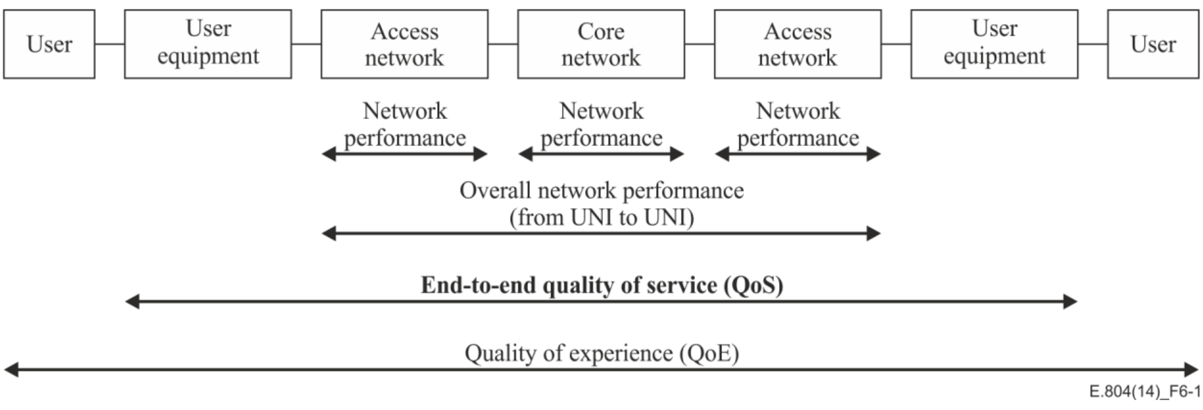


Figure 6-1 – End-to-end QoS

Quality measures in telecommunications can be determined in a hierarchical manner:

- **Network performance (NP)**: The network performance is assessed across a part of a network or a sub-network. Mostly, the NP is given in a technical way by assessing technical parameters which describe the performance of this part of the network in the desired way.

Examples are parameters like bit error ratio, sending and receiving power, transmission delay.

- **Overall NP**: If several network sections should be considered as being one integral part of the network ("black box"), the overall network performance has to be assessed. For example, the network performance of the complete network transmission between the two user network interfaces (UNIs) can be summarized in this way.
- **End-to-end quality of service (QoS)**: The assessment of the overall transmission chain from a user's perspective is considered to deliver the QoS in an objective manner. This implies that the most complete transmission chain without involving the user should be taken into account. Mostly, the generated measures rely on service related characteristics without knowing any details about the underlying network sections which are required to have an end-to-end service at all.
- **Quality of experience (QoE)**: The inclusion of the user to the overall quality in telecommunications extends the rather objective QoS to the highly subjective quality of experience (QoE). The QoE differs from user to user since it is influenced by personal experiences and expectations of the individual user.

6.1.2 Relationship between QoS and performance

It is important to understand that QoS differs from the network and terminal performance. QoS is the outcome of the user's experience/perception, while the network and terminal performance is determined by the performances of network elements one-by-one, or by the performance of the network as a whole including the performance of the attached terminals, i.e., the combination of the performance of all single elements. This means that the network performance may be used with an end-to-end meaning, but it may also be used to describe the performance of a network section.

Example: Access performance is usually separated from the core network performance in the operations of a single IP network, while Internet performance often reflects the combined NP of several autonomous networks.

However, the network and terminal performance have an influence on the QoS; they represent a part of it. The combined effect of the performance of all elements determines the overall service performance. There are intrinsic relationships between QoS and performance parameters, the former having a direct or indirect, and sometimes even inverse, influence on the latter. Furthermore, some performance measures can have a direct QoS meaning, while some others have to be combined in order to have a QoS signification.

Performance parameters are used to measure objectively the performance of specific network and terminal elements that have an influence on the resulting end-to-end quality of a service. Performance is measured and expressed in performance parameters. The main difference between QoS and network performance is that QoS provides quality information on an end-to-end and service related basis, whereas network performance specifies the technical operativeness of network and terminal elements or of network sections.

Recommendation [ITU-T I.350] provides the following conceptual categorization of QoS and network performance (NP) metrics as follows:

Quality of service parameter	Network performance parameter
User oriented	Network provider oriented
Service related attributes	Network element and technology related attributes
Focus on user observable effects	Focus on planning development (design), operations and maintenance

Quality of service parameter	Network performance parameter
Observed at service access points for the users, independent of network process and events	Observed at network connection element boundaries, e.g., relating to protocol specific interface signals

6.1.3 Relationship between QoS and QoE

In addition to the term QoS, the term quality of experience (QoE) is often used in order to stress the purely subjective nature of quality assessments in telecommunications and its focus on the user's perspective of the overall value of the service provided.

The increased significance of the term QoE is related to the fact that in the past, the term QoS was used laxly and mostly for only technical concepts focused on networks and networks elements. The definition of QoS, however, does include the degree of satisfaction of a user with a service. Thus, non-technical aspects are included, like, e.g., the user's environment, expectations, the nature of the content and its importance. But most service providers did use the QoS only in relation to the actual user-service interaction in order to cross-check whether the user requirements have been met by the service implementation of a provider (as perceived by the user). So there was a strong focus on the actual network performance and its immediate influence on user perceivable aspects while additional subjective and not directly service related aspects were omitted.

QoE is defined in Amendment 2 to [ITU-T P.10] as the overall acceptability of an application or service, as perceived subjectively by the end user. It includes the complete end-to-end system effects (client, terminal, network, services infrastructure, etc.) and may be influenced by user expectations and context. Hence the QoE is measured subjectively by the end user and may differ from one user to the other. However, it is often estimated using objective measurements.

Contributing to the QoE are objective service performance measures such as information loss and delay. Those objective measures together with human components that may include emotions, linguistic background, attitude, motivation, etc., determine the overall acceptability of the service by the end user. Figure 6-2 shows factors contributing to QoE. These factors are organized as those related to QoS and those that can be classified as human components.

QoE for video is often measured via carefully controlled subjective tests ([ITU-R BT.500-13] and [ITU-T P.800]) where video samples are played to viewers, who are asked to rate them on a scale. The rating assigned to each case are averaged together to yield the mean opinion score (MOS).

QoS is defined in [ITU-T E.800] as the collective effect of performance which determines the degree of satisfaction of a user of the service. In general, QoS is measured in an objective way.

In telecommunications, QoS is usually a measure of performance of services delivered by networks QoS mechanisms include any mechanism that contributes to improvement of the overall performance of the system and hence to improving the end-user experience. QoS mechanisms can be implemented at different levels.

Example: At the network level, QoS mechanisms include traffic management mechanisms such as buffering and scheduling employed to differentiate between traffic belonging to different applications. Other QoS mechanisms at levels other than the transport include loss concealment, application forward error correction (FEC), etc.

QoS parameters are used to describe the observed QoS. Similar to the QoS mechanisms, QoS parameters can be defined at different layers. Figure 6-2 gives an impression on factors that have an influence on the QoS and the QoE.

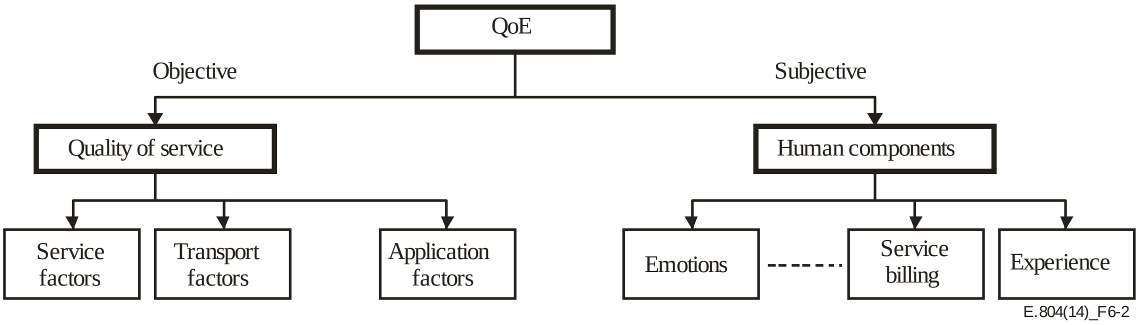


Figure 6-2 – QoE dimensions

In general, there is a correlation between the subjective QoE as measured by the MOS and various objective parameters of QoS.

Typically, there are multiple service level performance (QoS) metrics that impact overall QoE. The relation between QoE and service performance (QoS) metrics is typically derived empirically. Having identified the QoE/QoS relationship, it can be used in two ways:

Given a QoS measurement, one could predict the expected QoE for a user.

Given a target QoE for a user, one could deduce the net required service layer performance.

These prediction and deduction steps are built on assumptions and approximations. Due to the complexity of services and the many factors which have an influence on QoS/QoE, there is not a close one-to-one relationship which would allow statements like "If the bandwidth is increased by 200 kbit/s, the rating by the user will rise 0.5 points".

To ensure that the appropriate service quality is delivered, QoE targets should be established for each service and be included early on in system design and engineering processes where they are translated into objective service level performance metrics.

QoE will be an important factor in the marketplace success of triple-play services and is expected to be a key differentiator with respect to competing service offerings. Subscribers to network services do not care how service quality is achieved. What matters to them is how well a service meets their expectations for effectiveness, operability, availability, and ease of use.

6.1.4 QoS models in standardization documents

The relationship between user satisfaction, QoS and network performance is shown in Figure 6-3. This Recommendation has its focus on the technical aspects related to user satisfaction.

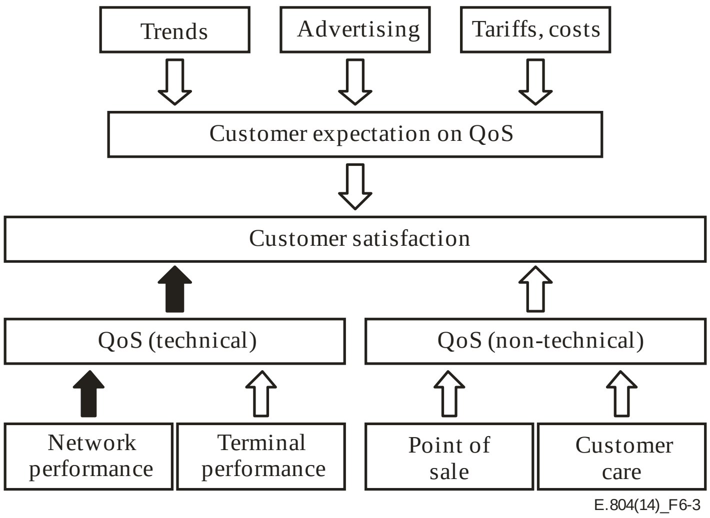


Figure 6-3 – Relationship between user satisfaction, QoS and network performance

6.1.4.1 Model of Recommendation ITU-T G.1000

In [ITU-T G.1000], a four-viewpoint model of QoS has been defined.

In general, the model discusses two dimensions:

- The relationship between a service user, the "user", and its service provider; and
- The QoS expectations and the achieved level of QoS for both parties; the user and the service provider.

Figure 6-4 describes the different QoS relations inspired by Figure 2 of [ITU-T G.1000].

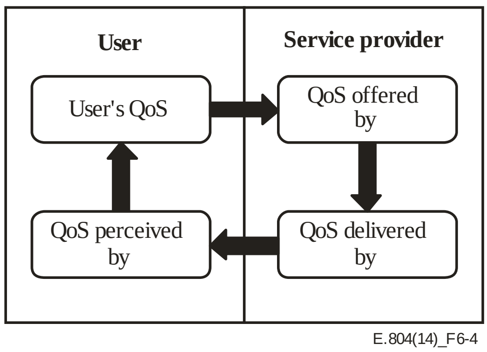


Figure 6-4 – Four-viewpoint model of QoS

The expected and perceived QoS from the user's perspective might be given in more descriptive terms whereas the provider will use more technically oriented terms to handle the offered and achieved level of QoS.

6.1.4.1.1 QoS required by the user (QoSR)

The QoS requirements of the user are described in a common, non-technical language. To describe its needs, the user reflects its expectations from an end-user's perspective. This means he formulates the requirements expected from the services delivered over the network. The user needs not to be aware of the technical feasibilities or implementation limitations that may occur.

Depending on the boundary conditions, the required QoS might also be part of the contractual terms.

The user's QoS requirements are the basis for the QoS level which should be offered by the service provider. The service provider should take the given requirements to deliver a QoS level which is matching the user's needs.

6.1.4.1.2 QoS offered by the service provider (QoSO)

The service provider states the desired QoS level. This can be done in two ways:

- In a non-technical manner to ease the comprehensibility of the given information towards the user.
- In a technical manner to allow an assessment by qualified experts, to allow to set up service level agreements (SLA) or to ease technical planning purposes.

For technical purposes, the QoS level is defined by the use of parameter definitions and corresponding values which should be reached. This information should be given separately for each kind of offered service.

6.1.4.1.3 QoS delivered by the service provider (QoSD)

The QoS delivered by the service provider reflects the currently achieved state of the QoS. This QoS level again should be described by parameters with assigned values, e.g., from active or passive probing or other kinds of appropriate testing.

The comparison of the offered and the delivered QoS allows assessing the capabilities of the service provider to deliver the promised QoS. Deviations can be made transparent very easily.

6.1.4.1.4 QoS experienced by the user (QoSE)

The experienced or perceived QoS reflects the subjective view of the user on its individual situation. The user's satisfaction is one of the main drivers of this kind of QoS.

In general, the perceived QoS is described in a non-technical manner. Service providers can retrieve the level of perceived QoS by executing surveys with their users or by asking their users for other kinds of feedback.

At this stage, the user combines the personal and individual experience with the more technically oriented quality of the delivered services. Overall, its individual QoE measure is generated. In technical means, QoE and QoS can be mapped onto each other.

Besides the technology based factors, further factors have an influence on the QoE of a user. Starting with the signing of the contract, going on with handling of problems by the provider, willingness to fulfil the user's needs and other things up to the cessation of the contract, the complete relationship between provider and user might have an influence on the QoE.

Obviously, the relationship experience has an influence on rating issues and makes the mapping of QoS and QoE more complicated because of "hidden factors".

6.1.4.2 Model of Recommendation ITU-T E.800

Recommendation [ITU-T E.800] illustrates the relationship between QoS and network performance by discussing the overall transmission chain which is used to deliver services to the user.

Figure 6-5 shows a scenario where two mobile users are communicating with each other. In general, the same situation applies also to any other constellation where a user deploys a client-server-like service.

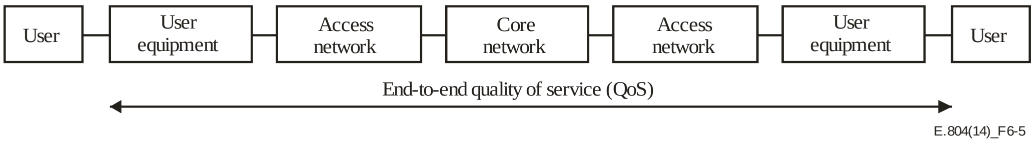


Figure 6-5 – Transmission chain discussed in [ITU-T E.800]

In the depicted case, there are two user equipments and two access networks, one on each end. The core network provides the link between the two access networks. It may consist of different networks run by different providers.

Each of the mentioned components has an influence on the achievable QoS level:

- If one of the user equipments has limited capabilities, e.g., reduced computational power, this will have an observable effect on the end-to-end QoS.
- The same applies to the access networks, where, e.g., the bandwidth of the transmission link has a major effect.
- Furthermore, if one of the providers linked within the core network violates SLAs between the providers, the end user may realise this by the perceived QoS.

Figure 6-6 provides a more abstract definition of QoS.

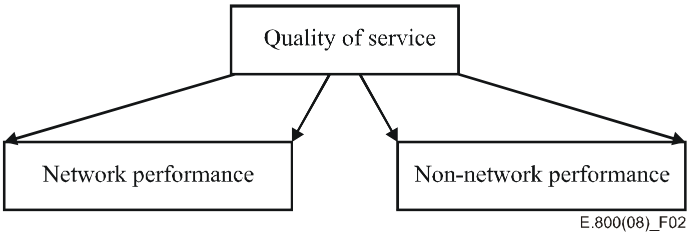


Figure 6-6 – Building blocks for QoS

The QoS perceived by a user is on the one hand influenced by technical terms like accessibility of a service or the set-up delay for dialup connections. On the other hand, factors like tariffs, repair times, hotline reachability and many others build the non-network performance.

Both components are integral parts of the end-to-end QoS perceived by the user.

6.1.4.3 Phase oriented aspect model

The usage of a service can also be separated in different consecutive phases. Here, the order in time is the criterion which differentiates the states of the service usage.

Figure 6-7 shows different phases of the network access, the service access and the service usage and the corresponding QoS aspects.

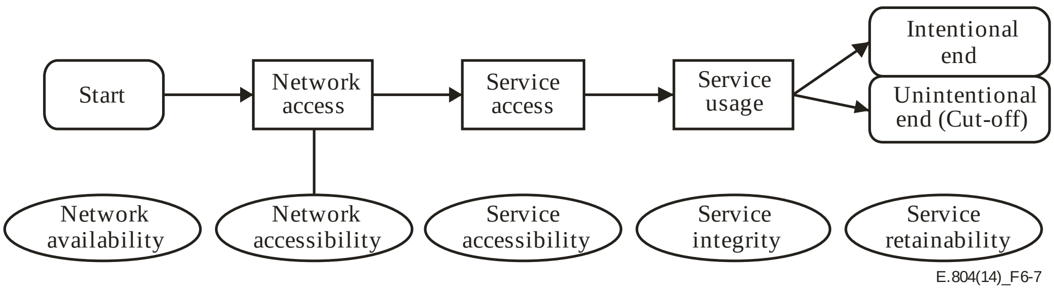


Figure 6-7 – QoS aspects related to different phases of service usage

The meaning of these phase related QoS aspects (Figure 6-7) is:

1) **Network availability**: Probability that the services are offered to a user via a network infrastructure.

2) **Network accessibility**: Probability that the user performs a successful registration on the network which delivers the service. The network can only be accessed if it is available to the user.

3) **Service accessibility**: Probability that the user can access the desired. A given network accessibility is a precondition for this phase.

4) **Service integrity**: This describes the QoS during service use and contains elements like the quality of the transmitted content, e.g., speech quality, video quality or number of bit errors in a transmitted file. The service integrity can only be determined if the service has been accessed successfully.

5) **Service retainability**: Service retainability describes the termination of services (in accordance with or against the will of the user). Examples for this are all kinds of cut-off parameters, e.g., the call cut-off ratio or the data cut-off ratio. Again, a previously performed successful service access is a precondition for this phase.

It is important to understand the interaction between these phases. As mentioned above, the phases are dependent on each other. Only if the previous phase has been passed successfully, the parameters of the consecutive phase can be determined.

Success of a phase is defined by successful execution of an attempt and in-time reaction of the network.

In most cases, this is the occurrence of a certain event (referenced as the "stop trigger") within a predefined time period (referenced as "timeout period").

The phase is not concluded successfully if at least one of the mentioned components is missing:

– If another than the specified event occurs, e.g., an event representing an error constellation.

– If the specified event occurs, but after expiration of the timeout period.

– If no event occurs at all.

While some of the phases have a greater importance for mobile networks, the general concept can be applied to any kind of network delivering services to the user.

6.2 QoS assessment process

To get reliable, reproducible and plausible QoS results, it is recommended to follow a generic scheme. This scheme is known as a QoS assessment process.

Following this predefined scheme, all relevant clauses are covered to reach the mentioned aim.

6.2.1 Objective of a QoS assessment

The objective of the criteria list is to have an agreed set of QoS criteria. They should allow easier external and internal benchmarking.

The services chosen are considered to be of a high relevance to the user in a national and international market and are common for most of the network operators.

The selected criteria are considered:

• to have main influence on the user's satisfaction with regard to the service;

• to identify technical QoS aspects, which can be influenced by the performance of the network or the terminal;

• to be measurable by technical means;

• to be relevant for network operator's national and international benchmarking.

There is the need to specify independent QoS criteria for each service; whereas related preconditions may be necessary.

6.2.2 Boundary conditions for a QoS assessment

6.2.2.1 Setting the target of the assessment

This clause deals with areas which have to be considered when defining the target of an assessment.

- One-time snapshot
- Acceptance procedure
- Continuous monitoring
- Optimization cycle
- Benchmarking

6.2.2.2 Defining the boundary conditions

Furthermore, lots of different boundary conditions have been defined unambiguously to allow comparable, reliable and reproducible results. In general terms, the basic questions "What, when, which way, what not, where, who" have to be answered in advance to the execution of the assessment.

- Set of QoS parameters to be assessed.
- Excluded topics.
- Number of samples to be generated per QoS parameter to achieve a pre-defined uncertainty level (determines the test duration in the end).
- Definition of parameters like file sizes, timeout conditions, bearers to use.
- Foreseen timeframe: Overall timeframe, operating hours per day.
- Foreseen locations: Included areas, excluded areas.
- Modes of testing: Active/passive, intrusive/non-intrusive.
- Mode of automation: Manual testing, automated testing, autonomous testing.
- Mobility modes: Static, nomadic, drive testing.
- Used test platform: Host computer, mobile device, observation platform within the network.

6.2.2.3 Operational issues

In this clause, operational issues to execute the assessment are addressed. These considerations are still to be taken into account before the assessment starts.

Important questions to be answered are:

- How will the assessment be executed?
- Which personnel are involved? Which qualifications are required?
- Which manning is required? (e.g., one engineer, two campaign managers, four drive testers)
- Which substitutes are available?
- How is operational security assured?
- Which backup procedures are applied?

6.2.3 Execution of a QoS assessment

Practical steps from defining dedicated boundary conditions (place, time, duration, services, devices, network technologies, etc.) to handling of retrieved raw data before validation (backup, inspection of sampling during test, etc.) are considered at this stage:

- Is a validation step required? How will it be done? Which documentation is required?
- Which steps are to be done for data generation?
- How is data stored?
- Is there an online reporting?

- Is there an online monitoring?
- What is the protection of the data transmission chain (e.g., injection of structured query language (SQL) queries, cross side scripting, etc.)?

6.2.4 Validating and aggregating results of a QoS assessment

When the data collection phase of the assessment is finalized, the gathered data has to be checked in different ways before aggregations and further QoS parameter calculations are made:

- Checking the amount of generated data.
- Execution of plausibility checks.
- Removal of erroneous samples (e.g., due to system problems) by setting corresponding markers.
- Generated data should be marked as invalid instead of being deleted.
- Aggregation of data according to areas of interest (time, location, provider, QoS parameter).
- Execution of plausibility checks on aggregated data to detect possible systematic errors (e.g., deviation of the mean value of a QoS parameter while the observed error rate is in the normal range).
- Calculation of minimum and maximum values.
- Quantile values, typically 5% and 95% quantiles which give some outlier related information.
- Compressed footprints of collected data by giving a set of quantile values, e.g., 5%, 10%, 50% (median), 90% and 95% quantiles, extended by the mean value.

6.2.5 Reporting results of a QoS assessment

After the evaluation steps are done in detail, a set of QoS parameters is available. These QoS parameters are characteristic for the dedicated assessment and represent the outcome of the overall activity. They should be reported in an easy understandable format and should point out the key findings for each relevant constellation, e.g., for each question and also for each provider.

Lots of different schemes can be applied to visualise the determined QoS parameter sets. Since in many cases the readers of such reports are used to have a certain representation, changes in the reporting format should be applied carefully.

Finally, the report including some analysis and visual representations should be distributed to the stakeholders.

6.2.6 Matching QoS results with targets

Based on the retrieved QoS results, the current state of QoS parameters should be matched against the corresponding predefined target values. By doing this matching, deviations from the desired QoS state becomes obvious. Possible results might be:

- The current QoS situation is better than expected. QoS targets are reached completely. In this situation, one can think of defining tight targets or of relaxing QoS requirements to save operational costs. This is the most convenient situation.
- The current QoS situation matches exactly the target level. In this case, the QoS process delivers a stable outcome. For the future, the trend of the achieved QoS level has to be observed to assure that QoS targets are reached in long-term.
- The current QoS situation does not reach the expected QoS level. In this case, some actions should be taken to improve the QoS level. It is not recommended to just relax the required QoS level to reach the QoS targets again!

6.2.7 Optimization of QoS matters

If a negative deviation of the QoS level has been stated, some correcting actions should take place. At the end, the desired QoS level should be reached again.

Optimization efforts are always a matter of technical and economical feasibility. This means there must be a chance to change the technical implementation of a service and there must be available financial and personnel resources to implement the foreseen changes. If one of these factors fails, an effective optimization and therefore improvement of the QoS level is not achievable.

6.3 Basic settings for QoS assessments

This clause defines generic statements which should ease the readers' efforts to implement and execute standardized QoS assessments. The main aim is to assure reproducible, reliable and comparable results of QoS assessments.

6.3.1 Location where the measurement is actually performed

6.3.1.1 Concept of PCOs

The point of control and observation (from now on called "point of observation" or PCO) is the location where the measurement is actually performed. The location can be either inside the network or at the end-point. The measurements should be done using standardized interfaces and protocols.

Possible points of observation for QoS parameters covered in this Recommendation are:

- Inside nodes in the network (e.g., radio network controller (RNC), base station and switch)
- Observations in the terminal:
  - End-point test tool; or
  - Measurements that are reported back from the terminal to the network

Figure 6-8 provides visualization of different PCOs.

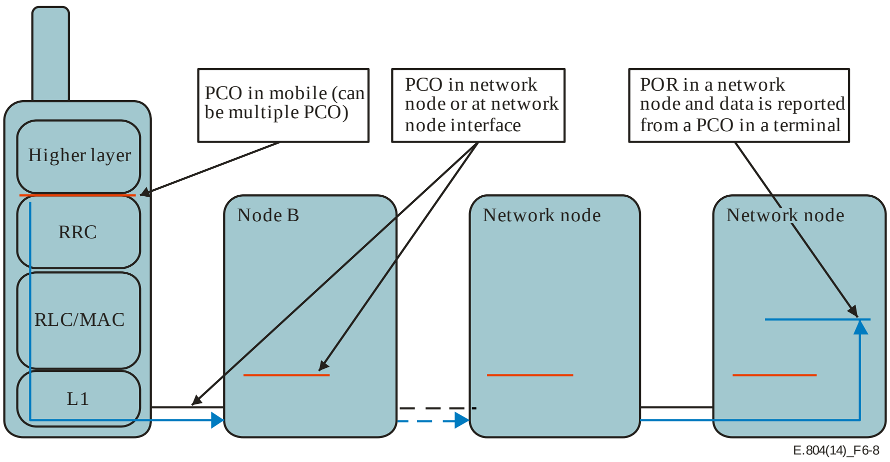


Figure 6-8 – Visualization of different PCOs

6.3.1.2 Point of recording (POR)

The point of recording (POR) is where the QoS parameters are recorded. The POR can be the same as the PCO or another point inside the terminal or the network. If the PCO and the POR are not the same, the measurement data must be reported from the PCO to the POR. Examples of such reporting are described in Appendix XI.

6.3.2 Usage of standardized units for data

In this Recommendation, the following definitions related to the amount of data have to be applied:

- 1 kbyte is defined as the amount of 1024 bytes.
- 1 Mbyte is defined as the amount of 1024 kbytes which is equivalent to 1048576 bytes.

This Recommendation orientates itself on units like kbyte or Mbyte to describe the amount of data, e.g., in a storage or during a data transfer.

Therefore, units like MebiBytes (MiB) which are officially standardized in IEC/IEEE will not be applied here.

6.3.3 Influence of timeout values on failure ratios

For many QoS parameters, especially those from the service phase "Accessibility", there are pairs of QoS parameters available. Mostly, there are two QoS parameters defined:

- one QoS parameter describes the success or non-success of a set-up trial (ratio QoS parameter); and
- a second QoS parameter describes the set-up delay for those attempts where the set-up has been successful (delay QoS parameter).

Following the principles of this Recommendation, both QoS parameters should be based on the same trigger points wherever possible.

However, both QoS parameters are not decoupled in general. The link between them is the timeout value which is always required to determine the delay QoS parameter in a defined manner. The timeout value assures that the system under test waits only for a predefined period of time before the attempt is terminated.

By choosing long intervals for the timeout period, two effects occur:

- The system waits for a long period of time before closing down the attempts. This will reduce the number of samples per hour for example.
- By waiting for a long period of time, the probability is high that the successful event (here the successful connection establishment event) will be observed. This will increase the first QoS parameter describing the set-up success ratio.

On the other hand, short intervals for the timeout period have these effects on the QoS parameters:

- The system waits for a short period of time before closing down the attempts. This will increase the number of samples per hour for example.
- By waiting for a short period of time, the probability is high that the timeout duration is reached often. Once the timeout condition is reached, the ratio QoS parameter counts an unsuccessful attempt. The corresponding delay QoS parameter cannot be determined in this case.

Comparing both scenarios, the overall influence of the choice of the timeout value for the outcome of both QoS parameter values should be obvious.

In simple words: the timeout value influences heavily the sample number per hour as well as the observed set-up success ratio. The shorter the timeout value is chosen, the lower the set-up success ratio will be, and the less delay QoS parameter values will be available.

6.4 Service independent QoS criteria

This clause gives some hints on how some of the very basic service independent terms are used in this Recommendation.

6.4.1 Unavailability

Unavailability describes the probability that a network or service is not offered to a user.

6.4.2 Non-accessibility

Non-accessibility parameters handle the probability that the user cannot perform a successful access to a network or service when the user intends to use the network or service.

6.4.3 Time parameters

Different time parameters occur in QoS measurements. Typical representatives are time parameters like access time, activation time or set-up time.

6.4.4 Transfer time

The transfer time is a basic parameter to calculate data rates, meaning to divide the amount of transmitted data by the time period which has been required to transfer this amount of data.

6.4.5 Content integrity

Content integrity parameters describe the quality of the transferred data. Typical representatives are parameters like "speech quality", "video quality" or "data integrity". Related to video quality, appropriate guidelines to determine the quality of video content in the right manner can be found in [ETSI TR 102 493].

6.5 Service dependent QoS criteria

This clause provides insight on how some of the very basic service dependent terms are used in this Recommendation.

6.5.1 Rate parameters

After a data link has been successfully established, these parameters provide the average data transfer rate measured throughout the entire connection time to the service. The data transfer shall be successfully terminated. The prerequisite for these parameters is the network and the service access.

Rate parameters are always time dependent. For example, a data rate expresses the relation of an amount of data which is transferred within a specific period of time.

The unit of rate parameters always carries a time unit in the denominator, like in kbit/s.

6.5.2 Ratio parameters

Ratio parameters reflect the relation between a subset and a basic set. For example, an error ratio reflects the number of errors which could be observed in relation to all observed attempts, tried or executed.

In general, dedicated preconditions have to be fulfilled before an attempt can be added to the basic set of attempts. These preconditions depend on the service that the ratio parameter should be determined for.

The unit of ratio parameters is always a percentage, identified by the % (per cent) character. A typical representative of a ratio parameter is a success ratio, which might reach any value between 0 per cent (meaning all attempts failed) and 100 per cent (meaning all attempts were executed successfully).

6.5.3 Service non-accessibility

The service non-accessibility ratio denotes the probability that a subscriber cannot access the desired service successfully. This includes all necessary steps like a required connection establishment.

In general, dedicated preconditions have to be fulfilled before this parameter can be assessed. E.g., for voice telephony, the network must be available, the terminal must be attached to the network and the addressed B-party should not be busy.

6.5.4 Set-up time

The set-up time is the time period needed to access the service successfully.

Example: Related to voice telephony, the set-up time is the period of time from starting the dial-up connection to the point of time when the content is sent or received.

This parameter is not calculated unless the corresponding setup attempt is successful.

6.5.5 Failure ratio

A failure ratio is the probability that a user experiences a malfunction of a certain requested transaction. Typical examples are set-up failure ratios.

6.5.6 Cut-off ratio

A cut-off ratio is the probability that a user experiences an unintended termination of a transaction. Typical examples are voice cut-off ratio or streaming cut-off ratio. The precondition for the assessment of this parameter is the successful establishment attempt, e.g., for a voice call or a data session.

6.5.7 End-to-end failure ratio

The term end-to-end failure ratio is the probability that a transaction with end-to-end meaning fails. End-to-end meaning represents the fact that the whole transmission chain from the data source to the data sink is included in this consideration.

Typically, failures in a part of the end-to-end transmission chain will also be reflected in the end-to-end failure ratio.

6.5.8 End-to-end delivery time

The end-to-end delivery time represents the delay that occurs when content is transferred across the complete transmission chain from the data source to the data sink.

There is a need to differentiate between a unidirectional transmission and a bidirectional transmission where the transmission chain is passed twice.

7 Definition of quality of service parameters and their computation

7.1 QoS parameter basics

7.1.1 General overview

Figure 7-1 shows a model for quality of service parameters. This model has four layers.

The first layer is the "network availability", which defines QoS rather from the viewpoint of the service provider than the service user. The second layer is the "network access". From the service user's point of view this is the basic requirement for all the other QoS aspects and parameters. The third layer contains the other three QoS aspects "service access", "service integrity" and "service retainability". The different services are located in the fourth layer. Their outcomes are the QoS parameters.

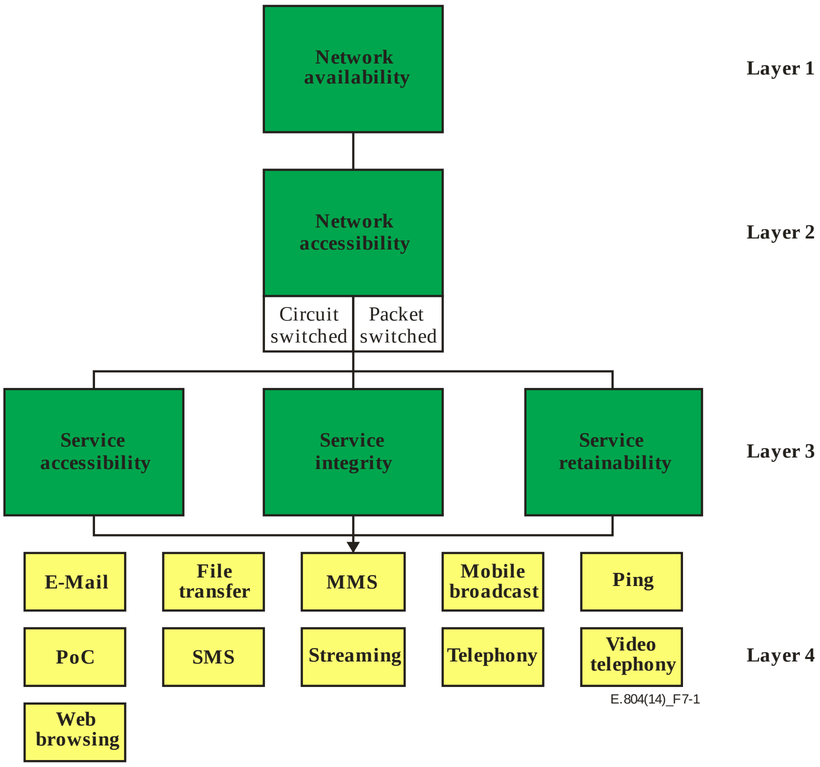


Figure 7-1 – QoS aspects and the corresponding QoS parameters

7.1.2 FTP, HTTP and e-mail issues

Currently two main views about the best way to reflect the user's experience for these services are in place:

- One preferring the payload throughput philosophy and the other preferring the transaction throughput philosophy:
  - – Method A defines trigger points which are as independent as possible from the service used, therefore representing a more generic view (payload throughput).
  - – Method B defines trigger points on application layer, therefore representing a more service oriented view (transaction throughput).

An example of the different trigger points defined for each set is illustrated in Figures 7-2 and 7-3. The start trigger point for the "mean data rate" for web browsing is either the reception of the first packet containing data content (Method A) or the sending of the HTTP GET command (Method B).

A field test system compliant to this Recommendation shall measure both sets (Method A and B) of QoS indicators using commercial user equipments (UEs).

In addition, a set of technical QoS indicators is defined that covers the attach and packet data protocol (PDP) context activation procedure. Field test systems shall be able to measure these QoS indicators.

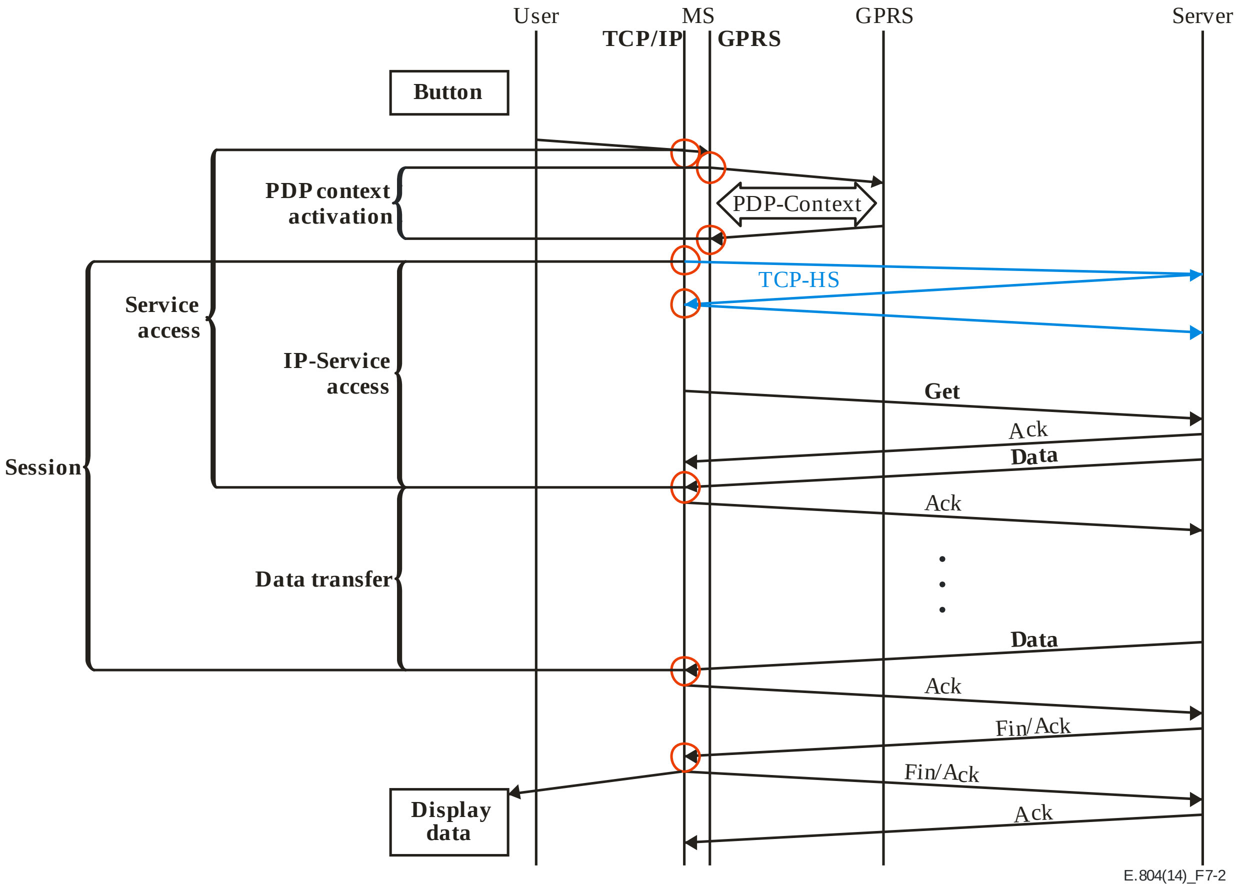


Figure 7-2 – QoS parameters version A (example: HTTP via GPRS)

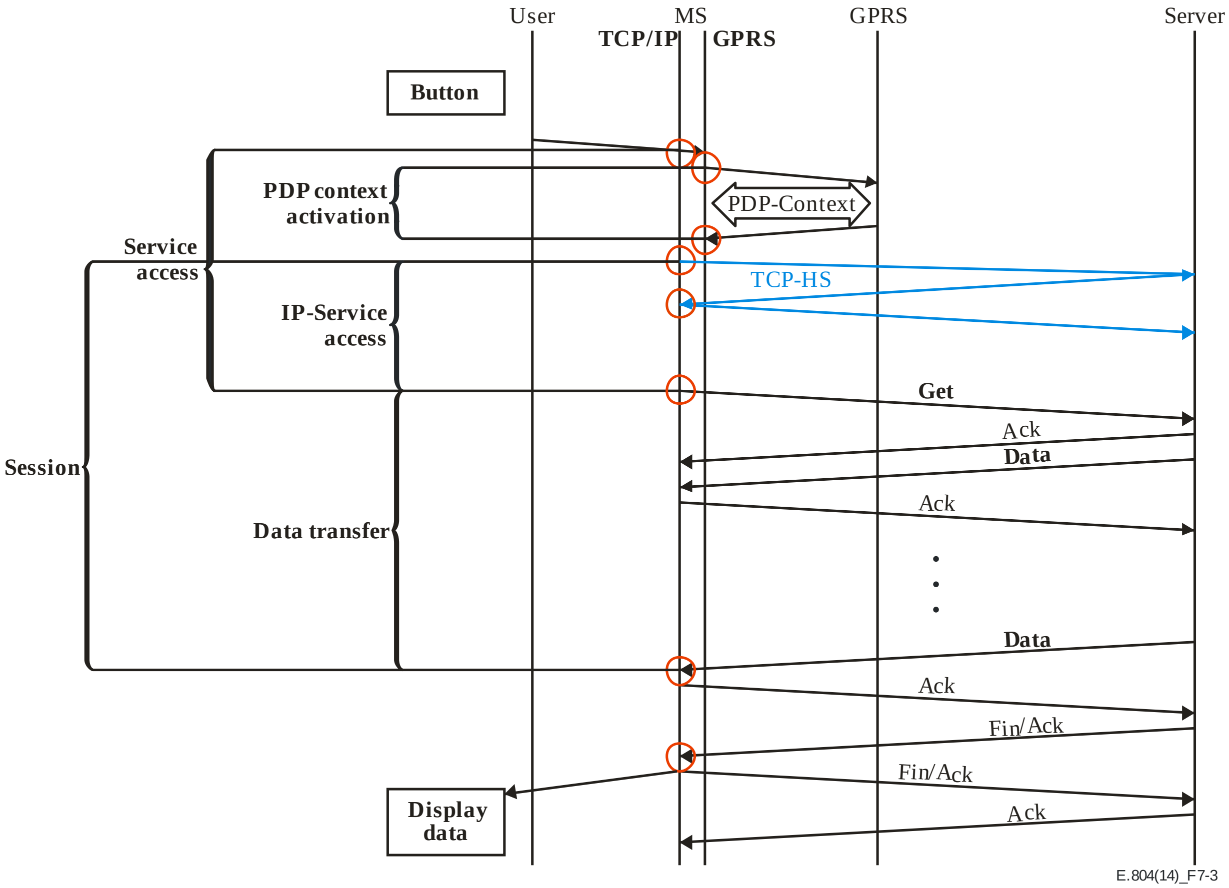


Figure 7-3 – QoS parameters version B (example: HTTP via GPRS)

7.1.2.1 Performance enhancement proxies

Performance enhancement proxies (PEPs, also called accelerators) are network elements employed to improve the performance of the data services offered by the mobile operator. To achieve this goal, such proxies typically employ different techniques:

- Content filtering (elimination of content of a certain type, e.g., audio files).
- Lossless content compression (e.g., compression of HTML or other text like files).
- Lossy content compression (e.g., recalculation of JPG files to a lower colour deepness or resolution of detail richness).
- Protocol optimization (e.g., for HTTP, POP3).

By these means PEPs achieve a reduction of the amount of data transferred from or to the end user and thus a reduction of the transfer time. Some of these techniques will have an impact on content integrity and/or on the content quality as perceived by the end user.

The following guidelines apply whenever performance enhancement proxies are employed:

- When reporting mean data rates, it shall be observed that the actual amount of transferred user data (rather than the original amount of hosted data) is used for calculations.
- When reporting session times it is recommended that an indication of the impact of the enhancement techniques on the content quality is given – e.g., the content compression ratio (amount of received and uncompressed content compared to the amount of originally hosted content).
- It is recommended to indicate the impact of the enhancement techniques on the content integrity, e.g., eliminated or modified content.

7.1.3 Timeouts

For regular testing, it is necessary to define timeout values for specific service transactions as testing time is a limited resource. These timeouts have a direct impact on the respective QoS parameters. A small timeout value for instance will result in higher failure ratio parameters while a large timeout value will lead to lower throughput rates and higher transfer times, statistically.

In this Recommendation, an expired timeout means that the stop trigger point given in the definition of the QoS parameter definition was not reached.

In case no timeout is stated in a technical description/protocol part for an expected response, this shall be understood implicitly in the sense that the response needs to be received within a predefined time. Otherwise, it is regarded as not having been received at all.

7.1.4 Trigger points

In this Recommendation, trigger point definitions are part of each QoS parameter definition.

For each trigger point definition, information concerning the technical description/protocol part is given as part of the definition. In particular, each trigger point may contain more than one technical description/protocol part, reflecting for example different reference points and/or protocol layers.

For example, a trigger point may be defined both by 'AT commands and responses' at the AT interface and 'layer 3 messages'. In such cases and if not stated differently in the respective subsection defining the trigger point, these descriptions/protocol parts are equally valid.

Data from measurements of QoS parameters being based on different technical descriptions/protocol parts for the same trigger point shall not be compared directly.

In general, for the calculation of QoS parameters, it is recommended to use related trigger points in a corresponding way, i.e., utilising the same protocol layer and reference point for the start and stop trigger, respectively.

In the case where more than one technical description/protocol part is present, it is up to the user to choose the technical description/protocol part suiting best the actual needs and/or situation. For instance, one of the related reference points might not be accessible for measurements whereas some other reference point is.

7.2 Service independent QoS parameters

7.2.1 Radio network unavailability [%]

7.2.1.1 Abstract definition

The radio network unavailability is the probability that the mobile services are not offered to a user.

7.2.1.2 Abstract equation

$$\text{Radio network unavailability}\,[\%] = \frac{\text{probing attempts with mobile services not available}}{\text{all probing attempts}} \times 100$$

7.2.1.3 Trigger points

GSM:

Event from abstract equation	Trigger point from user's point of view	Technical condition
Probing attempt	Not applicable.	Check C1-criteria.
Mobile services available	Not applicable.	GSM: C1-criteria > 0. Any emergency camping on any other than the target networks is considered as no network.
Mobile services not available	Technical condition not met.	

GPRS:

Event from abstract equation	Trigger point from user's point of view	Technical condition
Probing attempt	Not applicable.	Check GRPS specific signalling contained within system information 3.
Mobile service available	Not applicable.	Specific signalling contained in system information 3 exists on cell selection.
Mobile service not available	Technical condition not met.	

UMTS (WCDMA):

Event from abstract equation	Trigger point from user's point of view	Technical condition
Probing attempt	Not applicable.	Check S-criteria.
Mobile services available	Not applicable.	WCDMA: S-criteria satisfied. Any emergency camping on any other than the target networks is considered as no network.
Mobile services not available	Technical condition not met.	

NOTE 1 – For information on how the C1-criteria is defined, refer to [ETSI TS 145 008].

NOTE 2 – For information on how the S-criteria is defined, refer to [ETSI TS 125 304].

NOTE 3 – When the test mobile operates in dual-mode (GSM/UMTS), then the judgement on radio network unavailability is made with respect to the radio access technology in which the test device is at the moment of checking.

NOTE 4 – The target networks could constitute more than one network, e.g., to cover national or international roaming.

7.2.2 Network non-accessibility [%]

This parameter was replaced by the "network selection and registration failure ratio" and "network selection and registration time" parameter specified in clauses 7.2.2.1 and 7.2.2.2.

7.2.2.1 {Manual | Automatic} network selection and registration failure ratio [%]

7.2.2.1.1 Abstract definition

The network selection and registration failure ratio is the probability that the user cannot perform a successful selection (i.e., manual or automatic) and registration on a defined and desired public land mobile network (PLMN) or any other PLMN.

Remarks:

- The user equipment (UE) shall be deregistered from any available PLMN and shall not be within a registration procedure.
- Some network (automatic selection mode) or the desired network (manual selection mode) to which the UE should register as well as the desired access technology shall be available and the UE shall be allowed to register to this network.
- The UE shall support the +COPS command set according to the definition in [ETSI TS 127 007]:
  - The optional <AcT> field of the +COPS set command as defined in [ETSI TS 127 007] shall be supported by the UE, if used in the respective +COPS set command.
- The execution of the +COPS set command shall not be aborted by the sending of any other commands by the terminal equipment (TE).
- The UE shall support the +CREG command set according to the definition in [ETSI TS 127 007]:
  - The network registration unsolicited result code shall be enabled in the UE.
- The UE shall support the +CGREG command set according to the definition in [ETSI TS 127 007]:
  - The GPRS network registration status unsolicited result code shall be enabled in the UE.

- The mobile termination (MT) shall be in full functionality state.

7.2.2.1.2 Abstract equation

$$\{\text{Manual}|\text{Automatic}\}\text{network selection and registration failure ratio }[\%] = \frac{\text{unsuccessful selection and registration attempts on PLMN}}{\text{all selection and registration attempts}} \times 100$$

7.2.2.1.3 Trigger points

Manual network selection and registration – circuit switched (CS) case:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Manual network selection and registration attempt	Start: User initiates manual network selection and registration.	Start: The set command "+COPS=1,<format>,<oper> [,<AcT>]" for the +COPS command is sent.
Successful manual network selection and registration	Stop: Operator logo appears in the display of the UE.	Stop: Reception of "OK" for the set command "+COPS=1,<format>,<oper>[,<AcT>]" and reception of the unsolicited result code for network registration status "+CREG" by TE with the value "1" or "5" for <stat> and reception of the value "1" for <mode> and the desired values for <oper>, and optionally <AcT>, for the read command "+COPS?".
Unsuccessful manual network selection and registration	Stop trigger point not reached.	

Automatic network selection and registration – CS case:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Automatic network selection and registration attempt	Start: User initiates automatic network selection and registration.	Start: The set command "+COPS=0,0" for the +COPS command is sent.
Successful automatic network selection and registration	Stop: Operator logo appears in the display of the UE.	Stop: Reception of "OK" for the set command "+COPS=0,0" and reception of the unsolicited result code for network registration status "+CREG" by the TE with the value "1" or "5" for <stat> and in addition and only in case a certain network operator is desired, reception of the value "0" for <mode> and the desired values for <oper>, and optionally <AcT>, for the read command "+COPS?".

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Unsuccessful automatic network selection and registration	Stop trigger point not reached.	

Manual network selection and registration – packet switched (PS) case:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Manual network selection and registration attempt	Start: User initiates manual network selection and registration.	Start: The set command "+COPS=1,<format>,<oper> [,<AcT>]" for the +COPS command is sent.
Successful manual network selection and registration	Stop: PS logo appears in the display of the UE.	Stop: Reception of "OK" for the set command "+COPS=1,<format>,<oper>[,<AcT>]" and reception of the unsolicited result code for GPRS network registration status "+CGREG" by TE with the value "1" or "5" for <stat> and reception of the value "1" for <mode> and the desired values for <oper>, and optionally <AcT>, for the read command "+COPS?".
Unsuccessful manual network selection and registration	Stop trigger point not reached.	

Automatic network selection and registration – PS case:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Automatic network selection and registration attempt	Start: User initiates automatic network selection and registration.	Start: The set command "+COPS=0,0" for the +COPS command is sent.
Successful automatic network selection and registration	Stop: PS logo appears in the display of the UE.	Stop: Reception of "OK" for the set command "+COPS=0,0" and reception of the unsolicited result code for GPRS network registration status "+CGREG" by the TE with the value "1" or "5" for <stat> and in addition and only in case a certain network operator is desired, reception of the value "0" for <mode> and the desired values for <oper>, and optionally <AcT>, for the read command "+COPS?".

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Unsuccessful automatic network selection and registration	Stop trigger point not reached.	

Some possible indicators for unsuccessful manual or automatic network selection and registration attempts are the following:

- In case verbose <err> values have been enabled according to [ETSI TS 127 007] via AT+CMEE=2: "+CMEERROR: <err>" is received for the +COPS set or read command.
- No answer is received for the +COPS set or read command within a pre-determined time.
- In case of manual network selection and registration:

  The desired value(s) from the +COPS set command for <oper>, and optionally for <AcT>, are not returned by the read command.
- No unsolicited result code for network registration status "+CREG" is received within a pre-determined time.
- The unsolicited result code for network registration status "+CREG" is not received by the TE with the desired value "1" or "5" for <stat> within a pre-determined time.
- No unsolicited result code for GPRS network registration status "+CGREG" is received within a pre-determined time.
- The unsolicited result code for GPRS network registration status "+CGREG" is not received by the TE with the desired value "1" or "5" for <stat> within a pre-determined time.

7.2.2.2 {Manual | Automatic} network selection and registration time [s]

7.2.2.2.1 Abstract definition

The network selection and registration time is the time it takes the user to perform a successful selection and registration (i.e., manual or automatic) on a defined and desired public land mobile network (PLMN) or any other PLMN.

Remarks:

- The user equipment (UE) shall be deregistered from any available PLMN and shall not be within a registration procedure.
- Some network (automatic selection mode) or the desired network (manual selection mode) to which the UE should register as well as the desired access technology shall be available and the UE shall be allowed to register to this network.
- The UE shall support the +COPS command set according to the definition in [ETSI TS 127 007].
- The optional <AcT> field of the +COPS set command as defined in [ETSI TS 127 007] shall be supported by the UE, if used in the respective +COPS set command.
- The execution of the +COPS set command shall not be aborted by the sending of any other commands by the terminal equipment (TE).
- The UE shall support the +CREG command set according to the definition in [ETSI TS 127 007].
- The network registration unsolicited result code shall be enabled in the UE.
- The UE shall support the +CGREG command set according to the definition in [ETSI TS 127 007].
- The GPRS network registration status unsolicited result code shall be enabled in the UE.

• The MT shall be in full functionality state.

7.2.2.2.2 Abstract equation

$$\text{Network selection and registration time}[s] = \left(t_{\text{start of network selection and registration attempt}} - t_{\text{successful network selection and registration}}\right)[s]$$

7.2.2.2.3 Trigger points

Manual network selection and registration – CS case:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Start of network selection and registration attempt	Start: User initiates manual network selection and registration.	Start: The set command "+COPS=1,<format>,<oper> [,<AcT>]" for the +COPS command is sent.
Successful network selection and registration	Stop: Operator logo appears in the display of the UE.	Stop: The point in time where the unsolicited result code for network registration status "+CREG" is received by TE with the value "1" or "5" for <stat> in case of the reception of "OK" for the set command "+COPS=1,<format>,<oper>[,<AcT>]" and reception of the value "0" for <mode> and the desired values for <oper>, and optionally <AcT>, for the read command "+COPS?".

Automatic network selection and registration – CS case:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Start of network selection and registration attempt	Start: User initiates automatic network selection and registration.	Start: The set command "+COPS=0,0" for the +COPS command is sent.
Successful network selection and registration	Stop: Operator logo appears in the display of the UE.	Stop: The point in time where the unsolicited result code for network registration status "+CREG" is received by TE with the value "1" or "5" for <stat> in case of the reception of "OK" for the set command "+COPS=0,0" and in addition and only in case a certain network operator is desired, reception of the value "0" for <mode> and the desired values for <oper>, and optionally <AcT>, for the read command "+COPS?".

Manual network selection and registration – PS case:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Start of network selection and registration attempt	Start: User initiates manual network selection and registration.	Start: The set command "+COPS=1,<format>,<oper> [,<AcT>]" for the +COPS command is sent..
Successful network selection and registration	Stop: PS logo appears in the display of the UE.	Stop: The point in time where the unsolicited result code for GPRS network registration status "+CGREG" is received by TE with the value "1" or "5" for <stat> in case of the reception of "OK" for the set command "+COPS=1,<format>,<oper>[,<AcT>]" and reception of the value "0" for <mode> and the desired values for <oper>, and optionally <AcT>, for the read command "+COPS?".

Automatic network selection and registration – PS case:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Start of network selection and registration attempt	Start: User initiates automatic network selection and registration.	Start: The set command "+COPS=0,0" for the +COPS command is sent.
Successful network selection and registration	Stop: PS logo appears in the display of the UE.	Stop: The point in time where the unsolicited result code for GPRS network registration status "+CGREG" is received by TE with the value "1" or "5" for <stat> in case of the reception of "OK" for the set command "+COPS=0,0" and in addition and only in case a certain network operator is desired, reception of the value "0" for <mode> and the desired values for <oper>, and optionally <AcT>, for the read command "+COPS?".

7.2.3 Attach failure ratio [%]

7.2.3.1 Abstract definition

The attach failure ratio is the probability that a subscriber cannot attach to the PS network.

7.2.3.2 Abstract equation

$$\text{Attach failure ratio}\,[\%] = \frac{\text{unsuccessful attach attempts}}{\text{all attach attempts}} \times 100$$

7.2.3.3 Trigger points

GPRS/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Attach attempt	Start: User turns the UE on.	Start: Layer 3 (GMM): The "ATTACH REQUEST" message is sent by the UE. AT: "AT+CGATT=1" is sent by the TE.
Successful attach attempt	Stop: Attach logo appears in the display of the UE.	Stop: Layer 3 (GMM): The "ATTACH ACCEPT" message is received by the UE. AT: "OK" is received by the TE.
Unsuccessful attach attempt	Stop trigger point not reached.	

Remarks:

- GPRS: Indicator will only be updated by event (a loss of SI13 signalling or a coverage hole will not be detected if no attach, routing area update or temporary block flow (TBF) request is initiated).
- It might occur that the UE sends more than one attach request towards the serving GPRS support node (SGSN), since retries are necessary. A maximum of four retries are possible.

  These retries should not have impact on the attach failure ratio, since only one attach request message should be counted in the calculation.
- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1).
- The mobile shall be in detached state. "AT+CGATT?" may be used to check the attach state.

7.2.4 Attach set-up time [s]

7.2.4.1 Abstract definition

The attach set-up time is the time period needed to attach to the PS network.

7.2.4.2 Abstract equation

$$\text{Attach set-up time}\,[s] = \left(t_{\text{attach complete}} - t_{\text{attach request}}\right)\,[s]$$

7.2.4.3 Trigger points

GPRS/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{attach request}}$: Time of attach request	Start: User turns the UE on.	Start: Layer 3 (GMM): The "ATTACH REQUEST" message is sent by the UE. AT: "AT+CGATT=1" is sent by the TE.
$t_{\text{attach complete}}$: Time when attach is complete	Stop: Attach logo appears in the display of the UE.	Stop: Layer 3 (GMM): The "ATTACH ACCEPT" message is received by the UE. AT: "OK" is received by the TE.

Remarks:

- The difference between an attach request of a known subscriber and an unknown subscriber will be reflected in the time period indicating the attach set-up time. In case of an unknown subscriber (meaning that the SGSN has changed since the last detach, or if it is the very first attach of the mobile to the network), the SGSN contacts the home location register (HLR) in order to receive the subscriber data. The attach set-up time of an unknown subscriber will be slightly longer than the one of a known subscriber.
- While determining the average attach set-up time only successful attach attempts are included in the calculations.
- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1).
- The mobile shall be in detached state. "AT+CGATT?" may be used to check the attach state.

7.2.5 PDP context activation failure ratio [%]

7.2.5.1 Abstract definition

The PDP context activation failure ratio is the probability that the PDP context cannot be activated. It is the proportion of unsuccessful PDP context activation attempts and the total number of PDP context activation attempts.

7.2.5.2 Abstract equation

$$\text{PDP context activation failure ratio}\,[\%] = \frac{\text{unsuccessful PDP context activation attempts}}{\text{all PDP context activation attempts}} \times 100$$

7.2.5.3 Trigger points

GPRS/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PDP context activation attempt	Start: User initiates the service access.	Start: Layer 3 (SM): The first "ACTIVATE PDP CONTEXT REQUEST" message is sent by the UE. AT: "AT+CGACT=1, 1" is sent by the TE.
PDP context activation attempt	Stop: PDP context logo appears in the display of the UE.	Stop: Layer 3 (GMM): The "ATTACH ACCEPT" message is received by the UE. AT: "OK" is received by the TE.
Unsuccessful attempt	Stop trigger point not reached.	

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PDP context activation attempt	Start: User initiates the service access.	Start: Layer 3 (SNDCP): The first "SN-ACTIVATE PDP CONTEXT DEMAND" message is sent by the UE. AT: "ATD*99#" is sent by the TE.
Successful PDP context activation attempt	Stop: PDP context logo appears in the display of the UE.	Stop: Layer 3 (SNDCP): The "SN-ACTIVATE PDP CONTEXT ACCEPT" message is received by the UE. AT: The "CONNECT" indication is received by the TE.
Unsuccessful attempt	Stop trigger point not reached.	

Remarks:

- In GPRS/UMTS the "AT+CGACT=1,1" shall be sent when the UE has no active context using the selected access point name (APN). "AT+CGACT?" may be used to check the state. The PDP context should be defined with the "AT+CGDCONT" command.
- In GPRS/UMTS it might occur that the UE sends more than one PDP context activation request towards the SGSN since retries are necessary. A maximum of four retries are possible. The timer T3380 expires after 30 seconds for each attempt, see [ETSI TS 124 008].
- In TETRA the "ATD*99#" shall be sent when the UE has no active context. The PDP context should be defined with the "AT+CTSDC" command.
- In TETRA it might occur that the UE sends more than one PDP context activation request towards the SwMI, since retries are necessary. A maximum of RETRY_ACTIVATION = 3 retries are possible.
- For GPRS/UMTS the PS bearer has to be active in the cell where the attempt is initiated (see clause 7.2.1) and the UE has to be attached (see clause 7.2.3).
- For TETRA the PS services shall be enabled at the cell where the attempt is initiated.

7.2.6 PDP context activation time [s]

7.2.6.1 Abstract definition

The PDP context activation time is the time period needed for activating the PDP context.

7.2.6.2 Abstract equation

$$\text{PDP context activation time}[s] = \left(t_{\text{PDP context activation accept}} - t_{\text{PDP context activation request}}\right)[s]$$

7.2.6.3 Trigger points

GPRS/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{PDP context activation request}}$: Time of PDP context activation request	Start: User initiates the service access.	Start: Layer 3 (SM): The first "ACTIVATE PDP CONTEXT REQUEST" message is sent by the UE. AT: "AT+CGACT=1,1" is sent by the TE.
$t_{\text{PDP context activation accept}}$: Time when PDP context activation is complete	Stop: PDP context logo appears in the display of the UE.	Stop: Point of time when the UE receives the "Activate PDP context Accept" message (Layer 3).

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{PDP context activation request}}$: Time of PDP context activation request	Start: User initiates the service access.	Start: Layer 3 (SNDCP): The first "SN-ACTIVATE PDP CONTEXT DEMAND" message is sent by the UE. AT: "ATD*99#" is sent by the TE.
$t_{\text{PDP context activation accept}}$: Time when PDP context activation is complete	Stop: PDP context logo appears in the display of the UE.	Stop: Layer 3 (SNDCP): The "SN-ACTIVATE PDP CONTEXT ACCEPT" message is received by the UE. AT: The "CONNECT" indication is received by the TE.

Remarks:

- While determining the average PDP context activation time only successful activation attempts are included in the calculations (see clause 7.2.5).
- The PDP context activation time should be determined per service, since the service might have impact on the actual activation time, e.g., different access point names (APNs) for wireless application protocol (WAP).

7.2.7 PDP context cut-off ratio [%]

7.2.7.1 Abstract definition

The PDP context cut-off ratio is the probability that a PDP context is deactivated without being initiated intentionally by the user, allowing the network to deactivate the context after user idle time.

7.2.7.2 Abstract equation

$$\text{PDP context cut-off ratio}\,[\%] = \frac{\text{PDP context losses not initiated by the user}}{\text{all successfully activated PDP contexts}} \times 100$$

7.2.7.3 Trigger points

GPRS/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PDP context successfully activated (pre-condition)	Start: PDP context logo appears in the display of the UE.	Start: Layer 3 (SM): The "ACTIVATE PDP CONTEXT ACCEPT" message is received by the UE.
PDP context deactivation initiated by the user	Stop: PDP context logo disappears from the display of the UE.	Stop: Layer 3 (SM): The "DEACTIVATE PDP CONTEXT REQUEST" message is sent by the UE upon desired initiation.
PDP context deactivation initiated by network when user is idle for T seconds	Stop: PDP context logo disappears from the display of the UE.	Stop: Layer 3 (SM): The "DEACTIVATE PDP CONTEXT" message is received by the UE after Idle time.
PDP context loss not initiated by the user	Different trigger points for a PDP context deactivation not initiated intentionally by the user are possible: SGSN failure or GGSN failure on which the PDP context will be deactivated by the SGSN or GGSN No deactivation message received by the UE, but PDP context exhibits loss of connectivity.	

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PDP context successfully activated	Start: PDP context logo appears in the display of the UE.	Start: Layer 3 (SNDCP): The "SN-ACTIVATE PDP CONTEXT ACCEPT" message is received by the UE.
PDP context deactivation initiated by the user	Stop: PDP context logo disappears from the display of the UE.	Stop: Layer 3 (SNDCP): The "SN-DEACTIVATE PDP CONTEXT DEMAND" message is sent by the UE upon desired initiation.
PDP context loss not initiated by the user	Stop trigger point not reached.	

Remarks:

- Precondition for measuring this parameter is that a PDP context was successfully established first.
- Different trigger points for a PDP context deactivation not initiated intentionally by the user are possible: SGSN failure or gateway GPRS support node (GGSN) failure on which the PDP context will be deactivated by the SGSN or GGSN.

7.2.8 Data call access failure ratio [%]

7.2.8.1 Abstract definition

The data call access failure ratio is the probability that a subscriber (A-party) cannot take advantage of a service offering (as shown by the network ID in the display of the user equipment) to establish a data call to a B-party.

7.2.8.2 Abstract equation

$$\text{Data call access failure ratio}\,[\%] = \frac{\text{unsuccessful data call accesses}}{\text{all data call access attempts}} \times 100$$

7.2.8.3 Trigger points

GSM:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Data call access attempt	Start: CONNECT button is pressed.	Start: Layer 3 (RR): The "CHANNEL REQUEST" message is sent over the RACH. AT: The "ATD <dial number>" (MSISDN) command is sent by the A-party.
Successful data call access	Stop: Alerting tone occurs/connection established.	Stop: Layer 3 (CC): The "CONNECT" message is received by the A-party. AT: The "CONNECT" indication is received by the A-party.
Unsuccessful data call access	Stop trigger point not reached.	

Remarks:

- The "ATD <dial number>" (MSISDN) should be sent when there is no ongoing call.
- "AT+CEER" can be used to read out the error cause.

7.2.9 Data call access time [s]

7.2.9.1 Abstract definition

The data call access time is the time elapsing from initiating the data call to alerting or a busy signal that a subscriber (A-party) can take advantage of a service offering (as shown by the network ID in the display of the user equipment) to establish a data call to a B-party.

7.2.9.2 Abstract equation

$$\text{Data call access time}\,[s] = \left(t_{\text{successful call access}} - t_{\text{initiation of data call}}\right)[s]$$

7.2.9.3 Trigger points

GSM:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{initiation of data call}}$: Time of initiation of data call	Start: Time at which CONNECT button is pressed.	Start: Layer 3 (RR): The "CHANNEL REQUEST" message is sent over the RACH. AT: The "ATD <dial number>" (MSISDN) command is sent by the A-party.
$t_{\text{Psuccessful call access}}$: Time of successful data call access	Stop: Any answer received from the mobile network. CONNECT answer handled as a positive answer, any other as a negative one. Time at which alert or busy signal occurs/connection established.	Stop: Layer 3 (CC): The "CONNECT" message is received by the A-party. AT: The "CONNECT" indication is received by the A-party.

Remarks:

- The "ATD <dial number>" (MSISDN) should be sent when there is no ongoing call.
- "AT+CEER" can be used to read out the error cause.

7.2.10 DNS host name resolution failure ratio [%]

7.2.10.1 Abstract definition

The domain name service (DNS) host name resolution failure ratio is the probability that a host name to host address translation of a DNS resolver was not successful.

Remarks:

- this QoS parameter is only relevant for packet switched services;
- resolutions of different host names shall not be compared directly, since the time to perform a search in the DNS server differs depending on the host name;
- resolutions involving different DNS name servers are not directly comparable;
- resolutions utilizing transmission control protocol (TCP) cannot be directly compared to resolutions using user datagram protocol (UDP), since messages carried by UDP are restricted to 512 bytes. UDP is the recommended method for standard queries on the Internet.

7.2.10.2 Abstract equation

$$\text{DNS host name resolution failure ratio}[\%] = \frac{\text{unsuccessful DNS host name resolution requests}}{\text{DNS host name resolution requests}} \times 100$$

7.2.10.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Host name resolution request	Start: Request to resolve a host name.	Start: Protocol: DNS. Data packet containing DNS type A (host address) "Standard query" message for the desired host name.
Successful host name resolution request	Stop: Host address resolved successfully.	Stop: Protocol: DNS. Data packet received containing a type A (host address) "Standard query response, No error" response, the respective type A "Standard query" query and an answer including the desired host name to host address translation.
Unsuccessful host name resolution request	Stop: Host address not resolved.	Stop: Trigger point not reached.

Precondition for measurement:

- The resolver shall not have direct access to any local DNS name server or any name server's zone.

7.2.11 DNS host name resolution time [s]

7.2.11.1 Abstract definition

The DNS host name resolution time is the time it takes to perform a host name to host address translation.

Remarks:

- this QoS parameter is only relevant for packet switched services;
- resolutions of different host names shall not be compared directly, since the time to perform a search in the DNS server differs depending on the host name;
- resolutions involving different DNS name servers are not directly comparable;
- resolutions utilizing TCP cannot be directly compared to resolutions using UDP, since messages carried by UDP are restricted to 512 bytes. UDP is the recommended method for standard queries on the Internet.

7.2.11.2 Abstract equation

$$\text{DNS host name resolution time}\,[\mathrm{s}] = \left(t_{\text{Standard Query Response}} - t_{\text{Standard Query}}\right)[\mathrm{s}]$$

7.2.11.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{StandardQuery}$: Host name resolution request	Start: Request to resolve a host address from DNS server.	Start: Protocol: DNS. Data packet containing DNS type A (host address) "Standard query" query for the desired host name.
$t_{StandardQueryResponse}$: Host name resolution request answered	Stop: Host address received from DNS server.	Stop: Protocol: DNS. Data packet received containing a type A (host address) "Standard query response, No error" response, the respective type A "Standard query" query and an answer including the desired host name to host address translation.

Precondition for measurement:

- The resolver shall not have direct access to any local DNS name server or any name server's zone.
- For static measurement methodologies, as defined in clause 8, the queried DNS name server shall have any data related to the host name to be resolved available as authoritative data in one of the name server's zones, so that no recursive lookups have to be performed and no use of cached information will be required.
- If the related data is not stored locally in the name server's zone, the resolution time would vary due to DNS caching strategies.

7.3 Direct services QoS parameters

7.3.1 File transfer protocol (FTP)

7.3.1.1 FTP {download|upload} service non-accessibility [%]

7.3.1.1.1 Abstract definition

The service non-accessibility ratio denotes the probability that a subscriber cannot establish a PDP context and access the service successfully.

7.3.1.1.2 Abstract equation

$$\text{FTP}\{\text{download}|\text{upload}\}\ \text{service non-accessibility}[\%] = \frac{\text{unsuccessful attempts to reach the point when content is sent or received}}{\text{all attempts to reach the point when content is sent or received}} \times 100$$

7.3.1.1.3 Trigger points

Download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Service access attempt	Start: User initiates the service access.	Start: ATD command.
Successful attempt	Stop: File download starts.	Stop method A: Reception of the first data packet containing the content. Stop method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.
Unsuccessful attempt	Stop trigger point not reached.	

Upload:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Service access attempt	Start: User initiates the service access.	Start: ATD command.
Successful attempt	Stop: File upload starts.	Stop method A: Sending of the first data packet containing the content. Stop method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.
Unsuccessful attempt	Stop trigger point not reached.	

Remark:

- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3).

7.3.1.2 FTP {download|upload} set-up time [s]

7.3.1.2.1 Abstract definition

The set-up time is the time period needed to access the service successfully, from starting the dial-up connection to the point of time when the content is sent or received.

7.3.1.2.2 Abstract equation

$$\text{FTP}\left\{\text{download}\middle|\text{upload}\right\}\text{set-up time}\,[\text{s}] = \left(t_{\text{service access successful}} - t_{\text{service access start}}\right)[\text{s}]$$

7.3.1.2.3 Trigger points

Download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{service\ access\ start}$: Time of service access attempt	Start: User initiates the service access.	Start: ATD command.
$t_{service\ access\ successful}$: Time of successful service access	Stop: File download starts.	Stop method A: Reception of the first data packet containing the content. Stop method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.

Upload:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{service\ access\ start}$: Time of service access attempt	Start: User initiates the service access.	Start: ATD command.
$t_{service\ access\ successful}$: Time of successful service access	Stop: File upload starts.	Stop method A: Sending of the first data packet containing the content. Stop method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.

Remark:

- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3).

7.3.1.3 FTP {download|upload} IP-service access failure ratio [%]

7.3.1.3.1 Abstract definition

The IP-service access failure ratio is the probability that a subscriber cannot establish a TCP/IP connection to the server of a service successfully.

7.3.1.3.2 Abstract equation

$$\text{FTP}\{\text{download}|\text{upload}\}\ \text{IP-service access failure ratio}[\%] = \frac{\text{unsuccessful attempts to establish an IP connection to the server}}{\text{all attempts to establish an IP connection to the server}} \times 100$$

7.3.1.3.3 Trigger points

Download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
IP-Service access attempt	Start: User initiates file download.	Start: First [SYN] sent on the data socket.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful attempt	Stop: File download starts.	Stop method A: Reception of the first data packet containing the content. Stop method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.
Unsuccessful attempt	Stop trigger point not reached.	

Upload:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
IP-Service access attempt	Start: User initiates file upload.	Start: First [SYN] sent on the data socket.
Successful attempt	Stop: File upload starts.	Stop method A: Sending of the first data packet containing the content. Stop method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.
Unsuccessful attempt	Stop trigger point not reached.	

Remark:

- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.1.4 FTP {download|upload} IP-service set-up time [s]

7.3.1.4.1 Abstract definition

The IP-service set-up time is the time period needed to establish a TCP/IP connection to the server of a service, from sending the initial query to a server to the point of time when the content is sent or received.

7.3.1.4.2 Abstract equation

$$\text{FTP}\{\text{download}|\text{upload}\}\,\text{IP-service set-up time}\,[\text{s}] = (t_{\text{IP-Service access successful}} - t_{\text{IP-Service access start}})\,[\text{s}]$$

7.3.1.4.3 Trigger points

Download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{IP-Service access start}}$: Time of IP-Service access attempt	Start: User initiates file download.	Start: First [SYN] sent.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{IP-Service access successful}}$: Time of successful IP-Service access	Stop: File download starts.	Stop method A: Reception of the first data packet containing the content. Stop method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.

Upload:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{IP-Service access start}}$: Time of IP-service access attempt	Start: User initiates file upload.	Start: First [SYN] sent.
$t_{\text{IP-Service access successful}}$: Time of successful IP-service access	Stop: File upload starts.	Stop method A: Sending of the first data packet containing the content. Stop method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.

Remark:

- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.1.5 FTP {download|upload} session failure ratio [%]

7.3.1.5.1 Abstract definition

The session failure ratio is the proportion of uncompleted sessions and sessions that were started successfully.

7.3.1.5.2 Abstract equation

$$\text{FTP}\{\text{download}|\text{upload}\}\text{session failure ratio}[\%] = \frac{\text{uncompleted sessions}}{\text{successfully started sessions}} \times 100$$

7.3.1.5.3 Trigger points

Download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started session	Start: User initiates file download.	Start: First [SYN] sent on the control socket.
Completed session	Stop: File download is successfully completed.	Stop: Reception of the last data packet containing content.
Uncompleted session	Stop trigger point not reached.	

Upload:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started session	Start: User initiates file upload.	Start: First [SYN] sent on the control socket.
Completed session	Stop: File upload is successfully completed.	Stop: Reception of the [FIN, ACK] for the last data packet containing content.
Uncompleted session	Stop trigger point not reached.	

Remark:

- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.1.6 FTP {download|upload} session time [s]

7.3.1.6.1 Abstract definition

The session time is the time period needed to successfully complete a PS data session.

7.3.1.6.2 Abstract equation

$$\text{FTP}\{\text{download}|\text{upload}\}\text{session time}[s] = (t_{\text{sessionend}} - t_{\text{sessionstart}})[s]$$

7.3.1.6.3 Trigger points

Download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{session start}}$: Time of successfully started session	Start: User initiates file download.	Start: First [SYN] sent on the control socket.
$t_{\text{session end}}$: Time when session is completed	Stop: File download is successfully completed.	Stop: Reception of the last data packet containing the content.

Upload:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{session start}}$: Time of successfully started session	Start: User initiates file upload.	Start: First [SYN] sent on the control socket.
$t_{\text{session end}}$: Time when session is completed	Stop: File upload is successfully completed.	Stop: Reception of the [FIN, ACK] for the last data packet containing the content.

Remark:

- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.1.7 FTP {download|upload} mean data rate [kbit/s]

7.3.1.7.1 Abstract definition

The mean data rate is the average data transfer rate measured throughout the entire connect time (i.e., after a data link has been successfully established) to the service. The data transfer shall be successfully terminated. The prerequisite for this parameter is network and service access.

7.3.1.7.2 Abstract equation

$$\text{FTP}\{\text{download} \mid \text{upload}\}\text{mean data rate}\,[\text{kbit/s}] = \frac{\text{user data transferred}\,[\text{kbit}]}{\left(t_{\text{data transfer complete}} - t_{\text{data transfer start}}\right)[\text{s}]}$$

7.3.1.7.3 Trigger points

The average throughput is measured from opening the data connection to the end of the successful transfer of the content (file).

Download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{data transfer start}}$: Time when successfully started the data transfer	Start: File download starts.	Start method A: Reception of the first data packet containing the content. Start method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.
$t_{\text{data transfer complete}}$: Time when data transfer is complete	Stop: File download is successfully completed.	Stop: Reception of the last data packet containing content.

Upload:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{data transfer start}}$: Time when successfully started the data transfer	Start: File upload starts.	Start method A: Sending of the first data packet containing the content. Start method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections; sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.
$t_{\text{data transfer complete}}$: Time when data transfer is complete	Stop: File upload is successfully completed.	Stop: Reception of the [FIN, ACK] for the last data packet containing the content.

Remark:

- The mobile station is already attached (see clause 7.2.3), a PDP context is activated (see clause 7.2.5) and a service was accessed successfully (see service non-accessibility).

7.3.1.8 FTP {download|upload} data transfer cut-off ratio [%]

7.3.1.8.1 Abstract definition

The data transfer cut-off ratio is the proportion of incomplete data transfers and data transfers that were started successfully.

7.3.1.8.2 Abstract equation

$$\text{FTP}\{\text{download}|\text{upload}\}\text{data transfer cut-off ratio}[\%] = \frac{\text{incomplete data transfers}}{\text{successfully started data transfers}} \times 100$$

7.3.1.8.3 Trigger points

Download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started data transfer	Start: File download starts.	Start method A: Reception of the first data packet containing the content. Start method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.
Complete data transfer	Stop: File download successfully completed.	Stop: Reception of the last data packet containing content.
Incomplete data transfer	Stop trigger point not reached.	

Upload:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started data transfer	Start: File upload starts.	Start method A: Sending of the first data packet containing the content. Start method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.
Complete data transfer	Stop: File upload is successfully completed.	Stop: Reception of the [FIN, ACK] for the last data packet containing the content.
Incomplete data transfer	Stop trigger point not reached.	

Remark:

- The mobile station is already attached (see clause 7.2.3), a PDP context is activated (see clause 7.2.5) and a service was accessed successfully (see service non-accessibility).

7.3.2 Mobile broadcast

Mobile broadcast is an end-to-end broadcast system for delivery of any types of digital content and services using IP-based mechanisms. An inherent part of the mobile broadcast system is that it comprises of a unidirectional broadcast path (e.g., DVB-H, MBMS, and other broadcast bearers) and a bidirectional mobile/cellular interactivity path (e.g., GSM, GPRS or UMTS). The mobile broadcast service is thus a platform for convergence of services from mobile/cellular and broadcast/media domains.

Figure 7-4 depicts the basis for a generic service concept for mobile broadcast. As being a composite service, two different bearers may be involved in the mobile broadcast services. Unidirectional broadcast information is transmitted over the broadcast channel, whereas interactive procedures are related to the interactivity channel provided by a mobile network. The independent procedures at both bearers may interact with each other and build a common end-to-end procedure.

In general, this concept is not dedicated to specific bearer technologies as is possible with different bearer technologies or their combinations.

Remarks:

- The concept depends, for example, on the implementation of the electronic service guide (ESG). If the ESG implementation does not allow the user to recognize the reception of ESG information, the corresponding parameters have to be adapted.
- Content encryption may be a central element of DVB-H implementations. This issue is not dealt with explicitly in what follows and needs further consideration.

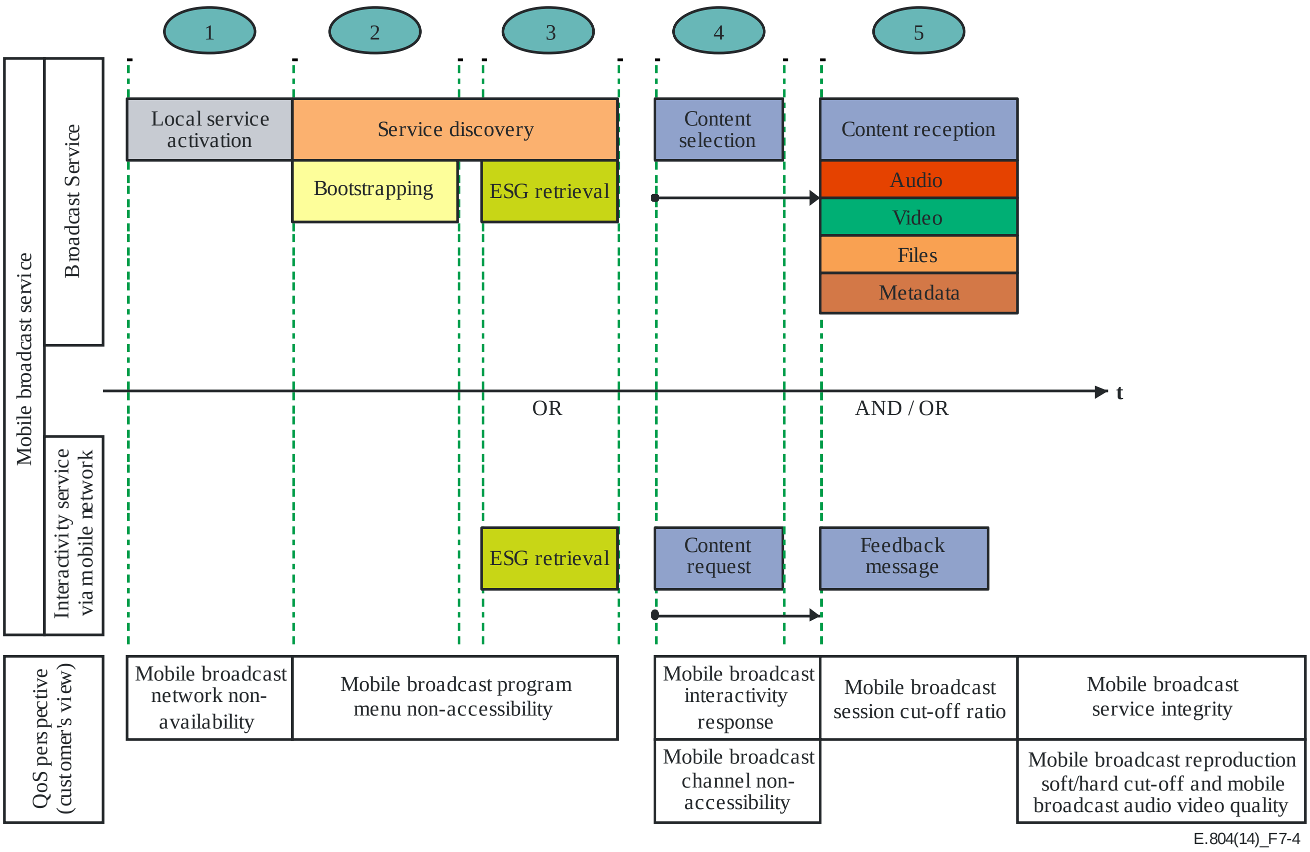


Figure 7-4 – Service phases of mobile broadcast

From a user's point of view, the usage cycle of mobile broadcast services can be divided into:

- Terminal registration/local service activation: The broadcast receiver is switched on and the terminal registers to the broadcast bearer. This procedure includes the detection of a broadcast service signal.
- Bootstrapping: During this phase, the detected broadcast signal is decoded. At the end of this phase, a list of receivables channels is available. Each channel offers additional information via its own electronic service guide (ESG).

- ESG retrieval: At this stage, the information where to find ESG information is available. After a channel is selected, the channel related information is received, decoded and presented to the user. For example, an overview over the current and following programmes can be shown on the display. The ESG information itself can be retrieved either via the broadcast bearer or via the interactivity service, for example via a WAP portal.
- Service discovery: This phase includes the bootstrapping phase and the ESG retrieval phase. Please note that manual channel selection may lead to an additional delay between both phases. During this phase, the detected broadcast signal is decoded. Afterwards, the information on where to find ESG information is available. The ESG information itself can be retrieved either via the broadcast bearer or via the interactivity service, for example via a WAP portal.
- Content reception: The generic term "content" comprises all kinds of content that can be transferred via the broadcast service. Examples for this kind of data are audio and video streams, file downloads and related metadata which describes the carried content.
- Interactivity based procedures: These procedures allow the interactive use of the mobile broadcast service. In general, all transmission capabilities offered by the mobile network can be used for this issue. Examples are:
  – content requests via a WAP GET request;
  – SMS voting;
  – request to receive ESG information via MMS service; or
  – voice control to request a dedicated file via the broadcast service.

The technical interpretation of this generic usage cycle leads to the phases:

- Mobile broadcast network non-accessibility.
- Mobile broadcast programme menu non-accessibility.
- Mobile broadcast channel non-accessibility.
- Mobile broadcast interactivity response.
- Mobile broadcast session cut-off ratio.
- Mobile broadcast service integrity.

The mentioned phases are covered by the parameters described subsequently.

7.3.2.1 Mobile broadcast network non-availability (broadcast bearer)

7.3.2.1.1 Abstract definition

The mobile broadcast network non-availability is the probability that the mobile broadcast services are not offered to an end user by the target network indicators on the user equipment (UE) in idle mode.

7.3.2.1.2 Abstract equation

$$\text{Mobile broadcast network non-accessibility}\,[\%] = \frac{\text{unsuccessful mobile broadcast registration attempts}}{\text{all mobile broadcast registration attempts}} \times 100$$

7.3.2.1.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Mobile broadcast registration attempt	Start: Start of registration procedure performed by the UE.	To be defined
Unsuccessful mobile broadcast registration attempt	Stop: "Mobile broadcast icon", which indicates successful registration, is not displayed on the UE.	To be defined

Preconditions for measurement:

- The terminal shall be in an area which is intended to be covered by the broadcast service.
- The receiver responsible for the reception of the mobile broadcast services shall be activated and initialized.

7.3.2.2 Mobile broadcast programme menu non-accessibility (bootstrapping bearer, ESG retrieval bearer)

7.3.2.2.1 Abstract definition

The mobile broadcast programme menu non-accessibility is the probability that the mobile broadcast programme menu is not successfully accessible by the user when requested.

Remark:

- This parameter depends on the actual implementation of the service discovery procedures (e.g., use of cached bootstrapping and/or ESG information).

7.3.2.2.2 Abstract equation

$$\text{Mobile broadcast programme menu non-accessibility}\,[\%] = \frac{\text{unsuccessful programme menu access attempts}}{\text{all programme menu access attempts}} \times 100$$

7.3.2.2.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Mobile broadcast programme menu access attempt	Start: Request to use the mobile broadcast service on the UE (push on TV button).	Start: To be defined
Unsuccessful mobile broadcast programme menu access attempt	Stop: Mobile broadcast service is not available on the UE (no TV channel list displayed).	Stop: To be defined

Preconditions for measurement:

- Mobile broadcast network availability must be given.

7.3.2.3 Mobile broadcast programme menu access time (bootstrapping bearer, ESG retrieval bearer)

7.3.2.3.1 Abstract definition

The mobile broadcast programme menu access time is the time period elapsed between a session start attempt of the mobile broadcast service and the reception of the complete menu channels list. Hereby, the time the device requires to discover the available channels for the first time is considered.

7.3.2.3.2 Abstract equation

$$\text{Mobile broadcast program menu access time}\,[s] = \left(t_{\text{program menu reception}} - t_{\text{program menu request}}\right)[s]$$

7.3.2.3.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{program menu request}}$: Time of mobile broadcast programme menu	Start: First request for the mobile broadcast service on the UE.	Start: To be defined
$t_{\text{program menu reception}}$: Time of successful mobile broadcast programme menu reception	Stop: Mobile broadcast channel list is given within a pre-determined time.	Stop: To be defined

Preconditions for measurement:

- Mobile broadcast network availability must be given.

7.3.2.4 Mobile broadcast channel non-accessibility (broadcast bearer)

7.3.2.4.1 Abstract definition

The mobile broadcast channel non-accessibility is the probability that the requested mobile broadcast channel is not started to be delivered to the user. This parameter applies also to zapping situations in which the user changes the offered streaming content frequently in short intervals.

7.3.2.4.2 Abstract equation

$$\text{Mobile broadcast channel non-accessibility}\,[\%] = \frac{\text{unsuccessful channel access attempts}}{\text{all channel attempts}} \times 100$$

7.3.2.4.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Channel access attempt	Start: Request channel button pressed by the user/request attempt from device.	Start: To be defined
Unsuccessful channel access attempt	Stop: Missing indication of reception of the channel content (channel displayed).	Stop: To be defined

Preconditions for measurement:

- Mobile broadcast network availability must be given.
- Mobile broadcast programme menu accessibility must be successful.

7.3.2.5 Mobile broadcast channel access time (broadcast bearer)

7.3.2.5.1 Abstract definition

The parameter mobile broadcast channel access time is the time period elapsed between the user's request to access the channel and the channel reception/displayed.

7.3.2.5.2 Abstract equation

$$\text{Mobile broadcast channel access time}[s] = \left(t_{channelreception} - t_{channelrequest}\right)[s]$$

7.3.2.5.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{channel\ request}$: Channel request time	Start: Request channel button pressed by the user.	Start: To be defined
$t_{channel\ reception}$: Channel reception time	Stop: reception of the channel content (channel displayed).	Stop: To be defined

Preconditions for measurement:

- Mobile broadcast network availability must be given.
- Mobile broadcast programme menu accessibility must be successful.

7.3.2.6 Mobile broadcast interactivity response failure ratio (mobile network bearer, broadcast bearer)

7.3.2.6.1 Abstract definition

The mobile broadcast interactivity response failure ratio measures the probability that a service request of a mobile broadcast service via an interactive channel does not result in an expected reaction (i.e., changes in content updated due to user's interaction, reception of any kind of notification to the user, etc.) on either the broadcast bearer or the mobile network bearer.

7.3.2.6.2 Abstract equation

$$\text{Mobile broadcast interactivity response failure ratio}[\%] = \frac{\text{unsuccessful mobile broadcast service outcomes/responses}}{\text{all mobile broadcast service requests over interactive channel}} \times 100$$

7.3.2.6.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Mobile broadcast service request over interactive channel	Start: Request of the mobile broadcast service on the UE.	Start: To be defined Content request on interactivity channel: Trigger points are chosen according to the parameter definitions for SMS, MMS, GPRS and PS-UMTS in this Recommendation.
Unsuccessful mobile broadcast service outcome/response	Stop: User's interactivity is not reflected in the updated content or indicated at the device.	Stop: To be defined Negative result code or timeout related to interactivity channel: Trigger points are chosen according to the parameter definitions for SMS, MMS, GPRS and PS-UMTS in this Recommendation.

Preconditions for measurement:

- For broadcast bearer:
  - Mobile broadcast network availability must be given.
- For mobile network bearer:
  - Mobile network availability must be given.
  - Mobile network service accessibility for circuit switched or packet switched data services must be given.

7.3.2.7 Mobile broadcast interactivity response time (mobile network bearer, broadcast bearer)

7.3.2.7.1 Abstract definition

The parameter mobile broadcast interactivity response time is the time elapsed between a service request attempt of the mobile broadcast service via an interactive channel and the reception of a notification to the user.

7.3.2.7.2 Abstract equation

$$\text{Mobile broadcast interactivity response time}\,[s] = \left(t_{\text{service response}} - t_{\text{service request}}\right)[s]$$

7.3.2.7.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{service request}}$: Mobile broadcast service request over interactive channel	Start: Request of the mobile broadcast service on the UE.	Start: To be defined Content request on interactivity channel: Trigger points are chosen according to the parameter definitions for SMS, MMS, GPRS and PS-UMTS in this Recommendation.
$t_{\text{service response}}$: Successful mobile broadcast service outcome/response	Stop: User's interactivity is not reflected in the updated content or indicated at the device.	Stop: To be defined Negative result code or timeout related to interactivity channel: Trigger points are chosen according to the parameter definitions for SMS, MMS, GPRS and PS-UMTS in this Recommendation.

Preconditions for measurement:

- For broadcast bearer:
  - Mobile broadcast network availability must be given.
- For mobile network bearer:
  - Mobile network availability must be given.
  - Mobile network service accessibility for circuit switched or packet switched data services must be given.

7.3.2.8 Mobile broadcast session cut-off ratio (broadcast bearer)

7.3.2.8.1 Abstract definition

The session cut-off ratio is the probability of abnormal termination of the specific service requested by the user.

7.3.2.8.2 Abstract equation

$$\text{Mobile broadcast session cut-off ratio}[\%] = \frac{\text{unsuccessfully terminated sessions}}{\text{all successfully established sessions}} \times 100$$

7.3.2.8.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully established sessions	Start: Channel reproduction started.	Start: To be defined
Unsuccessfully terminated sessions	Stop: Channel reproduction terminated abnormally (exit from the service).	Stop: To be defined

Preconditions for measurement:

- Mobile broadcast network availability must be given.
- Mobile broadcast programme menu accessibility must be successful.
- Mobile broadcast channel accessibility must be successful.

7.3.2.9 Mobile broadcast service integrity (broadcast bearer)

The mobile broadcast technology paves the way for network operators and service providers to offer a significant number of mobile services, which can be divided into the following categories:

- Streaming services.
- Packet switched data services.
- Short message service (SMS).
- Multimedia message service (MMS).
- Wireless application protocol (WAP).
- Digital video broadcasting – handheld (DVB-H).

According to [ITU-T E.800], the 'service integrity' describes the QoS during service use. Since the above mentioned services are already offered in other scenarios, in this Recommendation only a reference to the already defined QoS parameters will be made. Important to bear in mind is the fact that for mobile broadcast service, only the abstract definition of the parameters applies, since the underlying protocol stack may not be the same.

7.3.2.10 Mobile broadcast reproduction soft cut-off ratio (broadcast bearer)

7.3.2.10.1 Abstract definition

The mobile broadcast reproduction soft cut-off ratio denotes the probability that the end-user cannot see normally the channel when connected to the specific service.

7.3.2.10.2 Abstract equation

$$\text{Mobile broadcast reproduction soft cut-off ratio}[\%] = \frac{\sum\left(t_{\text{fluid audio/video restart}} - t_{\text{signal weak}}\right)}{t_{\text{reproduction finished}} - t_{\text{reproduction started}}} \times 100$$

7.3.2.10.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{signal weak}}$	Start: Channel displays, e.g., blue screen "TV signal weak search of the signal in progress".	Start: To be defined
$t_{\text{fluid audio/video restart}}$	Stop: The service restarts normally.	Stop: To be defined
$t_{\text{reproduction started}}$	Start: Beginning of reproduction.	Start: To be defined
$t_{\text{reproduction finished}}$	Stop: End of reproduction.	Stop: To be defined

Preconditions for measurement:

- Mobile broadcast network availability must be given.
- Mobile broadcast programme menu accessibility must be successful.
- Mobile broadcast channel accessibility must be successful.

7.3.2.11 Mobile broadcast reproduction hard cut-off ratio (broadcast bearer)

7.3.2.11.1 Abstract definition

The mobile broadcast reproduction hard cut-off ratio denotes that the end user cannot see normally the channel when connected to the specific service.

7.3.2.11.2 Abstract equation

$$\text{Mobile broadcast reproduction hard cut-off ratio}[\%] = \frac{\sum\left(t_{\text{fluid audio/video restart}} - t_{\text{signal absent}}\right)}{t_{\text{reproduction finished}} - t_{\text{reproduction started}}} \times 100$$

7.3.2.11.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{signal absent}}$	Start: Channel displays, e.g., blue screen "TV signal absent tuning in progress".	Start: To be defined
$t_{\text{fluid audio/video restart}}$	Stop: The service restarts normally.	Stop: To be defined
$t_{\text{reproduction started}}$	Start: Beginning of reproduction.	Start: To be defined
$t_{\text{reproduction finished}}$	Stop: End of reproduction.	Stop: To be defined

Preconditions for measurement:

- Mobile broadcast network availability must be given.
- Mobile broadcast programme menu accessibility must be successful.
- Mobile broadcast channel accessibility must be successful.

7.3.2.12 Mobile broadcast audio quality (broadcast bearer)

7.3.2.12.1 Abstract definition

The mobile broadcast audio quality describes the audio quality as perceived by the end user. Since the streams can contain not only the speech information, an algorithm like that of [ITU-T P.862] is not also suitable for all scenarios and should not be used.

An audio algorithm, such as that of [ITU-R BS.1387-1] may be used.

7.3.2.12.2 Abstract equation

To be defined.

7.3.2.12.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
To be defined	Start: Beginning of the audio reproduction.	Start: To be defined
To be defined	Stop: End of audio reproduction.	Stop: To be defined

Preconditions for measurement:

- Mobile broadcast network availability must be given.
- Mobile broadcast programme menu accessibility must be successful.
- Mobile broadcast channel accessibility must be successful.

7.3.2.13 Mobile broadcast video quality (broadcast bearer)

7.3.2.13.1 Abstract definition

The mobile broadcast video quality describes the video quality as perceived by the end user.

7.3.2.13.2 Abstract equation

To be defined.

7.3.2.13.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
To be defined	Start: Beginning of video reproduction.	Start: To be defined
To be defined	Stop: End of video reproduction.	Stop: To be defined

Preconditions for measurement:

- Mobile broadcast network availability must be given.
- Mobile broadcast programme menu accessibility must be successful.
- Mobile broadcast channel accessibility must be successful.

7.3.3 Ping

7.3.3.1 Ping round trip time [ms]

7.3.3.1.1 Abstract definition

The ping round trip time is the time required for a packet to travel from a source to a destination and back. It is used to measure the delay on a network at a given time. For this measurement, the service must have already been established.

7.3.3.1.2 Abstract equation

$$\text{Ping round trip time}\,[\text{ms}] = \left(t_{\text{packet received}} - t_{\text{packet sent}}\right)[\text{ms}]$$

7.3.3.1.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{packet sent}}$: Time when the packet is sent	Start: User starts to ping the client.	Start: ICMP echo request sent.
$t_{\text{packet received}}$: Time when the packet is received	Stop: Echo reply is displayed.	Stop: ICMP echo reply received by the sender.

As an alternative, the measurement of the round trip time can be done by considering the TCP handshake:

- Start: Point of time when the [SYN] is sent.
- Stop: Point of time when the [SYN, ACK] is received.

This applies to all services that are TCP based, e.g., file transfer (FTP), web browsing (HTTP) and e-mail (POP3, SMTP).

7.3.4 Push to talk over cellular (PoC)

This clause describes QoS parameters for the push to talk over cellular (PoC) service as described in [OMA-1], [OMA-2], [OMA-3] and [IETF RFC 3903].

To point out the development and effectiveness of these QoS parameters, a generic PoC signal flow is given. Here, some restricted information on the application layer is given. These events only show important user interactions. In this context it is important to point out that this Recommendation does not focus on the application layer or the user plane as described in [OMA-2].

The session description protocol (SDP) is not mentioned as an alternative to real time transport protocol (RTP) in [OMA-1], [OMA-2] and [OMA-3]. Thus, trigger points defined on the SDP layer are out of scope of this Recommendation.

NOTE – All QoS parameters defined for the PoC service which do not rely on RTP in terms of trigger point definition are to be applied when measuring a PoC service utilizing secure real time transfer protocol (SRTP).

Furthermore, some typical PoC signal flows are given in Appendix III together with some signal grouping. Here, signals have been grouped together in order to give a better insight into the signal flow details and their relation to some specific group of PoC QoS parameters.

The PoC service is characterized by a half-duplex form of communication, whereby one end user will communicate with other end users by pressing a button, or an equivalent function, on a terminal. In the following text it will be assumed without loss of generality that the terminal has a PoC button.

It is important to keep in mind that measurement equipment and techniques used can affect the data collected. The measurement equipment and techniques should be defined and their effects documented for all tests.

Remarks:

- All end trigger points defined in this Recommendation will occur after the appropriate start trigger points. The message flow between each two trigger points is described in the text or there is a reference to a figure that visualizes the message flow.
- All session initiation protocol (SIP) and RTP messages that are sent during a PoC session utilize UDP as transport layer.
- If a trigger point (technical description/protocol part) in this Recommendation states: "First data packet sent…" then the time stamp shall be the point in time when the message is posted to the UDP transport layer.
- If a trigger point (technical description/protocol part) in this Recommendation states: "First data packet received…" then the time stamp shall be the point in time when the message is received on the UDP transport layer.
- Trigger points for failure ratios (technical description/protocol part) may state: "No message received by the terminal within a pre-determined time", which means that the PoC server timed out. Here, the exact timeout has to be specified.
- If this Recommendation states: "active PoC talk session", then a PoC session with at least two joining parties is meant, regardless of the kind of session (1-1, ad hoc group talk, pre-arranged group talk or chat). Furthermore, one of the participating terminals shall create and send data packets containing speech data (RTP media stream).
- Unless explicitly stated differently, all terminals participating in the PoC sessions shall not generate notification messages. Otherwise, "SIP NOTIFY" messages may be sent to these clients leading to possible impacts on the measurement results.

7.3.4.1 Definitions

For the PoC, there are differences between on-demand and pre-established PoC sessions which need to be taken into account. Thus, a direct comparison between these session types shall be avoided.

Another difference to be aware of is the form of indication used. If confirmed indication is used, the initiator has to wait for the "talk burst granted" indication until at least one invited user has accepted the invitation. If unconfirmed indication is used, at least one invited user has to be registered and uses automatic answer. This results in different message flows as well as in different response times (especially if media buffering is supported by the PoC server).

Particularities occur when using a pre-arranged PoC group session. In this kind of session, the initiator invites a group of users. With confirmed indication, at least one user has to accept the invitation but with unconfirmed indication the right-to-speak is granted at once; regardless if a user of the group is connected to the PoC service or not.

Table 7-1 gives an overview of the defined QoS parameters. Groups of parameters are introduced to visualize interdependencies. The reason is that certain measurements can only take place if several preconditions are fulfilled.

Table 7-1 – QoS parameter and required preconditions

QoS group		Description	QoS parameter in this group	Preconditions
REG		PoC registration	7.3.4.3, 7.3.4.4	-
PUB		PoC publish	7.3.4.5, 7.3.4.6	REG
REG long		PoC registration + PoC publish	7.3.4.6.3, 7.3.4.7.3	-
On demand	INIT	PoC session initiation	7.3.4.8.3,7.3.4.9.3	PUB
	SETUP	PoC session set-up	7.3.4.14.3, 7.3.4.17	-
	PtS	Push to speech	7.3.4.18, 7.3.4.19	PUB
	LEAVE	PoC session leaving	7.3.4.20, 7.3.4.21	INIT or SETUP
Pre-established	NEGO	PoC media parameters negotiation	7.3.4.10.3, 7.3.4.11.3	PUB
	INIT	PoC session initiation	7.3.4.12.3, 7.3.4.14	NEGO
	SETUP	PoC session set-up	7.3.4.16, 7.3.4.17	-
	PtS	Push to speech	7.3.4.18, 7.3.4.19	PUB
	LEAVE	PoC session leaving	7.3.4.22, 7.3.4.23	INIT or SETUP
DeREG		PoC deregistration	7.3.4.24, 7.3.4.25	REG or SETUP
BUSY		Busy floor response	7.3.4.26, 7.3.4.27	SETUP or PtS
REQ		Talk burst request	7.3.4.28, 7.3.4.29	SETUP or PtS
CUT		PoC session cut-off	7.3.4.30	SETUP or PtS
DROP		Talk burst drop	7.3.4.31	SETUP or PtS
DELAY		Talk burst delay	7.3.4.32, 7.3.4.33	SETUO or PtS

7.3.4.2 Generic signal flow

This clause gives an overview of some signal flows evolving from PoC sessions. In Figure 7-5, a generic signal flow is given. Here, the main parts of a PoC session, also including the registration of the PoC service, are visualized. These are: PoC service registration (including PoC service settings establishment), PoC session initiation, PoC talk session, PoC session leaving and finally the PoC service deregistration.

Most of the PoC relevant (application layer-) events generated from or receivable by the user are included in Figure 7-5. These events are represented as dashed lines.

In this Recommendation greyed lines are optional signals which do not have to be sent (like the "SIP NOTIFY" message which will only be sent by the PoC server if the "norefersub" option tag was included in the "SIP REFER" request (see [OMA-3])). Provisional SIP responses as described in [OMA-3] (e.g., "SIP 100 Trying") are greyed for clarity. These messages are provisional responses and shall be turned off during measurements.

A generic PoC session:

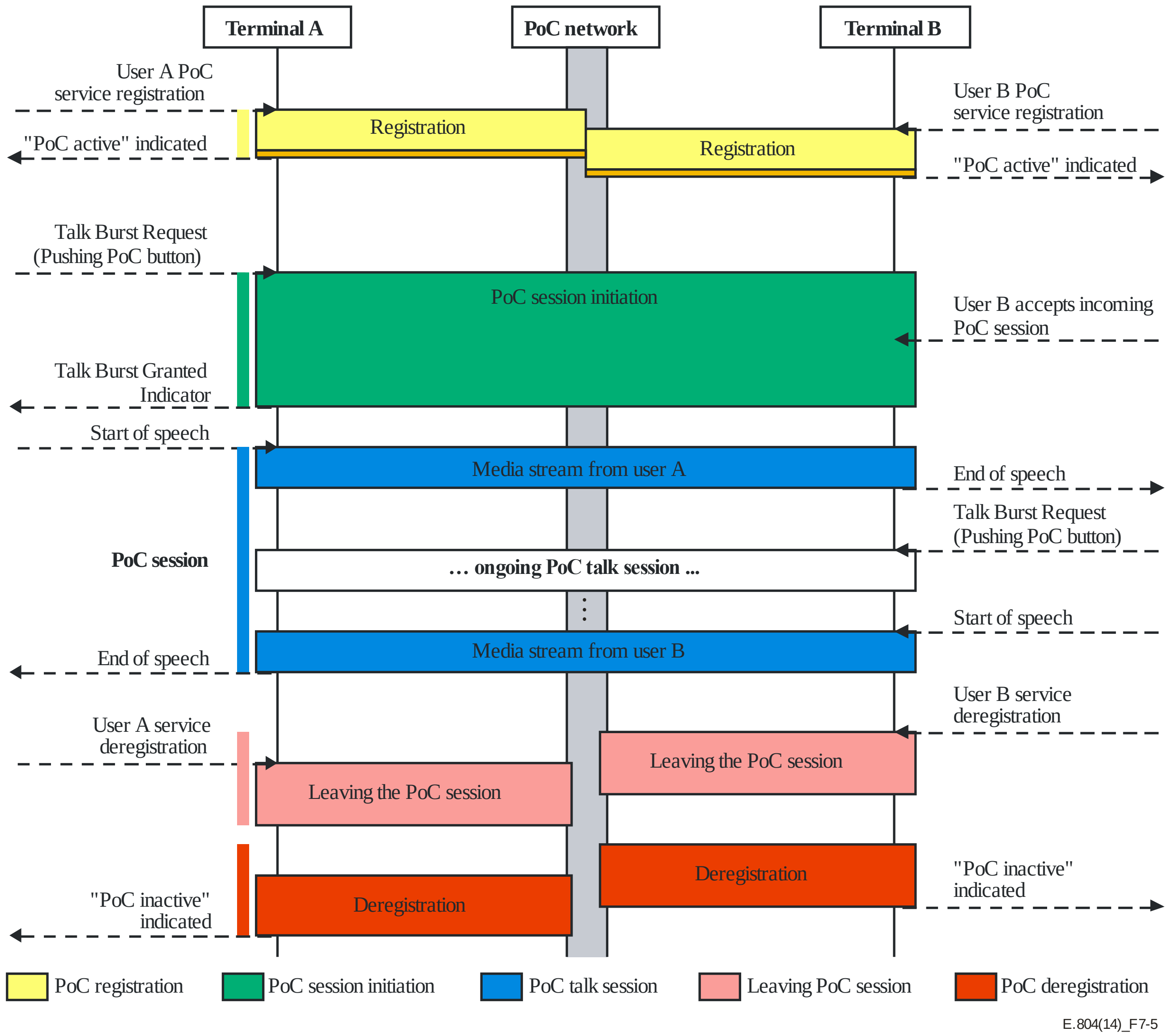
PoC registration.

PoC session initiation.

PoC talk session.

Leaving PoC session.

PoC deregistration.



NOTE – Here, the dashed arrows indicate events generated from or receivable by the user.

Figure 7-5 – Generic PoC session signal flow (including PoC service registration) on application layer

7.3.4.3 PoC registration failure ratio [%]

7.3.4.3.1 Abstract definition

The PoC registration failure ratio is the probability that the terminal cannot register with the PoC service when requested.

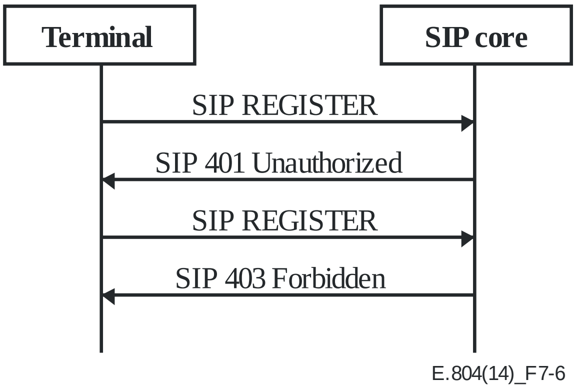
Remark:

- The terminal shall not be registered to the PoC service.

Figure 7-6 shows an example of an unsuccessful PoC registration procedure.

7.3.4.3.2 Abstract equation

$$\text{PoC registration failure ratio}[\%] = \frac{\text{unsuccessful PoC registration attempts}}{\text{all PoC registration attempts}} \times 100$$



NOTE – After the first "SIP REGISTER" request the terminal has to answer to a www-authentication challenge (see [OMA-3]). If the terminal does not answer correctly to this challenge, the SIP core will send a "SIP 403 Forbidden" message.

Figure 7-6 – Unsuccessful PoC registration example

7.3.4.3.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC registration attempt	Start: Activation of the PoC service on the terminal.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REGISTER" message.
Successful PoC registration attempt	Stop: PoC available is indicated.	Stop: Protocol: SIP. First data packet received containing a "SIP 200 OK" message.
Unsuccessful PoC registration attempt	Stop: PoC available indication is not given within a pre-determined time.	Stop: Protocol: SIP. Case 1: Second data packet received by the terminal (after sending the "SIP REGISTER" message) containing a message different from "SIP 200 OK". This message may be implementation-dependent (see [OMA-3]). Case 2: First data packet received by the terminal (after the authentication procedure) containing a message different from "SIP 200 OK". Case 3: No message received by the terminal within a pre-determined time.

7.3.4.4 PoC registration time [s]

7.3.4.4.1 Abstract definition

The PoC registration time is the time period between the registration request of the PoC service and being registered to the PoC service.

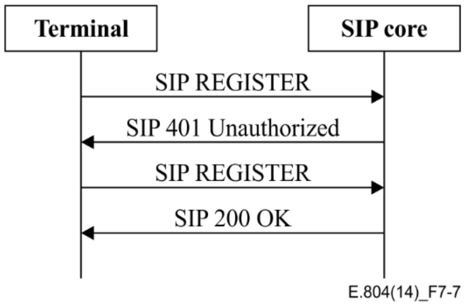
Remark:

- The terminal shall not be registered to the PoC service.

Figure 7-7 shows the message flow for an example of a successful PoC registration procedure (see [OMA-3]).

7.3.4.4.2 Abstract equation

$$\text{PoC registration time}[s] = (t_{PoCAvailable} - t_{PoCActivated})[s]$$



NOTE – In contrast to Figure 7-11, the terminal answered correctly to the authentication challenge (the second "SIP REGISTER" message).

Figure 7-7 – Successful PoC registration example

7.3.4.4.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{PoCActivated}$: Time of the PoC registration attempt	Start: Activation of the PoC service on the terminal.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REGISTER" message.
$t_{PoCAvailable}$: Time of successful PoC registration attempt	Stop: PoC available is indicated.	Stop: Protocol: SIP. First data packet received containing a "SIP 200 OK" message.

7.3.4.5 PoC publish failure ratio [%]

7.3.4.5.1 Abstract definition

The PoC publish failure ratio is the probability that the terminal cannot successfully publish its PoC service settings to the PoC server, after the terminal is registered to the PoC service.

Remarks:

- To set, update or refresh the PoC service settings, the terminal generates a "SIP PUBLISH" request with XML MIME content according to the rules and procedures of [IETF RFC 3903].
- The terminal shall be registered to the PoC service.
- PoC enabled user equipment may combine the PoC registration and the PoC publish request and may not give the user the opportunity to do these actions separately.

7.3.4.5.2 Abstract equation

$$\text{PoC publish failure ratio}[\%] = \frac{\text{unsuccessful PoC publish attempts}}{\text{all PoC publish attempts}} \times 100$$

7.3.4.5.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC publish attempt	Start: Attempt to publish the terminals PoC service settings.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP PUBLISH" message.
Successful PoC publish attempt	Stop: PoC service settings are published.	Stop: Protocol: SIP. First data packet received containing a "SIP 200 OK" message.
Unsuccessful PoC publish attempt	Stop: PoC service settings are not published.	Stop: Protocol: SIP Case 1: Data packet received by the terminal containing a message different from "SIP 200 OK". Case 2: No message received by the terminal within a pre-determined time.

7.3.4.6 PoC publish time [s]

7.3.4.6.1 Abstract definition

The PoC publish time is the amount of time that it takes to publish the terminal's PoC service settings to the PoC server.

Remarks:

- To set, update or refresh the PoC service settings, the terminal generates a "SIP PUBLISH" request with XML MIME content according to rules and procedures of [IETF RFC 3903].
- The terminal shall be registered to the PoC service.
- PoC enabled user equipment may combine the PoC registration and the PoC publish request and may not give the user the opportunity to do these actions separately.

Figure 7-8 shows the message flow for an example of a successful publish of PoC service settings.

7.3.4.6.2 Abstract equation

$$\text{PoC publish time}[s] = (t_{PoCPublishEnd} - t_{PoCPublishStart})[s]$$

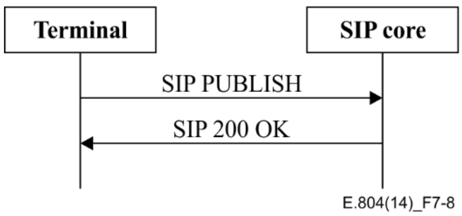


Figure 7-8 – Example of a successful publish of PoC service settings

7.3.4.6.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{PoCPublishStart}$: Time of PoC publish attempt	Start: Attempt to publish the terminals PoC service settings.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP PUBLISH" message.
$t_{PoCPublishEnd}$: Time of successful PoC publish attempt	Stop: PoC service settings are published.	Stop: Protocol: SIP. First data packet received containing a "SIP 200 OK" message.

7.3.4.7 PoC registration failure ratio (long) [%]

7.3.4.7.1 Abstract definition

The PoC registration failure ratio (long) is the probability that the terminal cannot successfully be registered to the PoC service and publish its PoC service settings.

Remarks:

- This QoS parameter is a combination of the PoC registration parameter (see clause 7.3.4.3) and the PoC publish parameter (see clause 7.3.4.5). It ought to reflect the behaviour of PoC enabled user equipment that may do the PoC publish automatically after the PoC register.
- The terminal shall not be registered to the PoC service.

7.3.4.7.2 Abstract equation

$$\text{PoC registration failure ratio (long) } [\%] = \frac{\text{R} + \text{P}}{\text{all PoC registration (long) attempts}} \times 100$$

R and P respectively indicate PoC registration and PoC publication.

7.3.4.7.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC registration attempt	Start: Activation of the PoC service on the terminal.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REGISTER" message.
Successful PoC publish attempt	Stop: PoC service settings are published.	Stop: Protocol: SIP. First data packet received containing a "SIP 200 OK" message.
Unsuccessful PoC publish attempt	Stop: PoC service settings are not published.	Stop: Protocol: SIP Case 1: Data packet received by the terminal containing a message different from "SIP 200 OK". Case 2: No message received by the terminal within a pre-determined time.

7.3.4.8 PoC registration time (long) [s]

7.3.4.8.1 Abstract definition

The PoC registration time (long) is the combined duration for a SIP registration and a SIP publish.

Remarks:

- This QoS parameter is a combination of the PoC registration parameter (see clause 7.3.4.3) and the PoC publish parameter (see clause 7.3.4.5). It ought to reflect the behaviour of PoC enabled user equipment that may do the PoC publish automatically after the PoC registers.
- The terminal shall not be registered to the PoC service.

Figure 7-9 shows message flow for an example of a successful PoC registration (long) procedure.

7.3.4.8.2 Abstract equation

$$\text{PoC registration time (long)}\,[s] = \left(t_{PoCPublishEnd} - t_{PoCPublishStart}\right)[s]$$

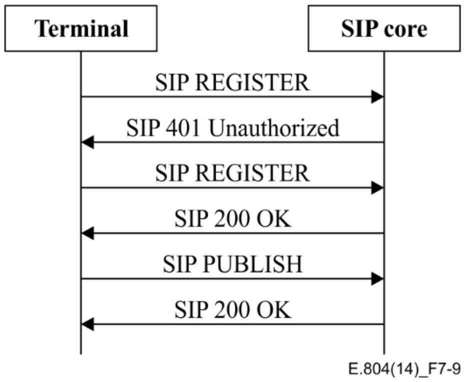


Figure 7-9 – Example of a successful PoC registration (long)

7.3.4.8.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{PoCActivated}$: Time of PoC registration attempt	Start: Activation of the PoC service on the terminal.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REGISTER" message.
$t_{PoCPublishEnd}$: Time of successful PoC publish attempt	Stop: PoC service settings are published.	Stop: Protocol: SIP. First data packet received containing a "SIP 200 OK" message.

7.3.4.9 PoC session initiation failure ratio (on-demand) [%]

7.3.4.9.1 Abstract definition

The PoC session initiation failure ratio (on-demand) is the probability that a PoC session cannot be successfully initiated. A PoC session is initiated when the user pushes the PoC button on the terminal (and thereby requests a talk burst) and is granted a talk burst (see Figure 7-10).

Remarks:

- The terminal notifies the user about the granted talk burst (e.g., by a "beep"-tone).
- There shall be at least one other participating terminal and the floor shall be idle. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- All terminals shall be registered to the PoC service and shall have successfully published their PoC service settings.
- There are different signal flows for confirmed and for unconfirmed invitations. In the confirmed case, at least one of the invited users has to accept the invitation to the PoC session

in order to get the talk burst granted (see [OMA-3]). If the PoC server supports media buffering, the talk burst confirm is send after the first received auto-answer. This automatic answer mode shall be used for the measurements and media buffering shall not be supported. In both cases (confirmed and unconfirmed) the trigger points for the measurement are the same. Data from confirmed and unconfirmed measurements cannot be directly compared.

- This parameter is applicable to different kinds of PoC session initiations, which has an impact on the comparability of the measurement data.
- The initial "SIP INVITE" message accepted by the PoC server is an implicit talk burst request.

7.3.4.9.2 Abstract equation

$$\text{PoC session initiation failure ratio (on-demand)}\ [\%] = \frac{\text{unsuccessful PoC session initiations}}{\text{all PoC session initiations}} \times 100$$

7.3.4.9.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC session initiation attempt	Start: The PoC button is pushed.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP INVITE" message.
Successful PoC session initiation attempt	Stop: Talk burst granted is indicated.	Stop: Protocol: RTCP:TBCP First data packet received by the terminal containing "RTCP:TBCP Talk Burst Granted".
Unsuccessful PoC session initiation attempt	Stop: Missing talk burst granted indication.	Stop: Protocol: SIP; RTCP: TBCP. Case 1: First data packet received by the terminal (after sending a "SIP INVITE" message) containing an error message or redirection message (e.g., a "403 Forbidden" or "488 Not Acceptable Here" message). Case 2: First data packet received by the terminal (after sending a "SIP INVITE" message and receiving a "SIP 200 OK" message) containing a message different to the "RTCP:TBCP Talk Burst Granted" message, e.g., "404 Not Found", "SIP 486 Busy Here" or "SIP 403 Forbidden" message. Case 3: No message received by the terminal within a pre-determined time.

7.3.4.10 PoC session initiation time (on-demand) [s]

7.3.4.10.1 Abstract definition

The PoC session initiation time (on-demand) is the time period between pushing the PoC button on the terminal in order to initiate a PoC session and being granted the talk burst, e.g., indicated by a "beep"-tone on the terminal.

Remarks:

- The terminal notifies the user about the granted talk burst (e.g., by a "beep"-tone).

- There shall be at least one other participating terminal and the floor shall be idle. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- All terminals shall be registered to the PoC service and shall have successfully published their PoC service settings.
- There are different signal flows for confirmed and for unconfirmed invitations. In the confirmed case, at least one of the invited users has to accept the invitation to the PoC session in order to get the talk burst granted (see [OMA-2]). If the PoC server supports media buffering, the talk burst confirm is sent after the first received auto-answer. While automatic answer mode shall be used for the measurements, however media buffering shall not be supported. In both cases (confirmed and unconfirmed) the trigger points for the measurement are the same. Data from confirmed and unconfirmed measurements cannot be directly compared.
- This parameter is applicable to different kinds of PoC session initiations, which has an impact on the comparability of the measurement data.
- The initial "SIP INVITE" message accepted by the PoC server is an implicit talk burst request.

Figure 7-10 shows message flow for an implicit talk burst request procedure at the initiation of the PoC session (on-demand).

7.3.4.10.2 Abstract equation

$$\text{PoC session initiation time (on-demand) } [s] = \left(t_{\text{beep received}} - t_{\text{PoC button pressed}}\right)[s]$$

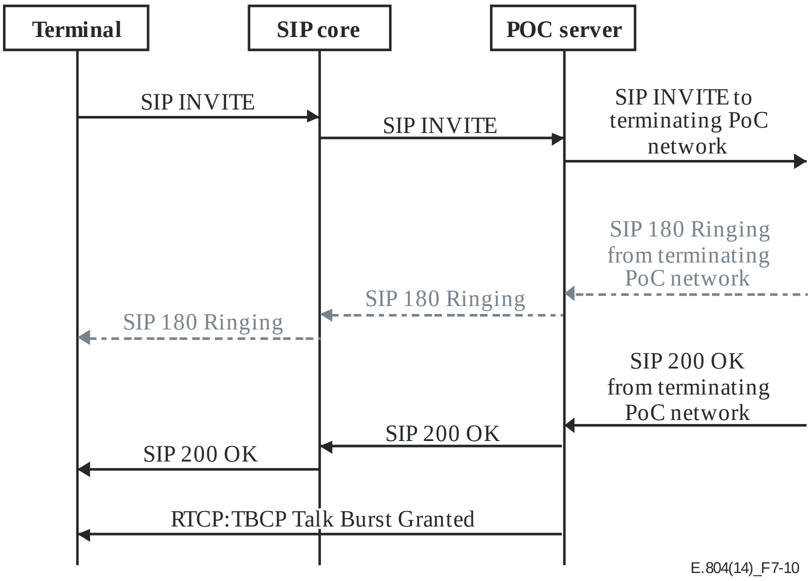


Figure 7-10 – Implicit talk burst request procedure at the initiation of the PoC session

Remark:

- The dashed arrows in Figure 7-10 only occur in case of a confirmed invitation with manual answer. In this case the time that elapses between the "SIP INVITE" message and the reception of the "SIP 200 OK" message depends on how fast an invited user on the terminating side accepts the invitation.

7.3.4.10.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{PoC\ button\ pressed}$: Time of PoC session initiation attempt	Start: Push the PoC button.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP INVITE" message.
$t_{beep\ received}$: Time of successful PoC session initiation attempt	Stop: Talk burst granted is indicated.	Stop: Protocol: RTCP:TBCP First data packet received by the terminal containing "RTCP: TBCP Talk Burst Granted".

7.3.4.11 PoC session media parameters negotiation failure ratio (pre-established) [%]

7.3.4.11.1 Abstract definition

The PoC session media parameters negotiation failure ratio (pre-established) is the probability that a negotiation procedure of media parameters for a posterior pre-established session cannot be successfully accomplished.

Remarks:

- The initial "SIP INVITE" message accepted by the PoC server is not an implicit talk burst request.
- All terminals shall be registered to the PoC service and shall have successfully published their PoC service settings.
- The PoC server performing the controlling PoC function shall determine the codec(s) and media parameters that should be used in the PoC session. The preferred media parameters should be determined according to the lowest negotiated media parameters (e.g., bandwidth) of the terminals that have joined the PoC session (see [OMA-1], page 102).
- User plane adaptation may be triggered, e.g., by roaming or when a new terminal with lower media parameters enters the PoC session (see [OMA-2], page 103).

7.3.4.11.2 Abstract equation

$$\text{PoC session media parameters negotiation failure ratio (pre-established) } [\%] = \frac{\text{unsuccessful negotiation attempts}}{\text{all negotiation attempts}} \times 100$$

7.3.4.11.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC session media parameters negotiation attempt	Start: PoC terminal initiates media parameters negotiation.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP INVITE" message with media parameters.
Successful PoC session media parameters negotiation attempt	Stop: Successful parameter negotiation indication.	Stop: Protocol: SIP. First "SIP Ack" data packet sent by the terminal after the reception of a "SIP OK" message.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Unsuccessful PoC session media parameters negotiation attempt	Stop: Media parameter negotiation is rejected or not indicated.	Stop: Protocol: SIP. Case 1: First data packet received by the terminal (after sending a "SIP INVITE" message and receiving a "SIP 100 TRYING" message) containing a message different to "SIP 200 OK"; e.g., a "SIP 403 Forbidden" or "SIP 488 Not Acceptable Here" message. Case 2: No message received by the terminal within a pre-determined time.

7.3.4.12 PoC session media parameters negotiation time (pre-established) [s]

7.3.4.12.1 Abstract definition

The PoC session media parameters negotiation time (pre-established) is the time period needed to accomplish a successful negotiation of media parameters.

Remarks:

- The initial "SIP INVITE" message accepted by the PoC server is not an implicit talk burst request.
- All terminals shall be registered to the PoC service and shall have successfully published their PoC service settings.
- The PoC server performing the controlling PoC function shall determine the codec(s) and media parameters that should be used in the PoC session. The preferred media parameters should be determined according to the lowest negotiated media parameters (e.g., bandwidth) of the terminals that have joined the PoC session (see [OMA-1], page 102).
- User plane adaptation may be triggered, e.g., by roaming or when a new terminal with lower media parameters enters the PoC session (see [OMA-1], page 103).

Figure 7-11 shows message flow for media parameters negotiation for pre-established session.

7.3.4.12.2 Abstract equation

$$\text{PoC session media parameters negotiation time (pre-established)}\,[s] = \left(t_{\text{ok received}} - t_{\text{negotiation initiation}}\right)[s]$$

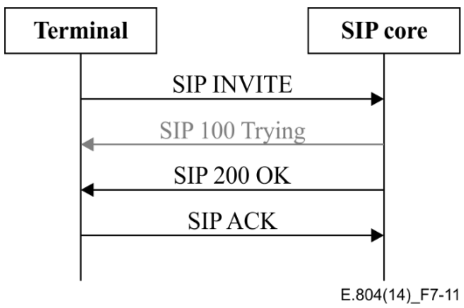


Figure 7-11 – Media parameters negotiation for pre-established session

7.3.4.12.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{negotiation\ initiation}$: Time of PoC pre-established session media parameters negotiation attempt	Start: PoC terminal initiates media parameters negotiation.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP INVITE" message with pedia parameters.
$t_{ok\ received}$: Time of successful PoC pre-established session media parameters negotiation attempt	Stop: Successful parameter negotiation indication.	Stop: Protocol: SIP. First "SIP Ack" data packet sent by the terminal after the reception of a "SIP OK" message.

7.3.4.13 PoC session initiation failure ratio (pre-established) [%]

7.3.4.13.1 Abstract definition

The PoC session initiation failure ratio (pre-established) is the probability that a pre-established session cannot be successfully initiated. After the negotiation of media parameters, a pre-established session is initiated when the user pushes the PoC button on the terminal (and thereby requests the talk burst) and is granted the talk burst.

Remarks:

- The terminal notifies the user about the granted talk burst (e.g., by a "beep"-tone).
- The initial "SIP REFER" message accepted by the PoC server is an implicit talk burst request.
- There shall be at least one other participating terminal and the floor shall be idle. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- The terminals shall have negotiated the session media parameters with the PoC server.
- All terminals in the PoC session shall be configured to use the auto-answer mode procedure (see [OMA-3]).
- There are different signal flows for confirmed and for unconfirmed invitations. In the confirmed case, at least one of the invited users has to accept the invitation to the PoC session in order to get the talk burst granted. The terminals on the terminating side may be configured to confirm the invitation automatically. This auto-answer mode should be used for measurements. In both cases (confirmed and unconfirmed) the trigger points for the measurement are the same. Data from confirmed and unconfirmed measurements cannot be directly compared.
- This parameter is applicable to different kinds of PoC session initiations, which has an impact on the comparability of the measurement data.

7.3.4.13.2 Abstract equation

$$\text{PoC session initiation failure ratio (pre-established)}\,[\%] = \frac{\text{unsuccessful pre-established session initiation attempts}}{\text{all pre-established session initiation attempts}} \times 100$$

7.3.4.13.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC session initiation attempt	Start: The PoC button is pushed.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REFER" message with the PoC session URL.
Successful PoC session initiation attempt	Stop: Talk burst granted is indicated.	Stop: Protocol: RTCP: TBCP. First data packet received by the terminal containing "Talk Burst Granted" message.
Unsuccessful PoC session initiation attempt	Stop: Missing talk burst granted indication.	Stop: Protocol: SIP; RTCP:TBCP. Case 1: First data packet received by the terminal (after sending a "SIP REFER" message) containing a message different to the "SIP 202 Accepted" message. Case 2: Data packet received by the terminal (after sending a "SIP REFER" message and receiving a "SIP 202 Accepted" message) containing a message different to "SIP NOTIFY", "RTCP:TBCP Connect" or "RTCP:TBCP Talk Burst Granted" (e.g., "SIP 404 Not Found", "SIP 486 Busy Here" or "SIP 403 Forbidden" message). Case 3: No message received by the terminal within a pre-determined time.

7.3.4.14 PoC session initiation time (pre-established) [s]

7.3.4.14.1 Abstract definition

The PoC session initiation time (pre-established) is the time period between pushing the PoC button on the terminal in order to initiate a pre-established session and being granted the talk burst, e.g., indicated by a "beep"-tone on the terminal.

Remarks:

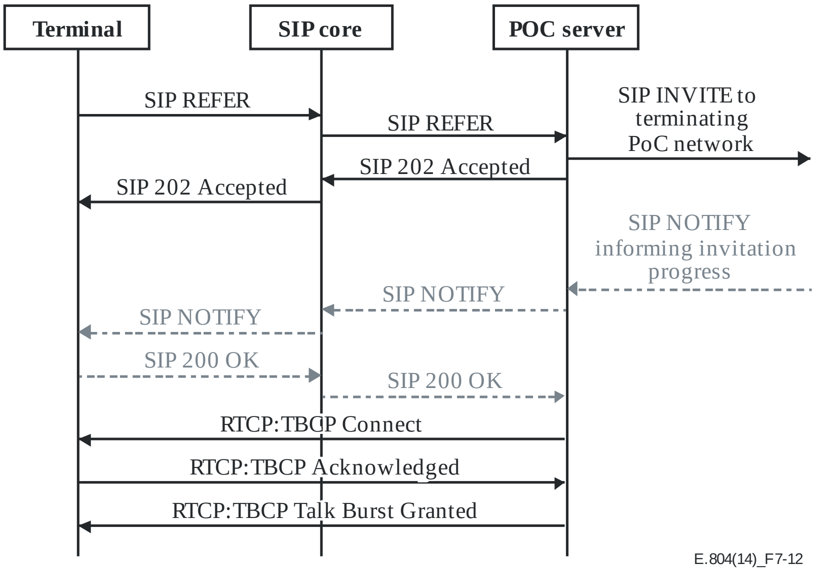
- The terminal notifies the user about the granted talk burst (e.g., by a "beep"-tone).
- The initial "SIP REFER" message accepted by the PoC server is an implicit talk burst request.
- There shall be at least one other participating terminal and the floor shall be idle. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- The terminals shall have negotiated the session media parameters with the PoC server.
- All terminals in the PoC session shall be configured to use the auto-answer mode procedure (see [OMA-3]).
- There are different signal flows for confirmed and for unconfirmed invitations. In the confirmed case, at least one of the invited users has to accept the invitation to the PoC session in order to get the talk burst granted. The terminals on the terminating side may be configured to confirm the invitation automatically. This auto-answer mode should be used for measurements. In both cases (confirmed and unconfirmed) the trigger points for the measurement are the same. Data from confirmed and unconfirmed measurements cannot be directly compared.

- This parameter is applicable to different kinds of PoC session initiations, which has an impact on the comparability of the measurement data.

Figure 7-12 shows message flow for talk burst request procedure of a pre-established PoC session.

7.3.4.14.2 Abstract equation

$$\text{PoC session initiation time (pre-established) [s]} = \left(t_{\text{beep received}} - t_{\text{PoC button pressed}}\right)[s]$$



NOTE – The dashed arrows in this figure only occur in case of a confirmed, manual answer invitation. In this case the time period between the "SIP INVITE" message and the reception of the "Talk Burst Granted" message depends on how fast an invited user on the terminating side answers to the invitation. Furthermore, the "SIP NOTIFY" message is defined as optional (see [OMA-2]) and might not be sent by the server at all. For this reason, the automatic answer mode shall be used during measurements.

Figure 7-12 – Talk burst request procedure of a pre-established PoC session

7.3.4.14.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{PoC button pressed}}$: Time of PoC session initiation attempt	Start: Push the PoC button.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REFER" message with the PoC session description.
$t_{\text{beep received}}$: Time of successful PoC session initiation attempt	Stop: Talk burst granted is indicated.	Stop: Protocol: RTCP: TBCP. First data packet received by the terminal containing "Talk Burst Granted" message.

7.3.4.15 PoC session set-up failure ratio (on-demand) [%]

7.3.4.15.1 Abstract definition

The PoC session set-up failure ratio (on-demand) is the probability that a terminal cannot successfully register to the PoC service and initialize an on-demand session.

Remarks:

- This QoS parameter is a combination of the PoC registration parameter and the PoC session initiation parameter. It has to reflect the behaviour of PoC enabled user equipment.

- Data between confirmed and unconfirmed measurements cannot be compared directly.

7.3.4.15.2 Abstract equation

Let R be the number of unsuccessful registration attempts and let S be the number of unsuccessful session initiations following a successful registration.

Then:

$$\text{PoC session set-up failure ratio (on-demand) } [\%] = \frac{R + S}{\text{all PoC session setup attempts}} \times 100$$

7.3.4.15.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC registration attempt	Start: Activation of the PoC service on the terminal.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REGISTER" message.
Successful PoC session initiation attempt	Stop: PoC available is indicated.	Stop: Protocol: RTCP:TBCP First data packet received by the terminal containing "RTCP: TBCP Talk Burst Granted".
Unsuccessful PoC session initiation attempt	Stop: Missing talk burst granted indication.	Stop: Protocol: SIP; RTCP: TBCP. Case 1: First data packet received by the terminal (after sending a "SIP INVITE" message) containing an error message or redirection message (e.g., a "403 Forbidden" or "488 Not Acceptable Here" message). Case 2: First data packet received by the terminal (after sending a "SIP INVITE" message and receiving a "SIP 200 OK" message) containing a message different from the "RTCP:TBCP Talk Burst Granted" message, e.g., "404 Not Found", "SIP 486 Busy Here" or "SIP 403 Forbidden" message. Case 3: No message received by the terminal within a pre-determined time.

7.3.4.16 PoC session set-up failure ratio (pre-established) [%]

7.3.4.16.1 Abstract definition

The PoC session set-up failure ratio (pre-established) is the probability that a terminal cannot successfully register to the PoC service and initialize a pre-established session.

Remarks:

- This QoS parameter is a combination of the PoC registration parameter and the PoC session initiation parameter. It has to reflect the behaviour of PoC enabled user equipment.
- Data between confirmed and unconfirmed measurements cannot be compared directly.

7.3.4.16.2 Abstract equation

Let R be the number of unsuccessful registration attempts and let S be the number of unsuccessful pre-established session media parameters negotiations following a successful registration. Let T be the number of unsuccessful session initiation attempts, which followed after a successful registration and after a successful pre-established session media parameters negotiation.

Then:

$$\text{PoC session setup failure ratio (pre-established)}\,[\%] = \frac{\text{R} + \text{S} + \text{T}}{\text{all PoC session setup attempts}} \times 100$$

7.3.4.16.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC registration attempt	Start: Activation of the PoC service on the terminal.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REGISTER" message.
Successful PoC session initiation attempt	Stop: PoC available is indicated.	Stop: Protocol: RTCP:TBCP First data packet received by the terminal containing "RTCP:TBCP Talk Burst Granted".
Unsuccessful PoC session initiation attempt	Stop: Missing talk burst granted indication.	Stop: Protocol: SIP; RTCP: TBCP. Case 1: First data packet received by the terminal (after sending a "SIP REFER" message) containing a message different to the "SIP 202 Accepted" message. Case 2: Data packet received by the terminal (after sending a "SIP REFER" message and receiving a "SIP 202 Accepted" message) containing a message different from "SIP NOTIFY", "RTCP:TBCP Connect" or "RTCP:TBCP Talk Burst Granted" (e.g., "SIP 404 Not Found", "SIP 486 Busy Here" or "SIP 403 Forbidden" message). Case 3: No message received by the terminal within a pre-determined time.

7.3.4.17 PoC session set-up time [s]

7.3.4.17.1 Abstract definition:

The PoC session set-up time is the time period for the registration to the PoC service plus the time period for the initiation of a PoC session.

Remarks:

- This QoS parameter is a combination of the PoC registration parameter and the PoC session initiation parameter. It has to reflect the behaviour of PoC enabled user equipment.
- Data between confirmed and unconfirmed measurements cannot be compared directly.
- Data between on-demand sessions and pre-established sessions cannot be compared directly.

Figure 7-13 shows message flow for PoC session set-up time and PoC push to speak time.

7.3.4.17.2 Abstract equation

$$\text{PoC session setup time}\,[s] = \left(t_{\text{beep received}} - t_{\text{PoCActivated}}\right)[s]$$

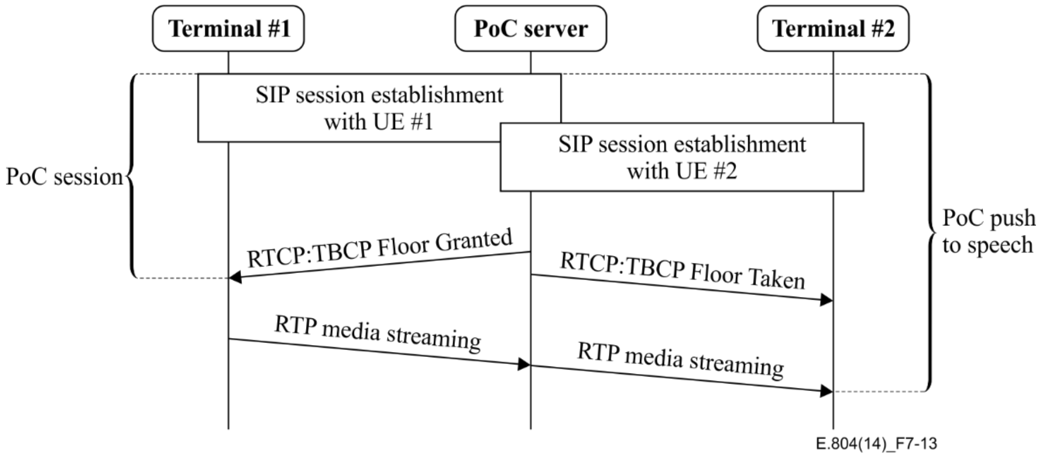


Figure 7-13 – PoC session set-up time and PoC push to speak time

7.3.4.17.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{PoCActivated}}$: Time of PoC registration attempt	Start: Activation of the PoC service on the terminal.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REGISTER" message.
$t_{\text{beep received}}$: Time of successful PoC registration attempt	Stop: PoC available is indicated.	Stop: Protocol: RTCP:TBCP First data packet received by the terminal containing "RTCP:TBCP Talk Burst Granted".

7.3.4.18 PoC push to speak failure ratio [%]

7.3.4.18.1 Abstract definition

The PoC push to speak failure ratio is the probability that terminal A cannot successfully set up a PoC session and start with speech leading to no other terminal receiving the speech.

Remarks:

- This QoS parameter is a combination of the PoC session set-up parameter and the PoC talk burst cut-off ratio parameter (see clause 7.3.4.30). It is ought to reflect the behaviour of PoC enabled user equipment.
- All terminals shall be registered to the PoC service and shall have successfully published their PoC service settings.
- Data between confirmed and unconfirmed measurements cannot be compared directly.

7.3.4.18.2 Abstract equation

Let S be the number of unsuccessful PoC session set-up attempts and let T be the number of talk burst cut-offs following a successful PoC session set-up.

Then:

$$\text{PoC push to speak failure ratio}[\%] = \frac{S+T}{\text{all PoC push to speak attempts}} \times 100$$

7.3.4.18.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC registration attempt	Start: Activation of the PoC service on the terminal.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REGISTER" message.
No unintended speech cut-off on terminal B	Stop: Sound received by terminal B.	Stop: Protocol: RTP. First data packet received by terminal B containing the speech data.
Unintended speech cut-off on terminal B	Stop: Terminal B does not receive the speech or does not receive the whole speech.	Stop: Protocol: RTP. Case 1: No packet containing speech data (RTP media stream) received by terminal B within a pre-determined time. The timeout should be chosen as greater than the average speech delay (see clause 7.3.4.32). Case 2: The media stream is only partially received by terminal B. Some of the data packets containing the speech data (RTP media stream) have not been received by terminal B.

7.3.4.19 PoC push to speak time [s]

7.3.4.19.1 Abstract definition

The PoC push to speak time is the period of time that it takes to set up a PoC session and start with speech in addition to the delay until terminal B receives the speech (as defined in clause 7.3.4.32).

Remarks:

- This QoS parameter is a combination of the PoC session set-up time parameter and the PoC speech transmission delay parameter (see clause 7.3.4.32). It has to reflect the behaviour of PoC enabled user equipment.
- All terminals shall be registered to the PoC service and shall have successfully published their PoC service settings.
- Data between confirmed and unconfirmed measurements cannot be compared directly.

7.3.4.19.2 Abstract equation

$$\text{PoC push to speak time}[s] = \left(t_{B\_hears} - t_{PoCActivated}\right)[s]$$

7.3.4.19.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{PoCActivated}$: Time of PoC registration attempt	Start: Activation of the PoC service on the terminal.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REGISTER" message.
t_{B_hears}: Time of output at terminal B	Stop: Sound received by terminal B.	Stop: Protocol: RTP. First data packet received by terminal B containing speech data.

7.3.4.20 PoC session leaving failure ratio (on-demand) [%]

7.3.4.20.1 Abstract definition

The PoC session leaving failure ratio (on-demand) is the probability that the user cannot leave the PoC session which is participating in.

Remarks:

- When a PoC session is left (terminated), the terminal is still registered to the PoC service.
- PoC enabled user equipment may not give the user the possibility to leave a PoC session explicitly. The PoC session leave request may only be sent when the terminal deregisters from the PoC service.
- The terminal shall be registered to the PoC service participating in a PoC session.

7.3.4.20.2 Abstract equation

$$\text{PoC session leaving failure ratio (on-demand)}\,[\%] = \frac{\text{unsuccessful PoC session leaving attempts}}{\text{all PoC session leaving attempts}} \times 100$$

7.3.4.20.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC session leaving attempt	Start: Leaving the participating PoC session.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP BYE" message.
Successful PoC session leaving attempt	Stop: PoC session left is indicated.	Stop: Protocol: SIP. First data packet received by the terminal containing a "SIP 200 OK" message.
Unsuccessful PoC session leaving attempt	Stop: Terminal is still connected to the PoC session.	Stop: Protocol: SIP. Case 1: First data packet received by the terminal (after sending the "SIP BYE" message) containing a message different from "SIP 200 OK". Case 2: No message received by the terminal within a pre-determined time.

7.3.4.21 PoC session leaving time (on-demand) [s]

7.3.4.21.1 Abstract definition

The PoC session leaving time (on-demand) is the time period between sending the on-demand session leaving request and being disconnected from the on-demand session.

Remarks:

- When a PoC session is left (terminated), the terminal is still registered to the PoC service.
- PoC enabled user equipment may not give the user the possibility to leave a PoC session explicitly. The PoC session leave request may only be sent when the terminal de-registers from the PoC service.
- The terminal shall be registered to the PoC service participating in a PoC session.

Figure 7-14 shows message flow for a successful PoC session leaving time.

7.3.4.21.2 Abstract equation

$$\text{PoC session leaving time (on-demand)}\,[s] = \left(t_{session\,left} - t_{session\,leave\,request}\right)[s]$$

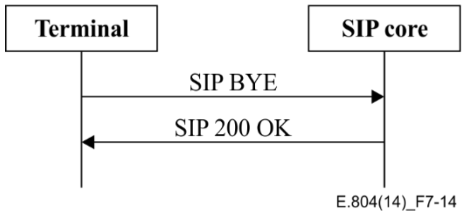


Figure 7-14 – Successful PoC session leaving

7.3.4.21.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{session\ leave\ request}$: Time of PoC session leaving attempt	Start: Leaving the participating PoC session.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP BYE" message.
$t_{session\ left}$: Time of successful PoC session leaving attempt	Stop: PoC session left is indicated.	Stop: Protocol: SIP. First data packet received by the terminal containing a "SIP 200 OK" message.

7.3.4.22 PoC session leaving failure ratio (pre-established) [%]

7.3.4.22.1 Abstract definition

The PoC session leaving failure ratio (pre-established) is the probability that the user cannot leave the PoC pre-established session which is participating in.

Remarks:

- The PoC session was established using pre-established signalling.
- The terminal may not give the user the possibility to leave a PoC session explicitly. The PoC session leave request may only be sent when the terminal deregisters from the PoC service.

7.3.4.22.2 Abstract equation

$$\text{PoC session leaving failure ratio (pre-established)}\,[\%] = \frac{\text{unsuccessful PoC session leaving attempts}}{\text{all PoC session leaving attempts}} \times 100$$

7.3.4.22.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC session leaving attempt	Start: Leaving the participating PoC session.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REFER BYE" message.
Successful PoC session leaving attempt	Stop: Terminal has successfully left the PoC session.	Stop: Protocol: SIP. First data packet received by the terminal containing a "SIP 202 ACCEPTED" message.
Unsuccessful PoC session leaving attempt	Stop: Terminal is still connected to the PoC session.	Stop: Protocol: SIP Case 1: First data packet received by the terminal (after sending the "SIP REFER BYE" message) containing a message different from "SIP 202 Accepted". Case 3: No message received by the terminal within a pre-determined time.

7.3.4.23 PoC session leaving time (pre-established) [s]

7.3.4.23.1 Abstract definition

The PoC session leaving time (pre-established) is the time period between sending the PoC session leaving request and being disconnected from the pre-established session.

Remarks:

- The PoC session was established using pre-established signalling.
- The terminal may not give the user the possibility to leave a PoC session explicitly. The PoC session leave request may only be sent when the terminal deregisters from the PoC service.

Figure 7-15 shows message flow for a successful PoC session leaving (pre-established session).

7.3.4.23.2 Abstract equation

$$\text{PoC session leaving time (pre-established)}\,[s] = \left(t_{session\,left} - t_{session\,leave\,request}\right)[s]$$

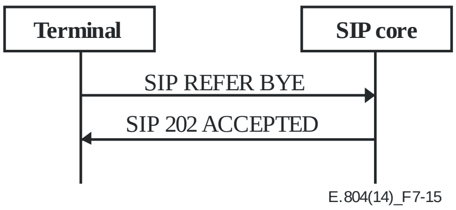


Figure 7-15 – Successful PoC session leaving (pre-established session)

7.3.4.23.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{session\ leave\ request}$: Time of PoC session leaving attempt	Start: Leaving the participating PoC session.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP REFER BYE" message.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{session\ left}$: Time of successful PoC session leaving attempt	Stop: Terminal has successfully left the PoC session.	Stop: Protocol: SIP. First data packet received by the terminal containing a "SIP 202 ACCEPTED" message.

7.3.4.24 PoC deregistration failure ratio [%]

7.3.4.24.1 Abstract definition

The PoC deregistration failure ratio is the probability that the user cannot be deregistered from the PoC service when requested.

Remark:

- The terminal shall be registered to the PoC service.

7.3.4.24.2 Abstract equation

$$\text{PoC deregistration failure ratio}\,[\%] = \frac{\text{unsuccessful PoC deregistration attempts}}{\text{all PoC deregistration attempts}} \times 100$$

7.3.4.24.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC deregistration attempt	Start: Deactivation of the PoC service on the terminal.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP Register" message, where the "Expires" header is set to 0.
Successful PoC deregistration attempt	Stop: PoC unavailable is indicated.	Stop: Protocol: SIP. First data packet received by the terminal containing a "SIP 200 OK" message.
Unsuccessful PoC deregistration attempt	Stop: PoC unavailable indication is not given within a predetermined time.	Stop: Protocol: SIP. Case 1: First data packet received by the terminal (after sending the second "SIP REGISTER" message) containing a message different from "SIP 200 OK". Case 2: No message received by the terminal within a pre-determined time.

7.3.4.25 PoC deregistration time [s]

7.3.4.25.1 Abstract definition

The PoC deregistration time is the time period between the deregistration request and the successful deregistration from the PoC service.

Remark:

- The terminal shall be registered to the PoC service.

Figure 7-16 shows message flow for successful PoC deregistration.

7.3.4.25.2 Abstract equation

$$\text{PoC deregistration time}\,[s] = \left(t_{\text{PoC deregistered}} - t_{\text{deregistration request}}\right)[s]$$

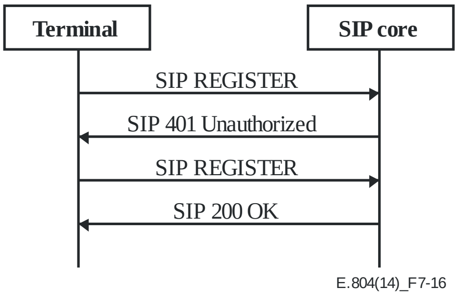


Figure 7-16 – Successful PoC deregistration example

7.3.4.25.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{deregistration request}}$: Time of PoC deregistration attempt	Start: Deactivation of the PoC service on the terminal.	Start: Protocol: SIP. First data packet sent by the terminal containing a "SIP Register" message, where the "Expires" header is set to 0.
$t_{\text{PoC deregistered}}$: Time of successful PoC deregistration attempt	Stop: PoC unavailable is indicated.	Stop: Protocol: SIP. First data packet received by the terminal containing a "SIP 200 OK" message.

7.3.4.26 PoC busy floor response failure ratio [%]

7.3.4.26.1 Abstract definition

The PoC busy floor response failure ratio is the probability that, once in a PoC session, the talk burst request from the terminal fails.

Remarks:

- The terminal shall be within an active PoC talk session. Thus, there shall be at least one other participating terminal.
- For the special case of requesting the idle floor, refer to QoS parameters defined in clauses 7.3.4.28 and 7.3.4.29.

7.3.4.26.2 Abstract equation

$$\text{PoC busy floor response failure ratio [\%]} = \frac{\text{unsuccessful talk burst requests}}{\text{all talk burst requests}} \times 100$$

7.3.4.26.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC talk burst request	Start: Push the PoC button.	Start: Protocol: RTCP: TBCP. First data packet sent by the terminal containing a "RTCP: TBCP Talk Burst Request" message.
Successful PoC talk burst request	Stop: Current floor state is indicated.	Stop: Protocol: RTCP: TBCP. First data packet received by the terminal containing information about the floor state.
Unsuccessful PoC talk burst request	Stop: No talk burst response is indicated (e.g., grant, queued).	Stop: Protocol: RTCP: TBCP. No message received by the terminal within a pre-determined time.

7.3.4.27 PoC busy floor response time [s]

7.3.4.27.1 Abstract definition

The PoC busy floor response time is the time period between requesting the talk burst and receiving the indication that the floor is busy within an already established PoC session.

Remarks:

- The terminal shall be within an active PoC talk session. Thus, there shall be at least one other participating terminal.
- For the special case of requesting the idle floor, refer to QoS parameters defined in clauses 7.3.4.28 and 7.3.4.29.

Figure 7-17 shows message flow for an example of a busy floor response.

7.3.4.27.2 Abstract equation

$$\text{PoC busy floor response time}\,[s] = \left(t_{\text{floor response}} - t_{\text{floor request}}\right)[s]$$

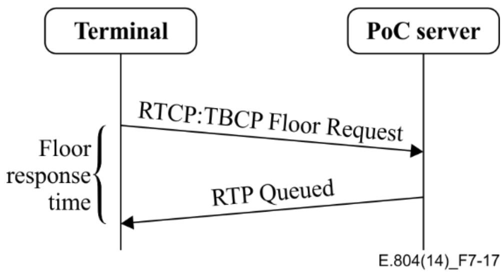


Figure 7-17 – Example of a busy floor response

7.3.4.27.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{floor\ request}$: Time of PoC talk burst request	Start: Push the PoC button.	Start: Protocol: RTCP: TBCP. First data packet sent by the terminal containing a "RTCP: TBCP Talk Burst Request" message.
$t_{floor\ response}$: Time of successful PoC talk burst request	Stop: Current floor state is indicated.	Stop: Protocol: RTCP: TBCP. First data packet received by the terminal containing information about the floor state.

7.3.4.28 PoC talk burst request failure ratio [%]

7.3.4.28.1 Abstract definition

The PoC talk burst request failure ratio is the probability that, once in a PoC session, the terminal's request of the idle floor fails.

Remarks:

- The terminal shall be within an active PoC session.
- There shall be at least one other participating terminal and the floor shall be idle. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- This parameter is defined explicitly because the server's response time and failure ratio to a request of the idle floor may be different to the response time and response failure ratio of a busy floor.

7.3.4.28.2 Abstract equation

$$\text{PoC talk burst request failure ratio}\,[\%] = \frac{\text{unsuccessful talk burst requests}}{\text{all talk burst requests}} \times 100$$

7.3.4.28.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC talk burst request	Start: Push the PoC button.	Start: Protocol: RTCP: TBCP. First data packet sent by the terminal containing a "RTCP: TBCP Talk Burst Request" message.
Successful PoC talk burst request	Stop: Talk burst granted is indicated.	Stop: Protocol: RTCP: TBCP. First data packet received by the terminal containing a "RTCP: TBCP Talk Burst Granted" message.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Unsuccessful PoC talk burst request	Stop: Talk burst granted is not indicated.	Stop: Protocol: RTCP: TBCP. Case 1: First data packet received by the terminal containing a floor state different from "RTCP: TBCP Talk Burst Granted" message. Possible floor states are listed in [OMA-1]. Case 2: No message received by the terminal within a predetermined time.

7.3.4.29 PoC talk burst request time [s]

7.3.4.29.1 Abstract definition

The PoC talk burst request time is the time period between requesting the talk burst and being granted the previously idle floor within an already established PoC session.

Remarks:

- The terminal shall be within an active PoC session.
- There shall be at least one other participating terminal and the floor shall be idle. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- This parameter is defined explicitly because the server's response time and failure ratio to a request of the idle floor may be different from the response time and response failure ratio of a busy floor.

Figure 7-18 shows message flow for an example of a successful talk burst request.

7.3.4.29.2 Abstract equation

$$\text{PoC talk burst request time}\,[s] = \left(t_{\text{floor granted}} - t_{\text{floor request}}\right)[s]$$

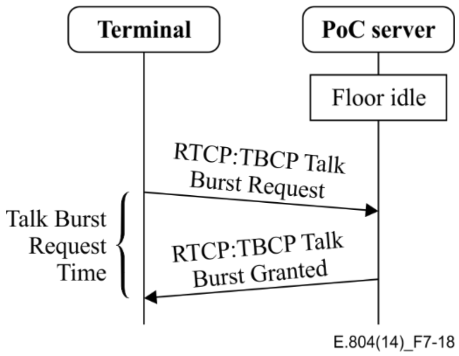


Figure 7-18 – Example of a successful talk burst request

7.3.4.29.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{floor\ request}$: Time of PoC talk burst request	Start: Push the PoC button.	Start: Protocol: RTCP: TBCP. First data packet sent by the terminal containing a "RTCP: TBCP Talk Burst Request" message.
$t_{floor\ granted}$: Time of successful PoC talk burst request	Stop: Talk burst granted is indicated.	Stop: Protocol: RTCP: TBCP. First data packet received by the terminal containing a "RTCP: TBCP Talk Burst Granted" message.

7.3.4.30 PoC talk burst cut-off ratio [%]

7.3.4.30.1 Abstract definition

The PoC talk burst cut-off ratio is the probability that the terminal on the originating side (terminal A) has the floor and creates and sends data packets containing speech data (RTP media stream), but the stream does not arrive (or arrives partially) at the terminating side (terminal B).

Remarks:

- There shall be at least one other active participating terminal and the floor shall be granted to terminal A. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- The implementation of a stop-talking timer is mandatory on the server side. When a user is granted a talk burst, the PoC server resets this stop-talking timer. When the timer expires, the PoC server revokes the talk burst from the user (see [OMA-1]). Hence this situation (talk burst revoked because of a timeout) shall not be considered for measurements.
- The time of a talk burst shall be shorter than the network-defined stop-talking timeout.

Figure 7-19 shows message flow for PoC talk burst cut-off.

7.3.4.30.2 Abstract equation

$$\text{PoC talk burst cut-off ratio}\,[\%] = \frac{\text{dropped talk bursts}}{\text{all talk bursts}} \times 100$$

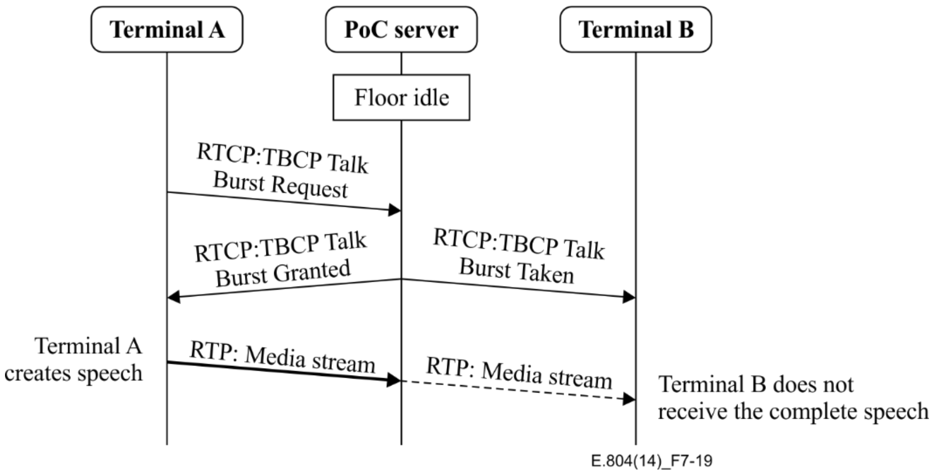


Figure 7-19 – PoC talk burst cut-off

7.3.4.30.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC talk burst granted and start of speech on terminal A	Start: Talk burst granted is indicated. Speech starts.	Start: Protocol: RTP. First data packet sent by terminal A containing speech data.
No unintended speech cut-off on terminal B	Stop: Sound received by terminal B.	Stop: Protocol: RTP. First data packet received by terminal B containing speech data.
Unintended speech cut-off on terminal B	Stop: Terminal B does not receive speech or does not receive the whole speech.	Stop: Protocol: RTP. Case 1: No packet containing speech data (RTP media stream) received by terminal B within a pre-determined time. The timeout should be chosen greater than the average speech delay (see clause 7.3.4.32). Case 2: The media stream is only partially received by terminal B. Some of the data packets containing speech data (RTP media stream) have not been received by terminal B.

7.3.4.31 PoC talk burst packet drop ratio [%]

7.3.4.31.1 Abstract definition

The PoC talk burst packet drop ratio is the ratio between the number of data packets containing speech data sent by the terminal on the originating side (terminal A) and the number of data packets containing speech data received on the terminating side (terminal B).

Remarks:

- There shall be at least one other active participating terminal and the floor shall be granted to terminal A. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- The implementation of a stop-talking timer is mandatory on the server side. When a user is granted a talk burst, the PoC server resets this stop-talking timer. When the timer expires, the PoC server revokes the talk burst from the user (see [OMA-1]). Hence this situation (talk burst revoked because of a timeout) shall not be considered for measurements.
- The time of a talk burst shall be shorter than the network-defined stop-talking timeout.

This ratio shall get calculated on a per-burst basis.

7.3.4.31.2 Abstract equation

$$\text{PoC talk burst packet drop ratio}\,[\%] = \frac{\text{dropped RTP speech packets}}{\text{all sent RTP speech packets}} \times 100$$

7.3.4.31.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
PoC talk burst granted and start of speech on terminal A	Start: Talk burst granted is indicated. Speech starts.	Stop: Protocol: RTP. First data packet sent by terminal A containing speech data.
End of speech on terminal B	Stop: End of speech is indicated or timeout occurred after terminal B has received speech.	Stop: Protocol: RTP. Case 1: First packet received by the terminal containing a "RTP: Last Packet" message after a data packet containing speech data has been received by terminal B. Case 2: No packet containing a "RTP: Last Packet" message received by terminal B within a pre-determined time after a data packet containing speech data has been received by terminal B.

7.3.4.32 PoC voice transmission delay (first) [s]

7.3.4.32.1 Abstract definition

The parameter PoC speech transmission delay (first) describes the period of time between a terminal sending speech data (RTP media stream) and the first terminal receiving the speech data for the first talk burst after a PoC session has been established successfully.

Remarks:

- Without loss of generality, the PoC session consists only of two active terminals (A and B) and terminal A is trying to create and send data packets containing speech data (RTP media stream). Thus, terminal B is the one which should receive the corresponding RTP media stream.
- Server side buffering has a high impact on measurement results. Depending on the configuration of the server, the PoC speech transmission delay (first) might in fact just describe the transmission delay between the server and terminal B. To avoid buffering at the server side, confirmed indication shall be used.
- Terminal A shall create an RTP media stream immediately after being granted the talk burst.
- This parameter is measured on the transport layer. Thus, the measured value may be smaller than the real user perceived speech delay. The perceived delay also depends on the encoding/decoding speed of the terminals.

Figure 7-20 shows message flow for PoC speech transmission delay (first) and PoC speech transmission delay (others).

7.3.4.32.2 Abstract equation

$$\text{PoC voice transmission delay (first)}[s] = (t_{B\_hears} - t_{A\_speaks})[s]$$

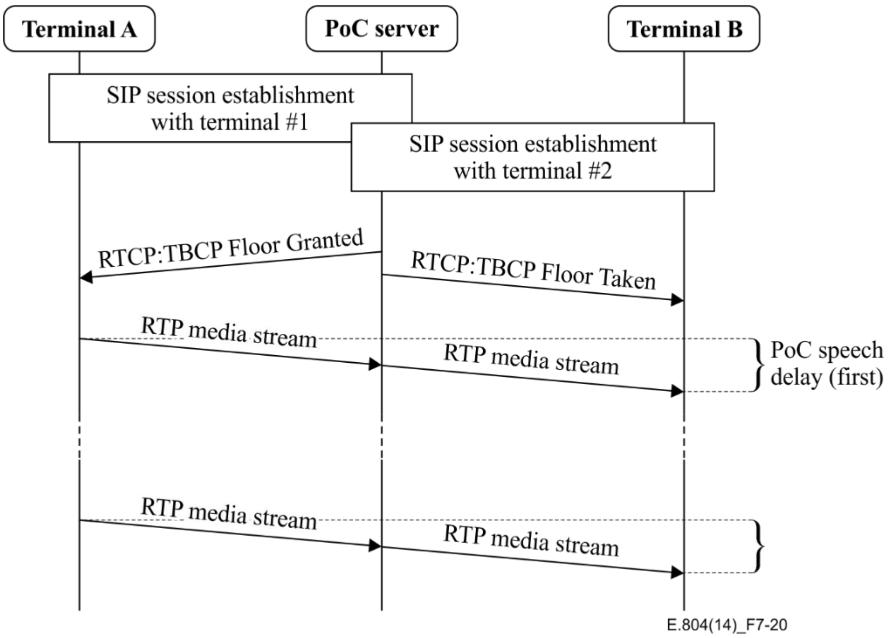


Figure 7-20 – PoC speech transmission delay (first) and PoC speech transmission delay (others)

7.3.4.32.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t_{A_speaks}: Time of input at terminal A	Start: Terminal A had the talk burst granted and then creates an RTP media stream (starts talking).	Start: Protocol: RTP. First data packet sent by terminal A containing the speech data.
t_{B_hears}: Time of output at terminal B	Stop: Sound received by terminal B.	Stop: Protocol: RTP. First data packet received by terminal B containing the speech data.

7.3.4.33 PoC speech transmission delay (others) [s]

7.3.4.33.1 Abstract definition

The parameter PoC speech transmission delay (others) is the period of time between a terminal sending speech data (RTP media stream) and the first terminal receiving the speech data (within an already established PoC session).

Remarks:

- Without loss of generality, the PoC session consists only of two active terminals (A and B) and terminal A is trying to create and send data packets containing speech data (RTP media stream). Thus, terminal B is the one which should receive the corresponding RTP media stream.
- Server side buffering has a high impact on measurement results. Depending on the configuration of the server, the PoC speech transmission delay (first) might in fact just describe the transmission delay between the server and terminal B. To avoid buffering at server side, confirmed indication shall be used.
- Terminal A shall create an RTP media stream immediately after being granted the talk burst.

- This parameter is measured on the transport layer. Thus the measured value may be smaller than the real user perceived speech delay. The perceived delay also depends on the encoding/decoding speed of the terminals.
- The speech delays on the terminating site depend on where the terminals are located (e.g., in another cell or another network).

7.3.4.33.2 Abstract equation

$$\text{PoC voice transmission delay (others)}[s] = \left(t_{B\_hears} - t_{A\_speaks}\right)[s]$$

7.3.4.33.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t_{A_speaks}: Time of input at terminal A	Start: Terminal A had the talk burst granted and then creates an RTP media stream (starts talking).	Start: Protocol: RTP. First data packet sent by terminal A containing speech data.
t_{B_hears}: Time of output at terminal B	Stop: Sound received at terminal B.	Stop: Protocol: RTP. First data packet received by terminal B containing speech data.

7.3.4.34 PoC speech quality

To be defined.

7.3.4.35 Group management QoS parameter

To be defined.

7.3.4.36 Group document related QoS parameter

To be defined.

7.3.4.37 Instant message QoS parameter

To be defined.

7.3.5 Streaming video

7.3.5.1 Definitions

7.3.5.1.1 Streaming session

Specification [IETF RFC 2326] defines a session as "a complete RTSP transaction", e.g., the viewing of a movie. A session typically consists of a client setting up a transport mechanism for the continuous media stream (SETUP), starting the stream with PLAY or RECORD, and closing the stream with TEARDOWN".

Referring to Figure 7-21, this means that the session starts at (B) and stops at (G).

7.3.5.2 Prerequisites

Precondition	Covered by	Reference document	Comment
Network accessibility given	Network accessibility indicator		
PDP context activated			

7.3.5.3 Streaming scenarios

The following two clauses describe the streaming scenario as a generic approach in order to understand the main principles and identify the relevant protocols and communication procedures.

7.3.5.3.1 Generic streaming signalling flow

A generic signal flow description for streaming is shown in Figure 7-21. The client communicates with the web server and media server entities and uses different protocols during the complete procedure, e.g., RTP, RTSP, RTCP, HTTP, etc.

The table below gives a basic description of the protocols and their usage.

Protocol	Reference in Figure 7-21	Description
HTTP	A	HTTP is used for the retrieval of the streaming file description data.
RTSP	B, C, F, G	RTSP is an application-level protocol. It provides different methods for the control of real time data, e.g., audio/video (see Note 1).
RTP	D	RTP is used for the transmission of real time data, e.g., audio/video (see Note 2).
RTCP	E	RTCP is the control protocol for RTP. Its main function is the provision of a quality feedback.
NOTE 1 – RTSP is not responsible for the delivery of the data, this is done by RTP. NOTE 2 – RTP is only used for the delivery of the data. No control and/or QoS are included.		

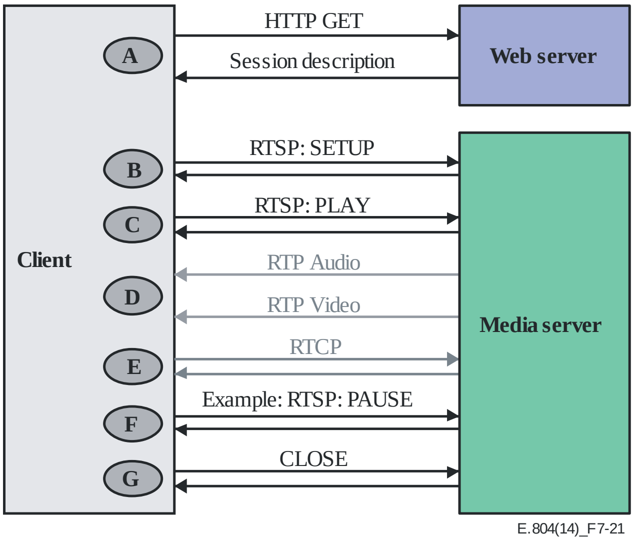


Figure 7-21 – Generic session signalling flow, based on [b-Schulzrinne]

Referring to Figure 7-21 and the definition of a session in clause 7.3.5.1.1, it is possible to divide the communication of the client with the server side in two phases:

- In the first phase, the client communicates with the web server in order to get a description of the file to be streamed. The used protocol is HTTP. Starting point is (A) and ending point is (B).
- In the second phase, the client starts the communication with the media server which is finally delivering the stream. This means that the session starts at (B) and stops at (G). Different protocols are used in this phase (RTSP, RTP, RTCP, etc.).

7.3.5.3.2 Parameter overview chart

Figure 7-22 gives an overview of the defined QoS parameters with their trigger points from user's point of view.

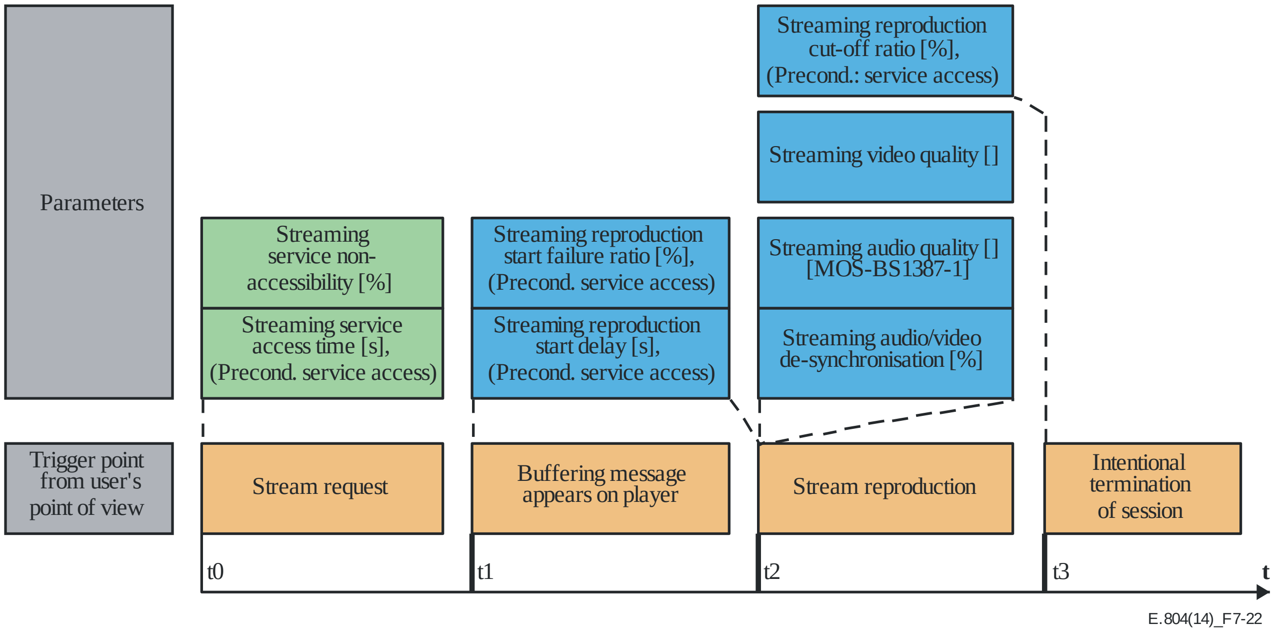


Figure 7-22 – Parameter overview with trigger points

7.3.5.4 Streaming service non-accessibility [%]

7.3.5.4.1 Abstract definition

The parameter streaming service non-accessibility is the probability that the first data packet of the stream cannot be received by the UE when requested by the user. The "packet reception" is completed by appearance of the "buffering" message on the player at the user side.

The first data packet refers to the RTP protocol.

7.3.5.4.2 Abstract equation

$$\text{Streaming service non-accessibility}\,[\%] = \frac{\text{unsuccessful stream request attempts}}{\text{all stream request attempts}} \times 100$$

7.3.5.4.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Service access attempt	Start: Stream request.	Start: • WAP 1.x, WAP 2.x: WSP Disconnect; • WAP 2.x: TCP SYN towards streaming platform.
Successful attempt	Stop: "Buffering" message.	Stop: Reception of first data packet.
Unsuccessful attempt	Stop trigger point not reached.	

7.3.5.5 Streaming service access time [s]

7.3.5.5.1 Abstract definition

The parameter streaming service access time describes the duration of a service access from requesting the stream at the portal until the reception of the first stream data packet at the UE.

The first data packet refers to RTP protocol.

7.3.5.5.2 Abstract equation

$$\text{Streaming service access time}\,[s] = \left(t_{\text{reception of first data packet}} - t_{\text{stream request}}\right)[s]$$

7.3.5.5.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{stream request}}$: Time when the stream is requested	Start: Stream request.	Start: • WAP 1.x, WAP 2.x: WSP Disconnect; • WAP 2.x: TCP SYN towards streaming platform.
$t_{\text{reception of first data packet}}$: Time when the first data packet is received	Start: "Buffering" message.	Stop: Reception of the first data packet.

7.3.5.6 Streaming reproduction cut-off ratio [%]

7.3.5.6.1 Abstract definition

The parameter streaming reproduction cut-off ratio is the probability that a successfully started stream reproduction is ended by a cause other than the intentional termination by the user.

7.3.5.6.2 Abstract equation

$$\text{Streaming reproduction cut-off ratio}\,[\%] = \frac{\text{unintentionally terminated stream reproductions}}{\text{all successfully started stream reproductions}} \times 100$$

7.3.5.6.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started media streaming reproduction	Start: Stream reproduction starts.	Start: Streaming player signals the start of the stream reproduction.
Intentionally terminated streaming reproduction	Stop: User presses the "Exit" button or end of stream is reached.	Stop: RTSP teardown method sent by UE and reception of confirmation "RTSP 200 OK" from media server.
Unintentionally terminated streaming reproduction	Stop trigger point not reached.	
NOTE – Not all players may signal the reproduction start.		

Some players do not send this TEARDOWN command at the end of the stream but a PAUSE command or in some cases nothing is sent at all. On the server side, a logic can then identify the status of the streams/clients.

Used players should send the RTSP: TEARDOWN command in order to give a stable trigger point for measurements.

7.3.5.7 Streaming audio quality

7.3.5.7.1 Abstract definition

The parameter streaming audio quality is the audio quality as perceived by the end user. Since the streams can contain not only speech information, an algorithm like [ITU-T P.862] is not suitable for all scenarios.

ITU-R has defined an algorithm defined for audio information which can be found in [OMA-3].

7.3.5.7.2 Abstract equation

To be defined.

7.3.5.7.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Tbd	Start: Beginning of audio stream reproduction.	Start: Streaming players signal when the reproduction of the stream starts.
Tbd	Stop: End of audio stream reproduction.	Stop: RTSP: TEARDOWN.

7.3.5.8 Streaming video quality

7.3.5.8.1 Abstract definition

The parameter streaming video quality measures the quality of the video stream.

NOTE – Although evaluation algorithms exist, there are no standardized solutions yet.

7.3.5.8.2 Abstract equation

NOTE – Although evaluation algorithms exist, there are no standardized solutions yet.

7.3.5.8.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Tbd	Start: Begin of video stream reproduction.	Start: Streaming players signal when the reproduction of the stream starts.
Tbd	Stop: End of video stream reproduction.	Stop: RTSP: TEARDOWN.

7.3.5.9 Streaming audio/video de-synchronization

7.3.5.9.1 Abstract definition

The parameter streaming audio/video de-synchronization is the percentage of times that time difference of the audio and video signal at the user side exceeds a predefined threshold.

7.3.5.9.2 Abstract equation

No validated or standardized algorithm has been selected for the evaluation of the video streaming content quality.

7.3.5.9.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Tbd	Start: Beginning of audio stream reproduction.	Start: Streaming players signal when the reproduction of the stream starts.
Tbd	Stop: End of audio stream reproduction.	Stop: RTSP: TEARDOWN.

7.3.5.10 Streaming reproduction start failure ratio [%]

7.3.5.10.1 Abstract definition

The parameter streaming reproduction start failure ratio is the probability of unsuccessful stream reproduction.

NOTE – This parameter can be affected:

- by the player;
- by the UE performance.

7.3.5.10.2 Abstract equation

$$\text{Streaming reproduction start failure ratio}\,[\%] = \frac{\text{reproduction failures}}{\text{all successful service accesses}} \times 100$$

7.3.5.10.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Service access attempt	Start: "Buffering" message.	Start: Reception of the first data packet.
Successful reproduction	Stop: Stream reproduction.	Stop: Streaming players signal when the reproduction of the stream starts.
Unsuccessful reproduction	Stop trigger point not reached.	

7.3.5.11 Streaming reproduction start delay [s]

7.3.5.11.1 Abstract definition

The parameter streaming reproduction start delay is the duration between the reception at UE of the first stream data packet and the start of the reproduction of the stream on the UE.

NOTE – This parameter can be affected:

- by the player;
- by the UE performance.

7.3.5.11.2 Abstract equation

$$\text{Streaming reproduction start delay}\,[\text{s}] = \left(t_{\text{start of stream reproduction}} - t_{\text{reception of first data packet}}\right)[\text{s}]$$

7.3.5.11.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{reception of first data packet}}$	Start: "Buffering" message.	Start: Reception of the first data packet.
$t_{\text{start of stream reproduction}}$	Stop: Stream reproduction.	Stop: Streaming players signal when the reproduction of the stream starts.

7.3.5.12 Streaming teardown failure ratio [%]

7.3.5.12.1 Abstract definition

The parameter streaming teardown failure ratio is the probability that the "Teardown" RTSP message is sent from the UE client to the server and no "200 OK" RTSP response is received from the server.

7.3.5.12.2 Abstract equation

$$\text{Teardown failure ratio}\,[\%] = \frac{\text{cases without teardown server response}}{\text{all teardown attempts by UE client}} \times 100$$

7.3.5.12.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Teardown attempt	Start: User presses the "Stop" button.	Start: RTSP: TEARDOWN.
Successful Teardown	Stop: Stream is torn down.	Stop: RTSP: 200 OK.
Unsuccessful Teardown	Stop trigger point not reached.	

Some players do not send this TEARDOWN command at the end of the stream but a PAUSE command or in some cases, nothing at all. On the server side, a logic can then identify the status of the streams/clients.

Participating players should send the RTSP: TEARDOWN command in order to give a stable trigger point for measurements.

7.3.5.13 streaming teardown time [s]

7.3.5.13.1 Abstract definition

The parameter streaming teardown time is the duration between the UE client sending the "Teardown" RTSP message and the "200 OK" RTSP response from the server.

7.3.5.13.2 Abstract equation

$$\text{Streaming teardown time}\,[\text{s}] = \left(t_{\text{server response to teardown message}} - t_{\text{UE client sending teardown message}}\right)[\text{s}]$$

7.3.5.13.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{UE client sending teardown message}}$	Start: User presses the "Stop" button.	Start: RTSP: TEARDOWN.
$t_{\text{server response to teardown message}}$	Stop: Stream is torn down.	Stop: RTSP: 200 OK.

Some players do not send this TEARDOWN command at the end of the stream but a PAUSE command or in some cases, nothing at all. On the server side, a logic can then identify the status of the streams/clients.

Participating players should send the RTSP: TEARDOWN command in order to give a stable trigger point for measurements.

7.3.5.14 Streaming rebuffering failure ratio [%]

7.3.5.14.1 Abstract definition

The parameter streaming rebuffering failure ratio is the probability that a stream goes into rebuffering mode and does not restart the stream reproduction, afterwards.

7.3.5.14.2 Abstract equation

$$\text{Streaming rebuffering failure ratio}\,[\%] = \frac{\text{unsuccessful rebuffering attempts}}{\text{all rebuffering attempts}} \times 100$$

7.3.5.14.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Rebuffering attempt	Start: "Buffering" message appears.	Start: Streaming player signals the start of the stream buffering.
Successful continuation of reproduction	Stop: Stream reproduction continues.	Stop: Streaming player signals the continuation of the stream reproduction.
Unsuccessful continuation of reproduction	Stop trigger point not reached.	

7.3.5.15 Streaming rebuffering time [s]

7.3.5.15.1 Abstract definition

The parameter streaming rebuffering time is the duration between a stream going into rebuffering mode and continuation of the stream, afterwards.

7.3.5.15.2 Abstract equation

$$\text{Streaming rebuffering time}\,[s] = \left(t_{\text{continuation of stream}} - t_{\text{rebuffering message appears}}\right)[s]$$

7.3.5.15.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{rebuffering message appears}}$	Start: "Buffering" message appears.	Start: Streaming player signals the start of the stream buffering.
$t_{\text{continuation of stream}}$	Stop: Stream reproduction continues.	Stop: Streaming player signals the continuation of the stream reproduction.

7.3.6 Telephony

7.3.6.1 Telephony service non-accessibility [%]

7.3.6.1.1 Abstract definition

The telephony service non-accessibility is the probability that the end-user cannot access the mobile telephony service when requested if it is offered by display of the network indicator on the UE.

NOTE – Due to network problems and despite B-party being not busy (see preconditions for measurement), it may even be possible for the A-party to receive a busy or not reachable signal. In this case, since no ALERTING message will be sent, the test sample will be treated as a failure.

Figures 7-23 and 7-24 respectively show signalling flow charts for the third generation (3G) telephony mobile originated call establishment and mobile initiated call disconnection procedures.

Figure 7-25 shows the individual call set-up using on/off hook signalling procedure while Figure 7-26 shows the individual call set-up using direct set-up signalling procedure.

7.3.6.1.2 Abstract equation

$$\text{Telephony service non-accessibility}\,[\%] = \frac{\text{unsuccessful call attempts}}{\text{all call attempts}} \times 100$$

7.3.6.1.3 Trigger points

GSM:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Call attempt	Start: Push the send button.	Start: Layer 3 (RR): The "CHANNEL REQUEST" message is sent over the RACH.
Successful call attempt	Stop: Alerting tone is heard by the A-party and B-party rings.	Stop: Layer 3 (CC): The "ALERTING" message is passed: 1. from the B-party to the MSC (uplink) and 2. from the MSC to the A-party (downlink) to indicate that the B-party rings.
Unsuccessful data call access	Stop trigger point not reached.	

Universal mobile telecommunications system (UMTS):

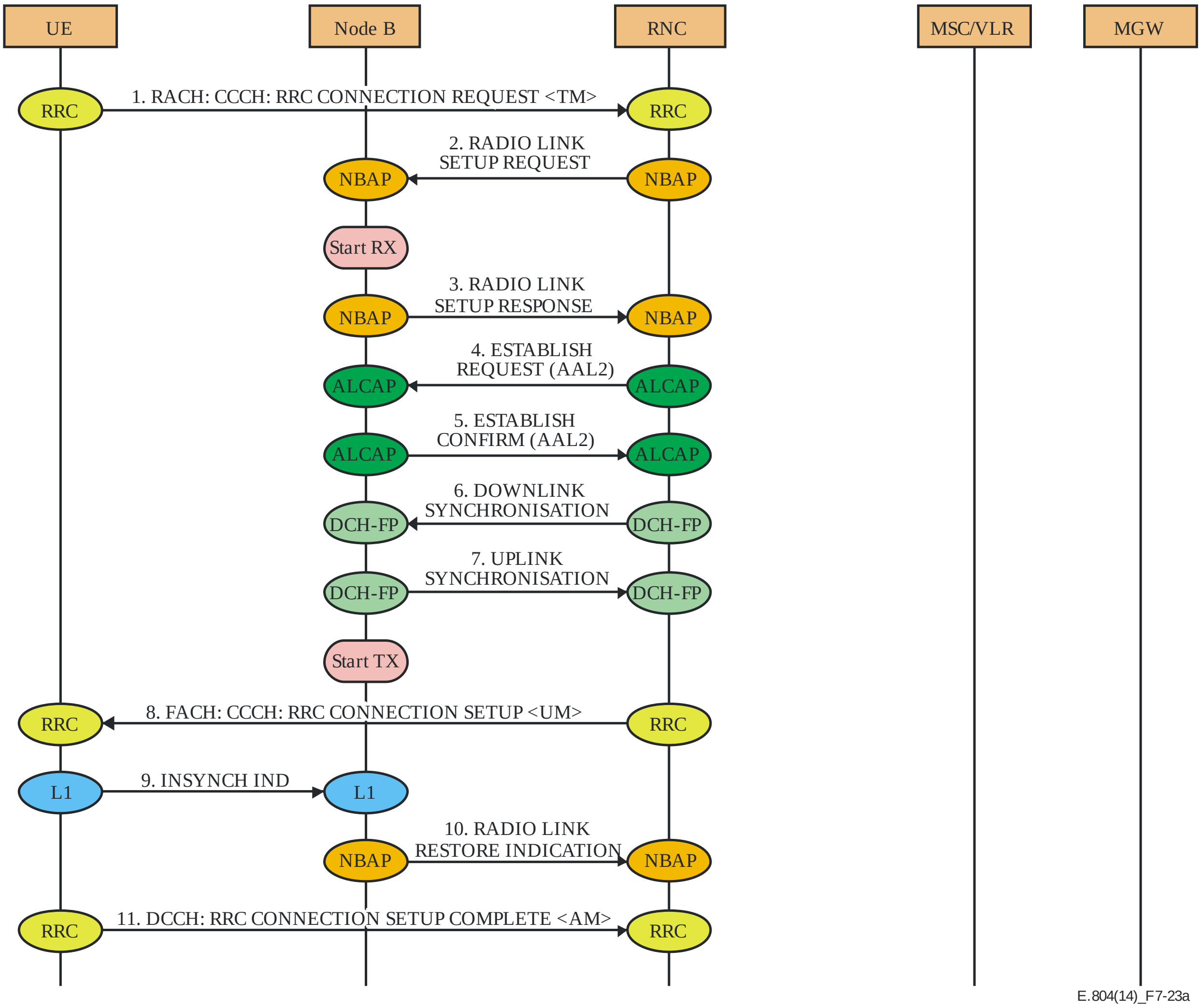
Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Call attempt	Start: Push the send button.	Start: Layer 3 (RRC): The first "RRC CONNECTION REQUEST" with establishment cause "Originating Conversational Call" message carried on the CCCH logical channel and mapped to the RACH transport channel is sent. (Figure 7-23; signalling point number 1). Comment: It is possible that the RRC connection is already established because of an, e.g., location update, then the start trigger is not reachable. In this case the current test sample should be deleted.
Successful call attempt	Stop: Alerting tone is heard by the A-party and the B-party rings.	Stop: Layer 3 (CC): The "ALERTING" message is passed: 1. from the B-party to the MSC (uplink) and 2. from the MSC to the A-party (downlink) to indicate that the B-party rings. (Figure 7-23; signalling point number 44).
Unsuccessful call attempt	Stop trigger point not reached.	
NOTE – With automatic tools, there is not a significant difference between considering the "ALERTING" or the "CONNECT" message, as the answering machine should always answer immediately.		

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Call attempt	Start: Push the send button.	Start: Layer 3 (CMCE): The "U-SETUP" message with appropriate signalling information is sent from the A-party. AT: The "ATD <dial string>" command is sent from the A-party, where <dial string> provides a unique identification of the desired B-party side. A preceding "AT+CTSDC" command is used to set the correct parameters for the dial command.
Successful call attempt	Stop: Alerting tone is heard by the A-party and the B-party rings.	Stop: Layer 3 (CMCE): 1. the "U-ALERT" message is passed from the B-party to the SwMI (uplink) and 2. the "D-ALERT" message is passed from the SwMI to the A-party (downlink) to indicate that the B-party rings. AT: The "ATA" command is sent by the B-party upon reception of the ring indication and the "AT+CTOCP: <CC instance>, <call status>, …" with <call status> = 2 (Called party paged) indication is received by the A-party to indicate that the B-party rings.
Unsuccessful call attempt	Stop trigger point not reached within desired time.	
NOTE – The described technical trigger points are valid for measurements with hook signalling enabled. In case direct signalling is used for the call establishment procedure, the relevant air interface protocol messages for the stop trigger are "U-CONNECT" and "D-CONNECT" (instead of "U-ALERT and "D-ALERT"), respectively. It shall be clearly stated which call establishment method is used for the telephony measurements.		

Preconditions for measurement:

Precondition	Covered by	Reference document
CS network available	Radio network unavailability	
CS attach successful		
B-party shall not be busy		



UE
Node B
RNC
MSC/VLR
MGW
RRC
1. RACH: CCCH: RRC CONNECTION REQUEST <TM>
RRC
2. RADIO LINK SETUP REQUEST
NBAP
NBAP
Start RX
3. RADIO LINK SETUP RESPONSE
NBAP
NBAP
4. ESTABLISH REQUEST (AAL2)
ALCAP
ALCAP
5. ESTABLISH CONFIRM (AAL2)
ALCAP
ALCAP
6. DOWNLINK SYNCHRONISATION
DCH-FP
DCH-FP
7. UPLINK SYNCHRONISATION
DCH-FP
DCH-FP
Start TX
RRC
8. FACH: CCCH: RRC CONNECTION SETUP <UM>
RRC
L1
9. INSYNCH IND
L1
10. RADIO LINK RESTORE INDICATION
NBAP
NBAP
RRC
11. DCCH: RRC CONNECTION SETUP COMPLETE <AM>
RRC
E.804(14)_F7-23a

Figure 7-23-a

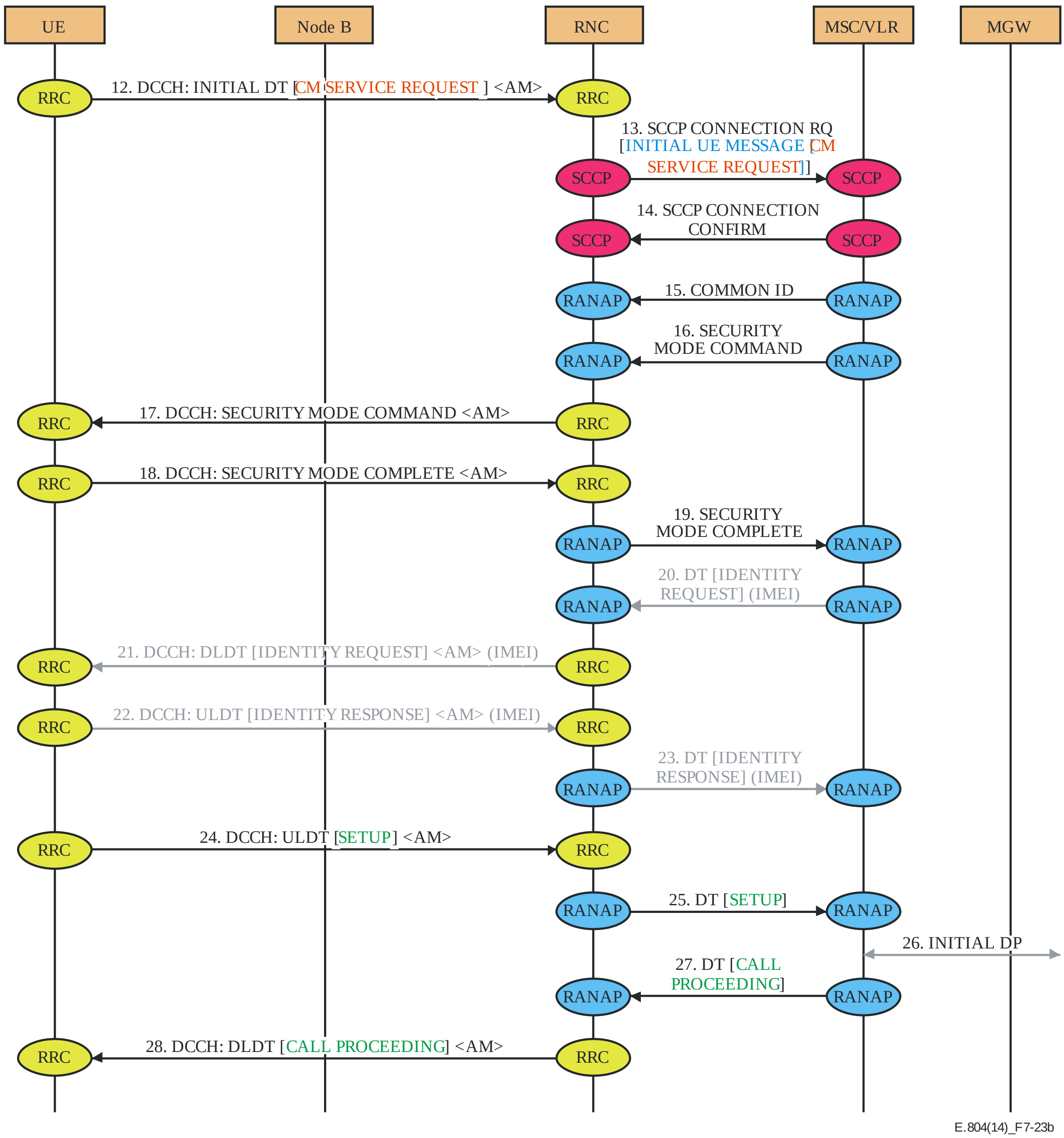

UE
Node B
RNC
MSC/VLR
MGW
RRC
12. DCCH: INITIAL DT [CM SERVICE REQUEST] <AM>
13. SCCP CONNECTION RQ [INITIAL UE MESSAGE [CM SERVICE REQUEST]]
SCCP
14. SCCP CONNECTION CONFIRM
15. COMMON ID
RANAP
16. SECURITY MODE COMMAND
17. DCCH: SECURITY MODE COMMAND <AM>
18. DCCH: SECURITY MODE COMPLETE <AM>
19. SECURITY MODE COMPLETE
20. DT [IDENTITY REQUEST] (IMEI)
21. DCCH: DLDT [IDENTITY REQUEST] <AM> (IMEI)
22. DCCH: ULDT [IDENTITY RESPONSE] <AM> (IMEI)
23. DT [IDENTITY RESPONSE] (IMEI)
24. DCCH: ULDT [SETUP] <AM>
25. DT [SETUP]
26. INITIAL DP
27. DT [CALL PROCEEDING]
28. DCCH: DLDT [CALL PROCEEDING] <AM>
E.804(14)_F7-23b

Figure 7-23-b

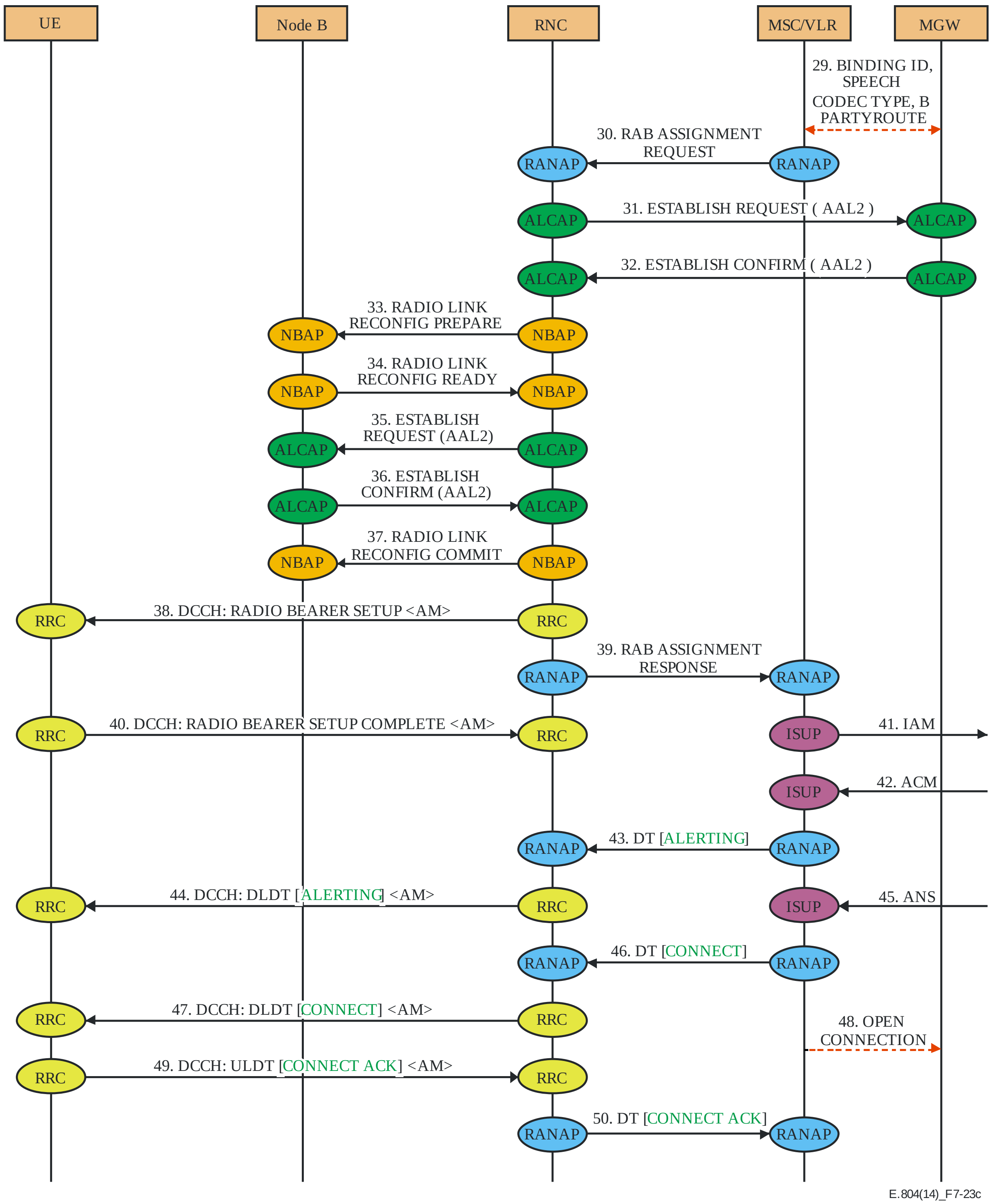


Figure 7-23-c

Figure 7-23 – Third-generation (3G) telephony signalling flow chart: mobile originated call establishment procedure

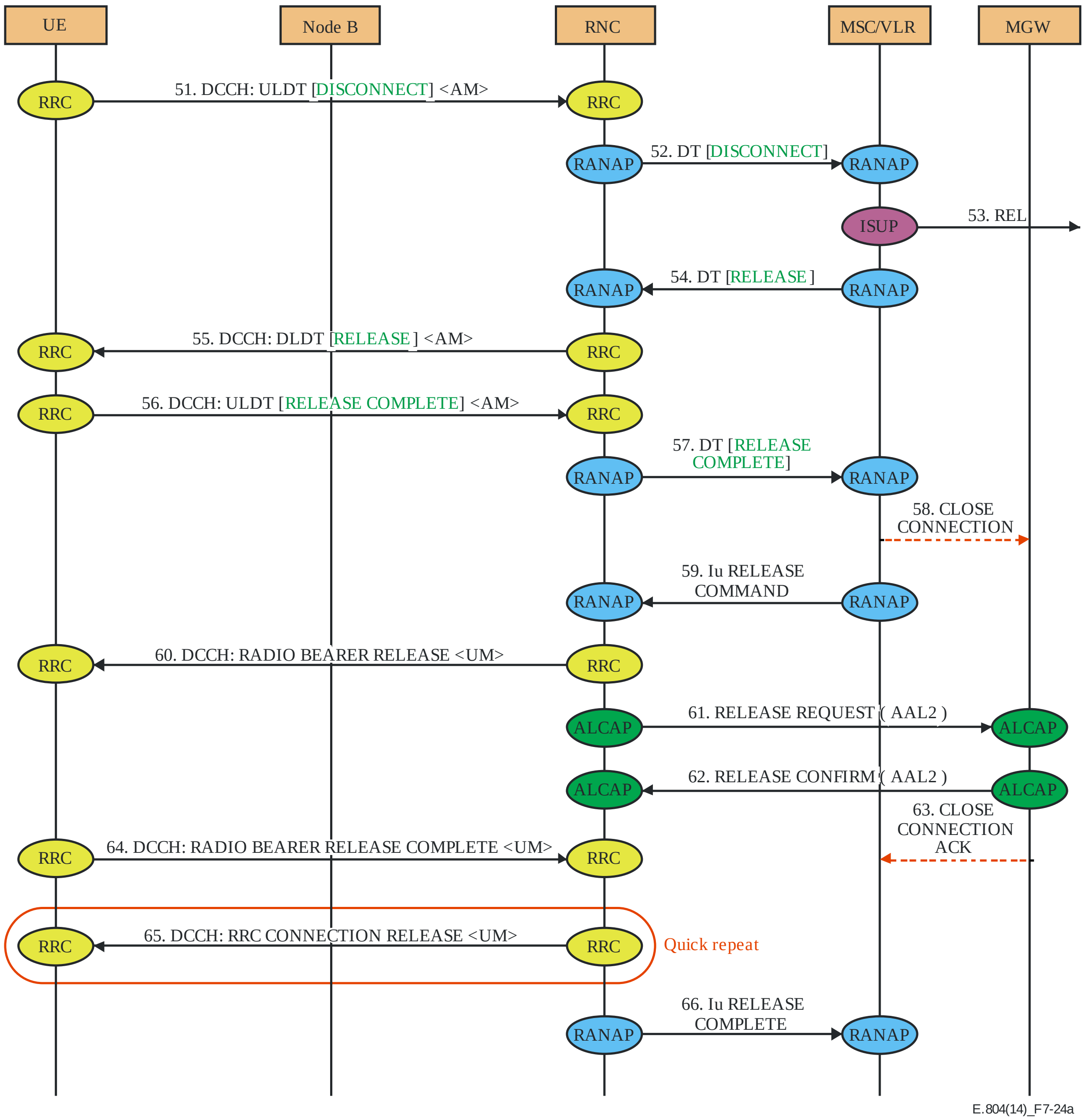

UE
Node B
RNC
MSC/VLR
MGW
RRC
51. DCCH: ULDT [DISCONNECT] <AM>
RANAP
52. DT [DISCONNECT]
ISUP
53. REL
54. DT [RELEASE]
55. DCCH: DLDT [RELEASE] <AM>
56. DCCH: ULDT [RELEASE COMPLETE] <AM>
57. DT [RELEASE COMPLETE]
58. CLOSE CONNECTION
59. Iu RELEASE COMMAND
60. DCCH: RADIO BEARER RELEASE <UM>
ALCAP
61. RELEASE REQUEST (AAL2)
62. RELEASE CONFIRM (AAL2)
63. CLOSE CONNECTION ACK
64. DCCH: RADIO BEARER RELEASE COMPLETE <UM>
65. DCCH: RRC CONNECTION RELEASE <UM>
Quick repeat
66. Iu RELEASE COMPLETE
E.804(14)_F7-24a

Figure 7-24-a

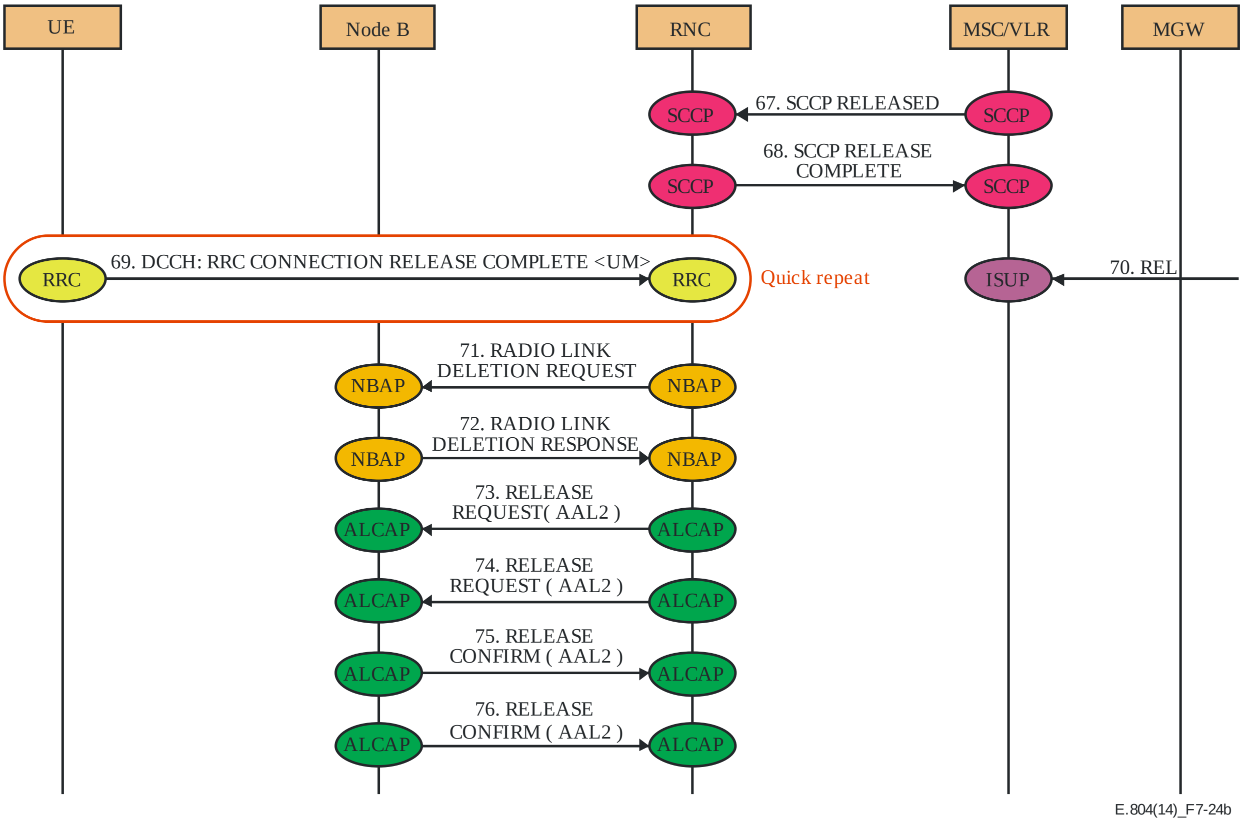


Figure 7-24-b

Figure 7-24 – Third-generation (3G) telephony signalling flow chart: mobile initiated call disconnection procedure

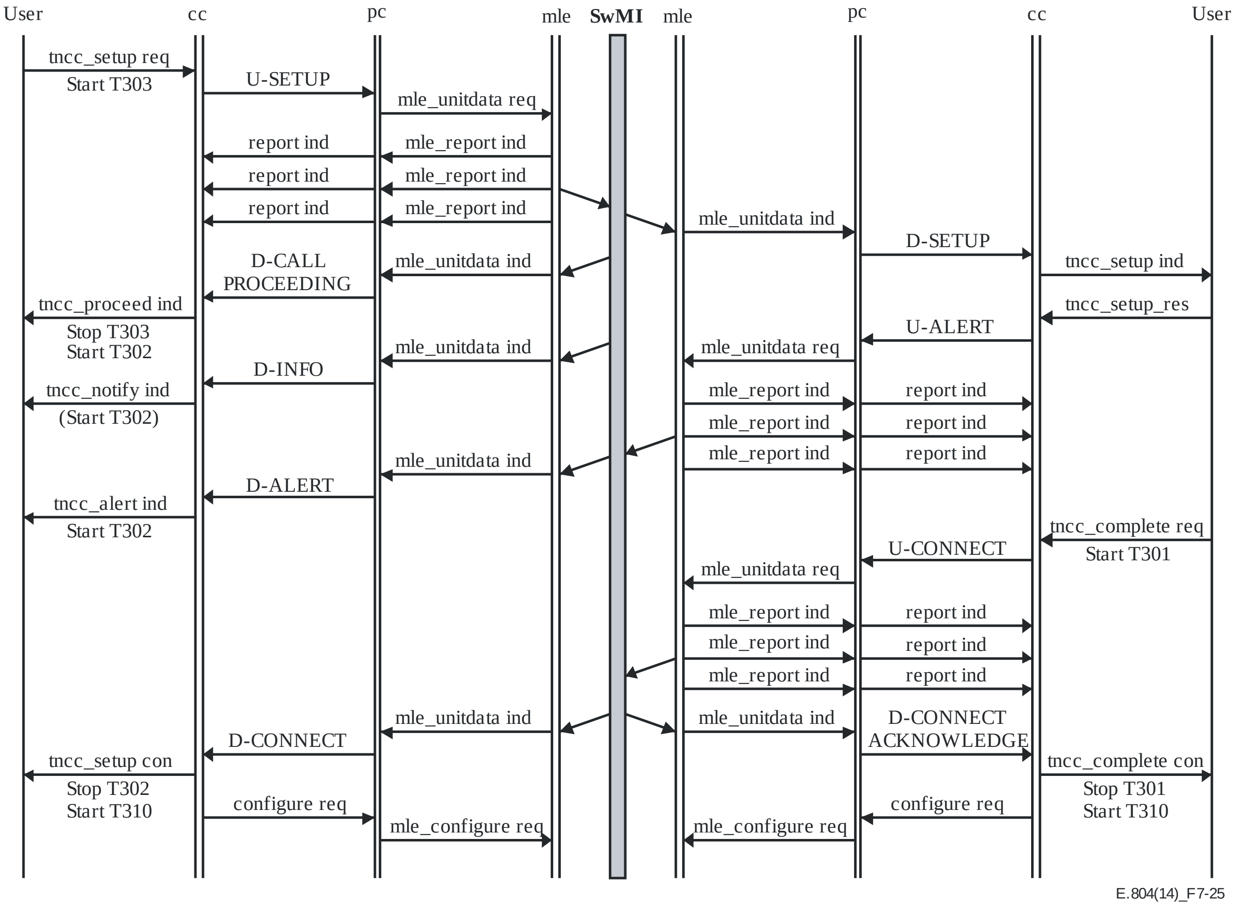


Figure 7-25 – Individual call set-up using on/off hook signalling ([ETSI EN 300 392-2], clause 14.5.1)

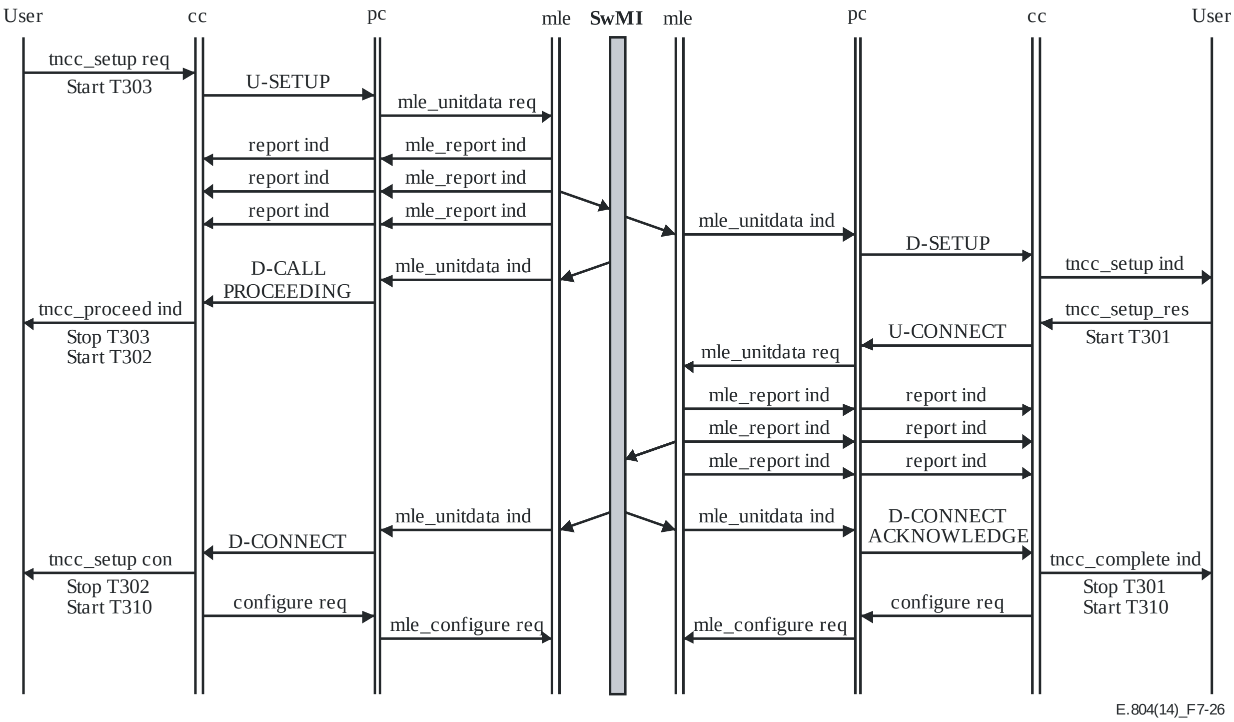


Figure 7-26 – Individual call set-up using direct set-up signalling ([ETSI EN 300 392-2], clause 14.5.1)

7.3.6.2 Telephony set-up time [s]

7.3.6.2.1 Abstract definition

The telephony set-up time is the time period between sending of complete address information and receipt of the call set-up notification.

7.3.6.2.2 Abstract equation

$$\text{Telephony set-up time}\,[s] = \left(t_{\text{connect established}} - t_{\text{user presses send button on UE}}\right)[s]$$

NOTE – This parameter is not calculated unless the telephony call set-up attempt is successful. It is assumed that early traffic channel assignment is used.

7.3.6.2.3 Trigger points

GSM:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{user presses send button on UE}}$: Time of call attempt	Start: Push the send button.	Start: Layer 3 (RR): The "CHANNEL REQUEST" message is sent over the RACH.
$t_{\text{connection established}}$: Time when connection is established (successful call attempt)	Stop: Alerting tone is heard by the A-party and the B-party rings.	Stop: Layer 3 (CC): The "ALERTING" message is passed: 1. from the B-party to the MSC (uplink) and 2. from the MSC to the A-party (downlink) to indicate that the B-party rings.

UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{user presses send button on UE}}$: Time of call attempt	Start: Push the send button.	Start: Layer 3 (RRC): The first "RRC CONNECTION REQUEST" with establishment cause "Originating Conversational Call" message carried on the CCCH logical channel and mapped to the RACH transport channel is sent. (Figure 7-23; signalling point number 1). Comment: It is possible that the RRC connection is already established because of an, e.g., location update, then the start trigger is not reachable. In this case the current test sample should be deleted.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{connection\ established}$: Time when connection is established (successful call attempt)	Stop: Alerting tone is heard by the A-party and the B-party rings.	Stop: Layer 3 (CC): The "ALERTING" message is passed: 1. from the B-party to the MSC (uplink) and 2. from the MSC to the A-party (downlink) to indicate that the B-party rings. (Figure 7-23; signalling point number 44).
NOTE – With automatic tools there is not a significant difference between considering the "ALERTING" or the "CONNECT" message, as the answering machine should always answer immediately.		

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{user\ presses\ send\ button\ on\ UE}$: Time of call attempt	Start: Push the send button.	Start: Layer 3 (CMCE): The "U-SETUP" message with appropriate signalling information is sent from the A-party. AT: The "ATD <dial string>" command is sent from the A-party, where <dial string> provides a unique identification of the desired B-party side. A preceding "AT+CTSDC" command is used to set the correct parameters for the dial command.
$t_{connection\ established}$: Time when connection is established (successful call attempt)	Stop: Alerting tone is heard by the A-party and the B-party rings.	Stop: Layer 3 (CMCE): 1. the "U-ALERT" message is passed from the B-party to the SwMI (uplink) and 2. the "D-ALERT" message is passed from the SwMI to the A-party (downlink) to indicate that the B-party rings. AT: The "ATA" command is sent by the B-party upon reception of the ring indication and the "AT+CTOCP: <CC instance>, <call status>, …" with <call status> = 2 (Called party paged) indication is received by the A-party to indicate that the B-party rings.
Unsuccessful call attempt	Stop trigger point not reached.	

Preconditions for measurement:

Precondition	Covered by	Reference document
CS network available	Radio network unavailability	
CS attach successful		
CS service access successful	Telephony service non-accessibility	

7.3.6.3 Telephony speech quality on call basis

7.3.6.3.1 Abstract definition

The telephony speech quality on call basis is an indicator representing the quantification of the end-to-end speech transmission quality of the mobile telephony service. This parameter computes the speech quality on the basis of completed calls.

NOTE – The acoustic behaviour of terminals is not part of this speech quality measurement.

7.3.6.3.2 Abstract equation

The validation of the end-to-end quality is made using MOS-listening speech quality objective (MOS-LQO) scales. These scales describe the opinion of users with speech transmission and its troubles (e.g., noise, robot voice, echo, dropouts, etc.) according to [ITU-T P.862] in conjunction with [ITU-T P.862.1], or according to [ITU-T P.863]. The algorithm used should be reported. The speech quality measurement is taken per call. An aggregation should be made on one value for speech quality per call.

$$\text{Telephony speech quality on call basis (received A-party)} = f(\text{MOS-LQO})$$

$$\text{Telephony speech quality on call basis (received B-party)} = f(\text{MOS-LQO})$$

Optionally, it might be useful to aggregate both speech quality (SpQ) values into one. In this case, the worst of both shall be used. This aggregated speech quality value shall be called SpQ (min).

7.3.6.3.3 Trigger points

GSM/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Not applicable.	Start: Interchange speech samples between the A-party and the B-party.	Start: Layer 3 (CC): The "CONNECT" message on the DCCH logical channel is passed from the MSC to the UE to indicate that the called user's end has been connected. (Figure 7-23; signalling point number 47).
Not applicable.	Stop: Release of connection.	Stop: Layer 3 (CC): The "DISCONNECT" message on the DCCH logical channel is intentionally sent from the UE (message sent when the user ends the call). (Figure 7-24; signalling point number 51).

TETRA:

The applicability of a suitable speech quality evaluation method for the narrow-band speech codec within TETRA networks is for further study.

7.3.6.4 Telephony speech quality on sample basis

7.3.6.4.1 Abstract definition

The telephony speech quality on sample basis is an indicator representing the quantification of the end-to-end speech transmission quality of the mobile telephony service. This parameter computes the speech quality on a sample basis.

NOTE – The acoustic behaviour of terminals is not part of this speech quality measurement.

7.3.6.4.2 Abstract equation

The validation of the end-to-end quality is made using MOS-LQO scales. These scales describe the opinion of users with speech transmission and its troubles (noise, robot voice, echo, dropouts, etc.) according to [ITU-T P.862] in conjunction with [ITU-T P.862.1], or according to [ITU-T P.863]. The algorithm used should be reported. The speech quality measurement is taken per sample. An aggregation of measurement activities or parts of it should be made on speech sample basis.

$$\text{Telephony speech quality on sample basis (received A-party)} = f(\text{MOS-LQO})$$

$$\text{Telephony speech quality on sample basis (received B-party)} = f(\text{MOS-LQO})$$

Optionally, it might be useful to aggregate both speech quality values into one. In this case the worst of both shall be used. This aggregated speech quality value shall be called SpQ (min).

7.3.6.4.3 Trigger points

The same as for speech quality on call basis (see clause 7.3.6.3.3).

7.3.6.5 Telephony cut-off call ratio [%]

7.3.6.5.1 Abstract definition

The telephony cut-off call ratio is the probability that a successful call attempt is ended by a cause other than the intentional termination by A- or B-party.

7.3.6.5.2 Abstract equation

$$\text{Telephony cut} - \text{off call ratio}[\%] = \frac{\text{unintentionally terminated telephony calls}}{\text{all successful telephony call attempts}} \times 100$$

7.3.6.5.3 Trigger points

GSM/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful telephony call attempt	Start: Alerting tone heard by the A-party coming from the B-party.	Start: Layer3 (CC): The "CONNECT" message on the DCCH logical channel is passed from the MSC to the UE to indicate that the connection has been established. (Figure 7-23; signalling point number 47). (See Note).
Intentionally terminated telephony call	Stop: Release of connection directly by the A- or the B-party.	Stop: Layer3 (CC): The "DISCONNECT" message on the DCCH logical channel is intentionally sent from the UE (message sent when the user ends the call). (Figure 7-24; signalling point number 51).
Unintentionally terminated telephony call	Stop trigger point not reached.	

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
NOTE – With automatic tools there is not a significant difference between considering the alerting or the connect message, as the answer machine should always answer immediately.		

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful telephony call attempt	Start: Connect indication received at the originating A-party side.	Start: Layer 3 (CMCE): The "D-CONNECT" message is received at the A-party to indicate that the called user's end has been connected. AT: The "AT+CTCC" indication is received by the A-party to indicate that the called user's end has been connected.
Intentionally terminated telephony call	Stop: Release of connection directly by the A- or the B-party.	Stop: Layer 3 (CMCE): The "U-DISCONNECT" message with disconnect cause "User requested disconnect" is sent from either the A-party or the B-party UE (message sent when the user ends the call). AT: The "ATH" command is sent by either the A-party or the B-party (message sent when the user ends the call).
Unintentionally terminated telephony call	Stop trigger point not reached.	

7.3.6.6 Telephony CLIP failure ratio [%]

7.3.6.6.1 Abstract definition

The telephony calling line identity presentation (CLIP) failure ratio denotes the percentage of call set-ups where a valid calling party number (CPN) parameter was sent but not received intact.

NOTE – To conform to legal request the calling line identity (CLI) may be suppressed in some (roaming) cases, taking into account that a roamed call may consist of two independent call legs.

7.3.6.6.2 Abstract equation

$$\text{Telephony CLIP failure ratio}[\%] = \frac{\text{number of calls received by B-party without intact CPN}}{\text{number of calls offered by A-party with valid CPN}} \times 100$$

7.3.6.6.3 Trigger points

GSM:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Calls offered by the A-party with valid CPN	Start: Push the send button at the A-party (calling party).	Start: Layer 3 (RR): The "CHANNEL REQUEST" message is sent by the UE over the RACH.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Calls received by the B-party without intact CPN	Stop: No presentation or presentation of invalid calling number on the display of the B-party mobile.	Stop: Layer 3 (CC): The "SETUP" message without valid calling party (A-party) number is received by the B-party.

UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Calls offered by the A-party with valid CPN	Start: Push the send button at the A-party (calling party).	Start: Layer 3 (RRC): The first "RRC CONNECTION REQUEST" message with establishment cause "Originating Conversational Call" message carried on the CCCH logical channel and mapped to the RACH transport channel is sent. (Figure 7-23: signalling point number 1) Comment: It is possible that the RRC connection is already established because of an, e.g., location update, then the start trigger is not reachable. In this case the current test sample should be deleted.
Calls received by the B-party without intact CPN	Stop: No presentation or presentation of invalid calling number on the display of the B-party mobile.	Stop: Layer 3 (CC): The "SETUP" message without valid calling party (A-party) number is received by the B-party.

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Calls offered by the A-party with valid CPN	Start: Push the send button at the A-party (calling party).	Start: Layer 3 (CMCE): The "U-SETUP" message with appropriate signalling information is sent from the A-party.
Calls received by the B-party without intact CPN	Stop: No presentation or presentation of invalid calling number on the display of the B-party mobile.	Stop: Layer 3 (CMCE): The "D-SETUP" message without valid calling party (A-party) number (calling party identifier) is received by the B-party.

7.3.7 Video telephony

7.3.7.1 Network accessibility/availability

Network availability and network accessibility are measured independently from the service, and will not be described further in this clause. Network availability and network accessibility are pre-conditions for the performance of the QoS measurement.

7.3.7.2 Parameter overview chart

To get a better overview of the following parameters, Figure 7-27 shows all steps of a video telephony (VT) call from origin to destination, and the related QoS parameters.

Preconditions for the measurements: It should be a bi-directional VT call. Both sides should allow the transmission of both audio and video.

Explanation of Figure 7-27: The upper half considers the trigger points and the parameters at the originated side and the lower half at the terminated side. The rectangles are connected to the trigger points that are relevant for analysis. For example: "t3, orig. side" (trigger point at originated side) and "t3, term. side" (trigger point at terminated side) are points of time that describe a similar event but it could be passed at slightly different times. The preconditions are specified in brackets beside the parameter name. The technical triggers are defined for positive successful cases, if the VT works fine. For failures the triggers are the opposite, this means the non-existence of the message indicates the failure. The bold lines behind the trigger points tx are the used one and the dashed one are unused.

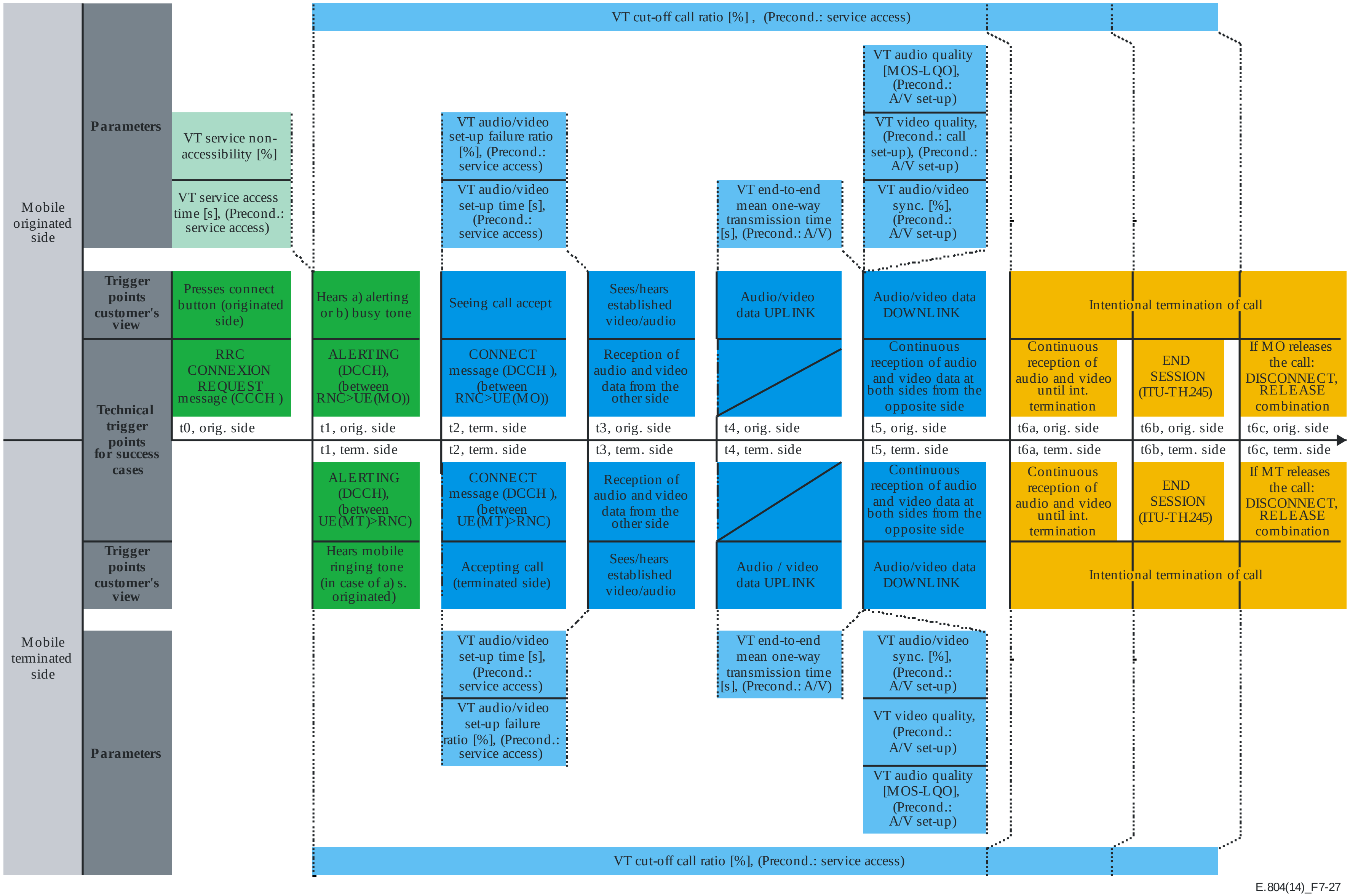


Figure 7-27 – Parameter overview with trigger points

7.3.7.3 VT service non-accessibility [%]

7.3.7.3.1 Abstract definition

Video telephony (VT) service non-accessibility is the probability that the end-user cannot access the service when requested while it is offered by network indication on the mobile equipment.

NOTE – Due to network problems and despite mobile originating (MO) side being not busy (see preconditions for measurement), it may even be possible for the MO side to receive a busy or not reachable signal. In this case, since no ALERTING message will be sent, the test sample will be treated as a failure.

7.3.7.3.2 Abstract equation

$$\text{VT service non-accessibility}\,[\%] = \frac{\text{unsuccessful video telephony call access attempts}}{\text{all video telephony call access attempts}} \times 100$$

7.3.7.3.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Video telephony call attempt	Start: Push the send button.	Start: The first RRC CONNECTION REQUEST with establishment of cause "Originating Conversational Call" message carried in the CCCH logical channel and mapped to the RACH transport channel is sent. (Figure 7-28, signalling point number 1). Comment: It is possible that more than one RRC CONNECTION REQUEST message per call attempt is sent. Only the first RRC CONNECTION REQUEST with establishment of cause "Originating Conversational Call" message should be taken into account for the calculation. It is possible that the RRC connection is already established because of an, e.g., location update, then the start trigger is not reachable. In this case the current test sample should be deleted.
Successful video telephony call attempt	Stop: Alerting tone is heard by the MO side coming from the MT side and the MT side rings.	Stop: The ALERTING message on the DCCH logical channel is passed: 1. from the UE at MT side to MSC (uplink) and 2. from the MSC to the UE at MO side (downlink) to indicate that the MT side rings. (Figure 7-28, signalling point number 44).
Unsuccessful video telephony call attempt	Stop: Trigger point not reached.	Stop: Trigger point not reached.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio network unavailability	
UMTS CS attach successful		
MT side shall not be busy		

7.3.7.4 VT service access time [s]

7.3.7.4.1 Abstract definition

The VT service access time is the time between pushing the send button after input of the mobile station ISDN number (MSISDN) and receipt of alerting message at the MO side.

Remark:

- This parameter is not calculated unless the video telephony call access attempt is successful. At the MT side, the mobile device shall ring.

7.3.7.4.2 Abstract equation

$$\text{VT service access time}\,[s] = \left(t_{\text{alerting tone}} - t_{\text{pushsendbutton}}\right)[s]$$

7.3.7.4.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{push send button}}$: Time of video telephony call attempt	Start: Push the send button.	Start: The first RRC CONNECTION REQUEST with establishment cause "Originating Conversational Call" message carried on the CCCH logical channel and mapped to the RACH transport channel is sent. (Figure 7-28, signalling point number 1). Comment: It is possible that more than one RRC CONNECTION REQUEST message per call attempt is sent. Only the first RRC CONNECTION REQUEST with establishment cause "Originating Conversational Call" message should be taken into account for the calculation. It is possible that the RRC connection is already established because of an, e.g., location update, then the start trigger is not reachable. In this case the current test sample should be deleted.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{alerting\ tone}$: Time of successful video telephony call attempt	Stop: Alerting tone is heard by the MO side coming from the MT side and the MT side rings.	Stop: The ALERTING message on the DCCH logical channel is passed: 1. from the UE at MT side to MSC (uplink) and 2. from the MSC to the UE at MO side (downlink) to indicate that the MT side rings. (Figure 7-28, signalling point number 44).

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio network unavailability	
UMTS CS attach successful		
UMTS CS service access	VT service access failure ratio	

7.3.7.5 VT audio/video set-up failure ratio [%]

7.3.7.5.1 Abstract definition

The VT audio/video set-up failure ratio is the probability of audio/video set-up failure after service access. The audio/video set-up is successful if audio and video output is performed at both sides.

Remarks:

- This parameter reports a failure if the end-trigger is not reached at both sides.
- This parameter is not calculated unless the VT service access attempt is successful.
- This parameter depends on the mobile device used and on the multimedia protocol stack implemented (e.g., answer fast feature).

7.3.7.5.2 Abstract equation

$$\text{VT audio/video set-up failure ratio}\,[\%] = \frac{\text{audio/video set-up failures}}{\text{all accepted calls at MT side}} \times 100$$

7.3.7.5.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Audio/video set-up attempt	Start: MO sees the call acceptance from the MT side.	Start: The CONNECT message on the DCCH logical channel is passed from the MSC to the UE at MO side to indicate that the connection has been established. (Figure 7-28, signalling point number 47)
Audio/video set-up success	Stop: Start of the audio and video output at both sides.	Stop: Start of reception of the audio and video data at both sides from the opposite side. Comment: All four data streams shall be received for a success.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Audio/video set-up failure	Stop: Trigger point not reached.	Stop: Trigger point not reached.

Preconditions of measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio network unavailability	
UMTS CS attach successful		
UMTS CS service access successful	VT service non-accessibility	

7.3.7.6 VT audio/video set-up time [s]

7.3.7.6.1 Abstract definition

The VT audio/video set-up time is the elapsed time from the MT call acceptance indicated at MO side until audio and video output starts at both sides.

Remarks:

- This parameter should report the worse time of both sides.
- This parameter is not calculated unless the VT audio/video set-up attempt is successful.
- This parameter depends on the mobile used and on the multimedia protocol stack implemented (e.g., answer fast feature).

7.3.7.6.2 Abstract equation

$$\text{VT audio/video set-up time}\,[s] = \left(t_{\text{audio/video start}} - t_{\text{MT accepts call}}\right)[s]$$

7.3.7.6.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{MT accepts call}}$: Time of the beginning of the audio/video set-up	Start: MO sees the call acceptance from the MT side.	Start: The CONNECT message on the DCCH logical channel is passed from the MSC to the UE at MO side to indicate that the connection has been established. (Figure 7-28, signalling point number 47).
$t_{\text{audio/video start}}$: Time of successful audio/video set-up	Stop: Start of the audio and video output at both sides.	Stop: Start of reception of audio and video data at both sides from the opposite side + constant time value for decoding. Comment: All four data streams shall be received for a success.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio network unavailability	
UMTS CS attach successful		
UMTS CS audio/video set-up successful	VT audio/video set-up failure ratio	

7.3.7.7 VT cut-off call ratio [%]

7.3.7.7.1 Abstract definition

The VT cut-off call ratio is the probability that a successful service access is ended by a cause other than the intentional termination of the user (calling or called party).

Remark:

- This parameter is not calculated unless the VT service access attempt is successful. A VT call is considered dropped:
  - if the call acceptance fails after alerting;
  - if audio/video set-up fails; or
  - if either the audio, the video or both are lost at one or both sides for an interruption timeout and before the end of "predefined call duration".

The "predefined call duration" is the difference between the indication of the call acceptance at the MO side and the intentional release of the call.

7.3.7.7.2 Abstract equation

$$\text{VT cut-off call ratio}\,[\%] = \frac{\text{video telephony dropped calls}}{\text{all successful video telephony service access attempts}} \times 100$$

7.3.7.7.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful video telephony service access attempt	Start: Alerting tone is heard by the MO side coming from MT side and the MT side rings.	Start: The ALERTING message on the DCCH logical channel is passed: 1. from the UE at the MT side to MSC (uplink) and 2. from the MSC to the UE at the MO side (downlink) to indicate that the MT side rings. (Figure 7-28, signalling point number 44).

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Video telephony successful call	Stop: No loss of video and/or audio without any intention by the MO or the MT side longer than the interruption timeout within the predefined call duration.	Stop: 1. If the test system can capture audio/video information: Continuous reception of audio and video data at both sides from the opposite side without an interruption longer than the interruption timeout until intentional call release. 2. If the test system cannot capture audio/video information: The following information shall not be seen in the signalling before intentional call release but they shall be seen after the intentional call release: • H.245 EndSession command (endSessionCommand disconnect) or • the following trigger combination (all triggers on the DCCH logical channel): [M1: DISCONNECT (uplink).] and [M2: DISCONNECT (downlink) or RELEASE (downlink)] (Figure 7-28, signalling point number 51). Comment: In some cases the mobile devices do not use the EndSession command but only the DISCONNECT or RELEASE command.
Video telephony dropped calls	Stop trigger point not reached.	Stop: Trigger point not reached.

If the reception of audio and/or video is interrupted shortly before the predefined call duration, then the call duration shall be extended to check if the interruption persists for the interruption timeout or not. If the interruption is shorter than the interruption timeout, the call shall be released immediately and rated as success. Otherwise, the sample shall be rated as failure and the call will be released.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio network unavailability	
UMTS CS attach successful		
UMTS CS service access successful	VT service non-accessibility	

7.3.7.8 VT speech quality on call basis

7.3.7.8.1 Abstract definition

The VT speech quality on call basis is an indicator representing the quantification of the end-to-end speech transmission quality of the video telephony service. This parameter computes the speech quality on the basis of completed calls.

Remarks:

- This parameter is not calculated unless the VT audio/video set-up attempt is successful.
- The speech quality measurement is taken per call. An aggregation of measurement activities or parts of it should be made on speech sample basis.
- The acoustic behaviour of terminals is not part of this audio quality measurement. The modelling of the acoustic part of the handset-terminals (e.g., frequency shaping) is incorporated in the speech quality assessment algorithm. Therefore, the test mobile devices used have to be connected at their electrical interfaces and not coupled acoustically. It has to be taken into account that a detailed way for insertion and capturing of audio signals is described in [ITU-T P.862.3].
- For wideband (7 kHz) applications, a standardized algorithm is available in [ITU-T P.862.2].
- Evaluation of a MO down link (DL) or MT DL and also for these both directions (sum) is possible by calculating the mean value of the results from all samples.
- Experience has shown a high variable delay in video calls.
- Recommendation [ITU-T P.862] is not suitable for testing such video call applications. It has to be taken into account that further studies, including auditory tests of video calls, have to be conducted.

7.3.7.8.2 Abstract equation

Recommendation [ITU-T P.862] together with the related mapping given in [ITU-T P.862.1] is recommended. This algorithm describes the opinion of users related to speech transmission quality (300 Hz through 3400 Hz) and its related impairments (e.g., background noise, unnatural voice, temporal clipping and interruptions, etc.).

The speech quality measurement is taken per call and per direction (DL at MO, DL at MT).

After mapping the raw [ITU-T P.862] results according to [ITU-T P.862.1], the speech quality assessment is presented in a MOS-like scale between one and five called MOS listening quality objective (MOS-LQO), as defined in [ITU-T P.800.1].

7.3.7.8.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful audio/video set-up attempt	Start: Start of the audio and video output at both sides.	Start: Start of reception of audio and video data at both sides from the opposite side. Comment: All four data streams shall be received for a success.
End of call (only intentional)	Stop: End of call.	Stop: End of continuous reception of audio and video data at both sides from the opposite side because of: • intentional call release.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio network unavailability	
UMTS CS attach successful		
UMTS CS service access successful	VT service non-accessibility	
UMTS CS audio/video set-up successful	VT audio/video set-up failure ratio	

7.3.7.9 VT speech quality on sample basis

7.3.7.9.1 Abstract definition

The VT speech quality on sample basis is an indicator representing the quantification of the end-to-end speech transmission quality as perceived by the user. This parameter computes the speech quality on a sample basis.

Remarks:

- This parameter is not calculated unless the VT audio/video set-up attempt is successful.
- Speech quality values from all video telephony calls should be taken into consideration for statistical quality analysis.
- The speech quality measurement is taken per sample. An aggregation of measurement activities or parts of it should be made on speech sample basis. Only complete received samples of a dropped call are evaluable.
- The acoustic behaviour of terminals is not part of this audio quality measurement. The modelling of the acoustic part of the handset-terminals (e.g., frequency shaping) is incorporated in the speech quality assessment algorithm. Therefore, the test mobile devices used have to be connected at their electrical interfaces and not coupled acoustically. It has to be taken into account that a detailed way for insertion and capturing of audio signals is described in [ITU-T P.862.3].
- For wideband (7 kHz) applications a standardized algorithm is available in [ITU-T P.862.2].
- Evaluation of a MO DL or MT DL and also for these both directions (sum) is possible by calculating the mean value of the results from all samples.
- Experience has shown a high variable delay in video calls.
- Recommendation [ITU-T P.862] is not suitable for testing such video call applications. It has to be taken into account that further studies including auditory tests of video calls have to be conducted.

7.3.7.9.2 Abstract equation

$$\text{VT speech quality on sample basis (received A-party)} = f(\text{MOS-LQO})$$

$$\text{VT speech quality on sample basis (received B-party)} = f(\text{MOS-LQO})$$

Recommendation [ITU-T P.862] together with the related mapping given in [ITU-T P.862.1] is recommended. This algorithm describes the opinion of users related to speech transmission quality (300 Hz through 3400 Hz) and its related impairments (e.g., background noise, unnatural voice, temporal clipping and interruptions, etc.).

The speech quality measurement is taken per sample and per direction (DL at MO, DL at MT).

After mapping the raw [ITU-T P.862] results according to [ITU-T P.862.1], the speech quality assessment is presented in a MOS-like scale between one and five called MOS listening quality objective (MOS-LQO), as defined in [ITU-T P.800.1].

7.3.7.9.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful audio/video set-up attempt	Start: Start of the audio and video output at both sides.	Start: Start of reception of audio and video data at both sides from the opposite side. Comment: All four data streams shall be received for a success.
End of the call (intentional or dropped)	Stop: End of the call.	Stop: End of continuous reception of the audio and video data at both sides from the opposite side because of: • an interruption for a predefined duration or longer or • intentional call release.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio network unavailability	
UMTS CS attach successful	Attach failure ratio	
UMTS CS service access successful	VT service non-accessibility	
UMTS CS audio/video set-up successful	VT audio/video set-up failure ratio	

7.3.7.10 VT video quality

7.3.7.10.1 Abstract definition

The VT video quality is an end-to-end quality of the video signal as perceived by the end user during a VT call. This parameter computes the video quality on a sample basis.

Remarks:

- This parameter is not calculated unless the VT audio/video set-up attempt is successful.
- Video quality values from all video telephony calls should be taken into consideration for statistical quality analysis.
- The video quality measurement is taken per sample. An aggregation of measurement activities or parts of it should be made on video sample basis. Only complete received samples of a dropped call are evaluable.
- Evaluation of a MO DL or MT DL and also for these both directions (sum) is possible by calculating the mean value of the results from all samples.

7.3.7.10.2 Abstract equation

To be specified.

7.3.7.10.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful audio/video set-up attempt	Start: Start of the audio and video output at both sides.	Start: Start of reception of the audio and video data at both sides from the opposite side. Comment: All four data streams shall be received for a success.
End of the call (intentional or dropped)	Stop: End of the call.	Stop: End of continuous reception of the audio and video data at both sides from the opposite side because of: • an interruption for a predefined duration or longer; or • intentional call release.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio network unavailability	
UMTS CS attach successful	Attach failure ratio	
UMTS CS service access successful	VT service non-accessibility	
UMTS CS audio/video set-up successful	VT audio/video set-up failure ratio	

7.3.7.11 VT end-to-end mean one-way transmission time [s]

7.3.7.11.1 Abstract definition

The VT end-to-end mean one-way transmission time is the delay time from input of the signal at MS (MO/MT) (mic/cam) to output of the signal at MS (MT/MO) (loudspeaker/display).

Remark:

- This parameter is not calculated unless the VT audio/video set-up attempt is successful.

7.3.7.11.2 Abstract equation

Time from input of the signal at MS (MO/MT) to output at MS (MT/MO).

Aggregation algorithm: ((Transmission time MO ->MT) + (Transmission time MT->MO))/2.

In case of a symmetrical channel, one party could be configured as loopback device. The other one can determine the double delay by correlating transmit and receive signal. The delay should be measured after the loopback at the top of the radio bearer.

As the delay of the codec is almost constant for a specific mobile implementation, the codec delay could be considered by a mobile depending offset. In each direction, one shall add the encoder and the decoder times. For the whole loopback, one shall calculate the following times:

MO → MT	Encoding of audio/video (slowest is used)	a
	Transmission of audio/video (slowest is used)	b
	Decoding of audio/video (slowest is used)	c
MT → MO	Encoding of audio/video (slowest is used)	d
	Transmission of audio/video (slowest is used)	e
	Decoding of audio/video (slowest is used)	f

$$\text{VT end-to-end mean one-way transmission time}\,[s] = \frac{a+b+c+d+e+f}{2}\,[s]$$

7.3.7.11.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful audio/video set-up attempt	Start: Start of the audio and video output at both sides.	Start: Start of reception of the audio and video data at both sides from the opposite side. Comment: All four data streams shall be received for a success.
End of call (intentional or dropped)	Stop: End of the call.	Stop: End of continuous reception of the audio and video data at both sides from the opposite side because of: • an interruption for a predefined duration or longer; or • intentional call release.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio network unavailability	
UMTS CS attach successful	Attach failure ratio	
UMTS CS service access successful	VT service non-accessibility	
UMTS CS audio/video set-up successful	VT audio/video set-up failure ratio	

7.3.7.12 VT audio/video synchronization [%]

7.3.7.12.1 Abstract definition

Percentage of times that the time differences of the audio and video signal at the user side exceeds a predefined threshold.

Remarks:

- This parameter is not calculated unless the VT audio/video set-up attempt is successful.
- Only if audio and video use different bearers this indicator would reflect the behaviour of the network and the mobiles.

7.3.7.12.2 Abstract equation

To be specified.

7.3.7.12.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful audio/video set-up attempt	Start: Start of the audio and video output at both sides.	Start: Start of reception of audio and video data at both sides from the opposite side. Comment: All four data streams shall be received for a success.
End of call (intentional or dropped)	Stop: End of the call.	Stop: End of continuous reception of the audio and video data at both sides from the opposite side because of: • an interruption for a predefined duration or longer; or• intentional call release.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio network unavailability	
UMTS CS attach successful	Attach failure ratio	
UMTS CS service access successful	VT service non-accessibility	
UMTS CS audio/video set-up successful	VT audio/video set-up failure ratio	

7.3.7.13 Signalling diagrams

Figure 7-28 shows the video telephony signalling flow chart of a mobile originated call until the call release. The point of view is the MO side.

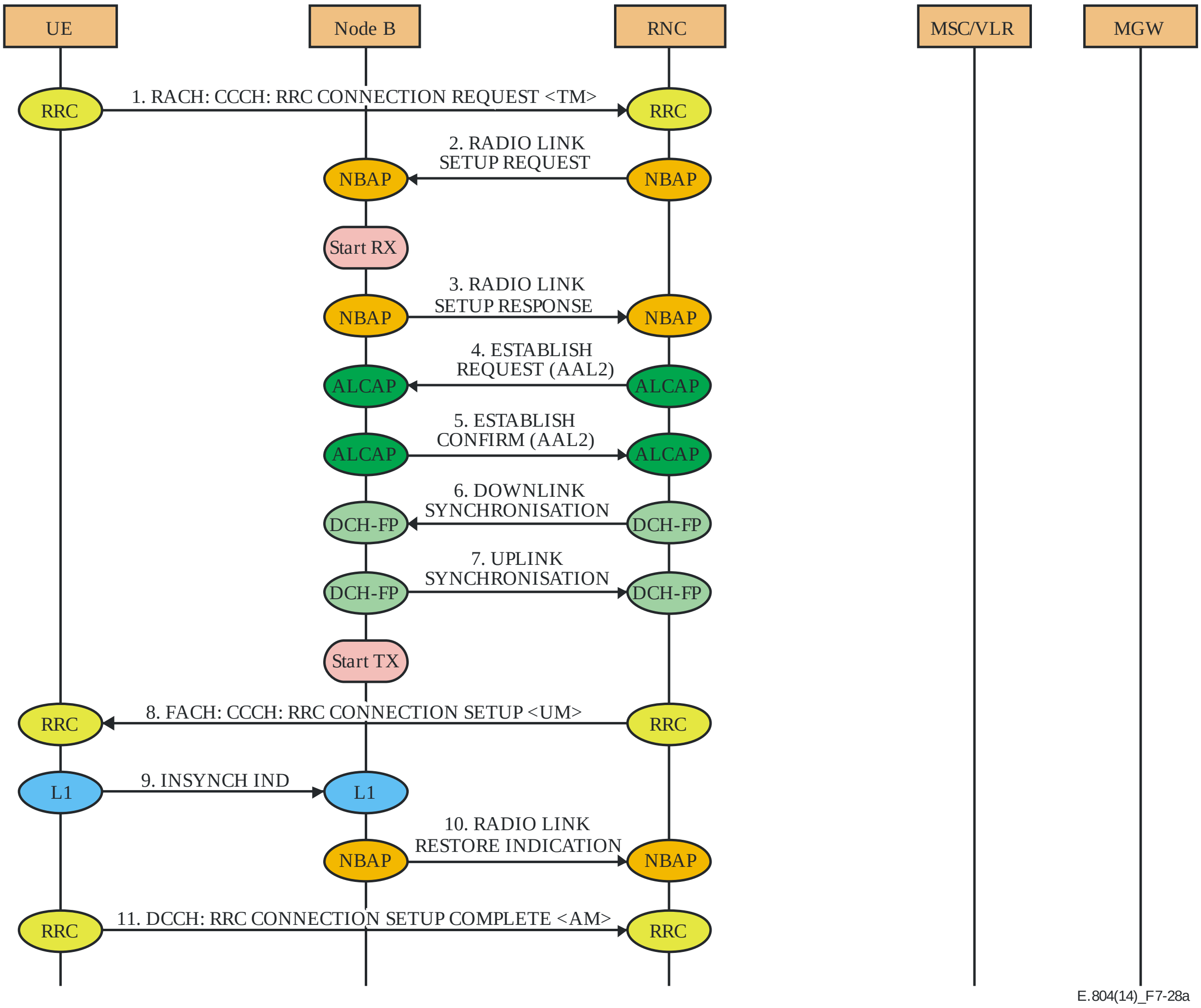

UE
Node B
RNC
MSC/VLR
MGW
RRC
1. RACH: CCCH: RRC CONNECTION REQUEST <TM>
RRC
2. RADIO LINK
SETUP REQUEST
NBAP
NBAP
Start RX
3. RADIO LINK
SETUP RESPONSE
NBAP
NBAP
4. ESTABLISH
REQUEST (AAL2)
ALCAP
ALCAP
5. ESTABLISH
CONFIRM (AAL2)
ALCAP
ALCAP
6. DOWNLINK
SYNCHRONISATION
DCH-FP
DCH-FP
7. UPLINK
SYNCHRONISATION
DCH-FP
DCH-FP
Start TX
RRC
8. FACH: CCCH: RRC CONNECTION SETUP <UM>
RRC
L1
9. INSYNCH IND
L1
10. RADIO LINK
RESTORE INDICATION
NBAP
NBAP
RRC
11. DCCH: RRC CONNECTION SETUP COMPLETE <AM>
RRC
E.804(14)_F7-28a

Figure 7-28a

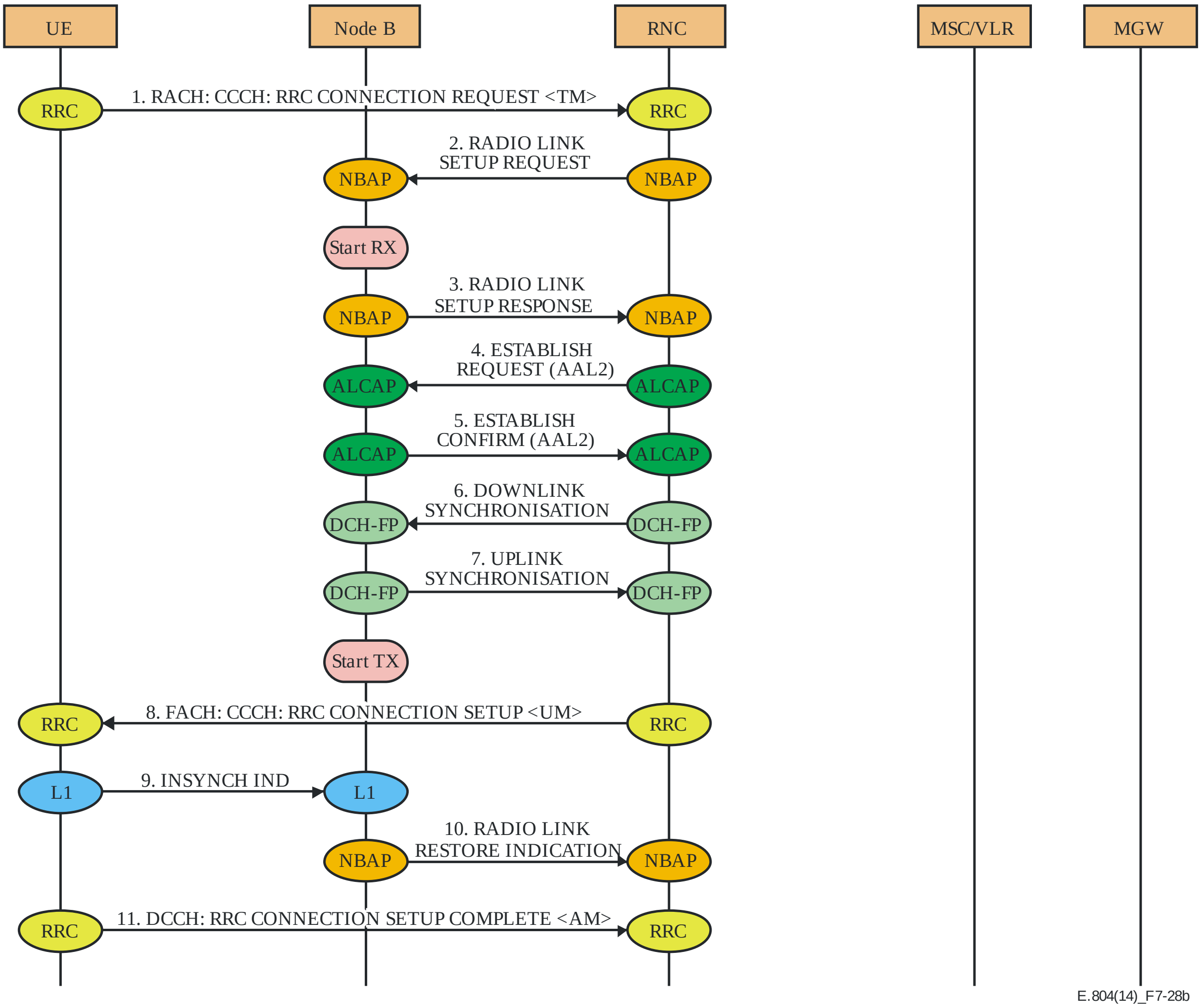

UE
Node B
RNC
MSC/VLR
MGW
RRC
1. RACH: CCCH: RRC CONNECTION REQUEST <TM>
2. RADIO LINK SETUP REQUEST
NBAP
Start RX
3. RADIO LINK SETUP RESPONSE
4. ESTABLISH REQUEST (AAL2)
ALCAP
5. ESTABLISH CONFIRM (AAL2)
6. DOWNLINK SYNCHRONISATION
DCH-FP
7. UPLINK SYNCHRONISATION
Start TX
8. FACH: CCCH: RRC CONNECTION SETUP <UM>
9. INSYNCH IND
L1
10. RADIO LINK RESTORE INDICATION
11. DCCH: RRC CONNECTION SETUP COMPLETE <AM>
E.804(14)_F7-28b

Figure 7-28b

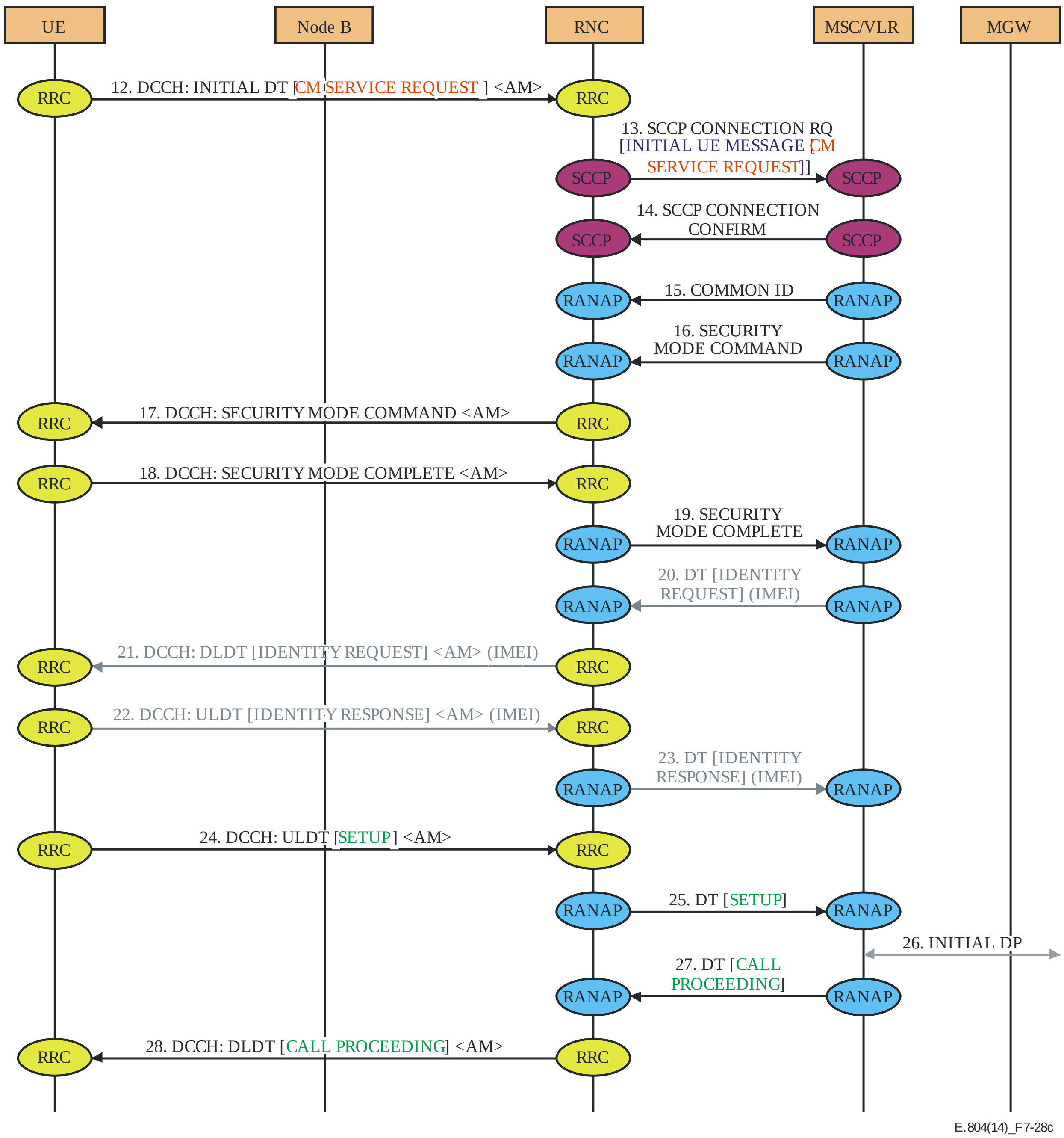
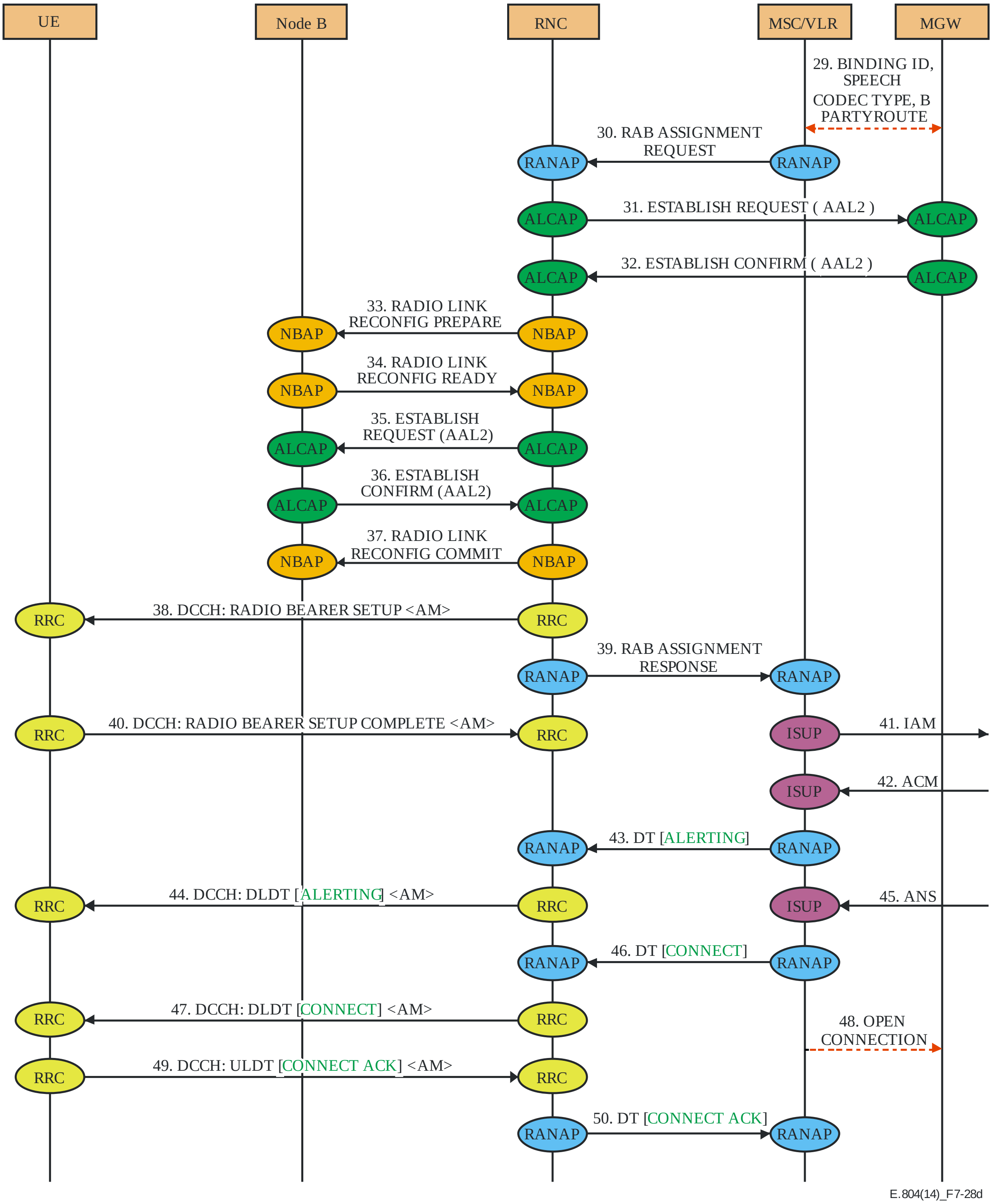

UE
Node B
RNC
MSC/VLR
MGW
RRC
12. DCCH: INITIAL DT [CM SERVICE REQUEST] <AM>
13. SCCP CONNECTION RQ [INITIAL UE MESSAGE [CM SERVICE REQUEST]]
SCCP
14. SCCP CONNECTION CONFIRM
15. COMMON ID
RANAP
16. SECURITY MODE COMMAND
17. DCCH: SECURITY MODE COMMAND <AM>
18. DCCH: SECURITY MODE COMPLETE <AM>
19. SECURITY MODE COMPLETE
20. DT [IDENTITY REQUEST] (IMEI)
21. DCCH: DLDT [IDENTITY REQUEST] <AM> (IMEI)
22. DCCH: ULDT [IDENTITY RESPONSE] <AM> (IMEI)
23. DT [IDENTITY RESPONSE] (IMEI)
24. DCCH: ULDT [SETUP] <AM>
25. DT [SETUP]
26. INITIAL DP
27. DT [CALL PROCEEDING]
28. DCCH: DLDT [CALL PROCEEDING] <AM>
E.804(14)_F7-28c

Figure 7-28c



UE
Node B
RNC
MSC/VLR
MGW
29. BINDING ID, SPEECH CODEC TYPE, B PARTYROUTE
30. RAB ASSIGNMENT REQUEST
RANAP
RANAP
31. ESTABLISH REQUEST (AAL2)
ALCAP
ALCAP
32. ESTABLISH CONFIRM (AAL2)
ALCAP
ALCAP
33. RADIO LINK RECONFIG PREPARE
NBAP
NBAP
34. RADIO LINK RECONFIG READY
NBAP
NBAP
35. ESTABLISH REQUEST (AAL2)
ALCAP
ALCAP
36. ESTABLISH CONFIRM (AAL2)
ALCAP
ALCAP
37. RADIO LINK RECONFIG COMMIT
NBAP
NBAP
38. DCCH: RADIO BEARER SETUP <AM>
RRC
RRC
39. RAB ASSIGNMENT RESPONSE
RANAP
RANAP
40. DCCH: RADIO BEARER SETUP COMPLETE <AM>
RRC
RRC
ISUP
41. IAM
ISUP
42. ACM
43. DT [ALERTING]
RANAP
RANAP
44. DCCH: DLDT [ALERTING] <AM>
RRC
RRC
ISUP
45. ANS
46. DT [CONNECT]
RANAP
RANAP
47. DCCH: DLDT [CONNECT] <AM>
RRC
RRC
48. OPEN CONNECTION
49. DCCH: ULDT [CONNECT ACK] <AM>
RRC
RRC
50. DT [CONNECT ACK]
RANAP
RANAP
E.804(14)_F7-28d

Figure 7-28d

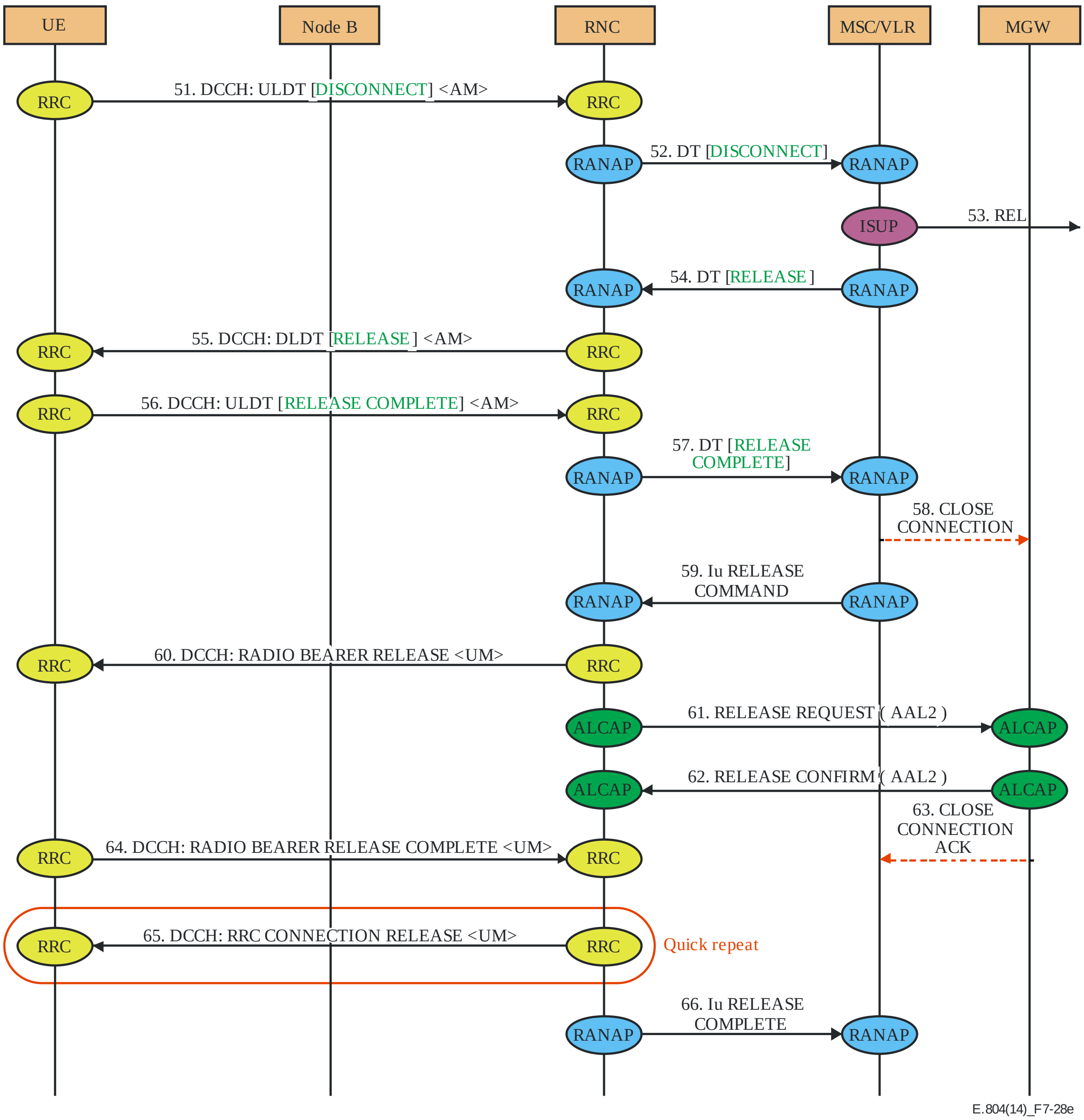


Figure 7-28e

Figure 7-28 – Video telephony signalling flow chart

7.3.8 Web browsing (HTTP)

7.3.8.1 HTTP Service non-accessibility [%]

7.3.8.1.1 Abstract definition

The HTTP service non-accessibility ratio is the probability that a subscriber cannot establish a PDP context and access the service successfully.

7.3.8.1.2 Abstract equation

$$\text{HTTP service non-accessibility}\,[\%] = \frac{\text{unsuccessful attempts to reach the point when content is received}}{\text{all attempts to reach the point when content is received}} \times 100$$

7.3.8.1.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Service access attempt	Start: User initiates the service access.	Start: ATD command.
Successful attempt	Stop: First content is received.	Stop method A: Reception of the first data packet containing the content. Stop method B: Sending of the first GET command.
Unsuccessful attempt	Stop trigger point not reached.	

Remark:

- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3).

7.3.8.2 HTTP set-up time [s]

7.3.8.2.1 Abstract definition

The HTTP set-up time is the time period needed to access the service successfully, from starting the dial-up connection to the point of time when the content is sent or received.

7.3.8.2.2 Abstract equation

$$\text{HTTP set-up time}\,[s] = (t_{\text{service access successful}} - t_{\text{service access start}})\,[s]$$

7.3.8.2.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{service access start}}$: Time of service access attempt	Start: User initiates the service access.	Start: ATD command.
$t_{\text{service access successful}}$: Time of successful service access	Stop: First content is received.	Stop method A: Reception of the first data packet containing content. Stop method B: Sending of the first GET command.

Remark:

- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3).

7.3.8.3 HTTP IP-service access failure ratio [%]

7.3.8.3.1 Abstract definition

The HTTP IP-service access ratio is the probability that a subscriber would not be able to establish a TCP/IP connection to the server of a service successfully.

7.3.8.3.2 Abstract equation

$$\text{HTTP IP} - \text{service access failure ratio}\,[\%] = \frac{\text{unsuccessful attempts to establish an IP connection to the server}}{\text{all attempts to establish an IP connection to the server}} \times 100$$

7.3.8.3.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
IP-Service access attempt	Start: User enters the URL and hits "Return".	Start: First [SYN] sent.
Successful attempt	Stop: Web page download starts.	Stop method A: Reception of the first data packet containing the content. Stop method B: Sending of the first GET command.
Unsuccessful attempt	Stop trigger point not reached.	

Remark:

- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.8.4 HTTP IP-service set-up time [s]

7.3.8.4.1 Abstract definition

The HTTP IP-service set-up time is the time period needed to establish a TCP/IP connection to the server of a service, from sending the initial query to a server to the point of time when the content is sent or received.

7.3.8.4.2 Abstract equation

$$\text{HTTP IP} - \text{service set-up time}\,[s] = \left(t_{\text{IP-Service access successful}} - t_{\text{IP-Service access start}}\right)[s]$$

7.3.8.4.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{IP-Service access start}}$: Time of IP-Service access attempt	Start: User enters the URL and hits "Return".	Start: First [SYN] sent.
$t_{\text{IP-Service access successful}}$: Time of successful IP-service access	Stop: Web page download starts.	Stop method A: Reception of the first data packet containing the content. Stop method B: Sending of the first GET command.

Remark:

- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.8.5 HTTP session failure ratio [%]

7.3.8.5.1 Abstract definition

The HTTP session failure ratio is the proportion of uncompleted sessions and sessions that were started successfully.

7.3.8.5.2 Abstract equation

$$\text{HTTP session failure ratio}\,[\%] = \frac{\text{uncompleted sessions}}{\text{successfully started sessions}} \times 100$$

7.3.8.5.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started session	Start: User enters the URL and hits "Return".	Start: First [SYN] sent.
Completed session	Stop: The complete web page appears in the browser window.	Stop: Reception of the last data packet containing content.
Uncompleted session	Stop trigger point not reached.	

Remark:

- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.8.6 HTTP session time [s]

7.3.8.6.1 Abstract definition

The HTTP session time is the time period needed to successfully complete a PS data session.

7.3.8.6.2 Abstract equation

$$\text{HTTP session time}\,[s] = (t_{\text{session end}} - t_{\text{session start}})\,[s]$$

7.3.8.6.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{session start}}$: Time of successfully started session	Start: User enters the URL and hits "Return".	Start: First [SYN] sent.
$t_{\text{session end}}$: Time when session completed	Stop: The complete web page appears in the browser window.	Stop: Reception of the last data packet containing the content.

Remark:

- The PS bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.8.7 HTTP mean data rate [kbit/s]

7.3.8.7.1 Abstract definition

After a data link has been successfully established, this parameter describes the average data transfer rate measured throughout the entire connect time to the service. The data transfer shall be successfully terminated. The prerequisite for this parameter is network and service access.

7.3.8.7.2 Abstract equation

$$\text{HTTP mean data rate}\,[\text{kbit/s}] = \frac{\text{user data transferred}\,[\text{kbit}]}{\left(t_{\text{data transfer complete}} - t_{\text{data transfer start}}\right)[\text{s}]}$$

7.3.8.7.3 Trigger points

The average throughput is measured from opening the data connection to the end of the successful transfer of the content (web page).

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{data transfer start}}$: Time of successfully started data transfer	Start: Web page download starts.	Start method A: Reception of the first data packet containing the content. Start method B: Sending of the first GET command.
$t_{\text{data transfer complete}}$: Time when data transfer is complete	Stop: Web page download successfully completed.	Stop: Reception of the last data packet containing the content.

Remark:

- The mobile station is already attached (see clause 7.2.3), a PDP context is activated (see clause 7.2.5) and a service was accessed successfully (see service non-accessibility).

7.3.8.8 HTTP data transfer cut-off ratio [%]

7.3.8.8.1 Abstract definition

The HTTP data transfer cut-off ratio is the proportion of incomplete data transfers and data transfers that were started successfully.

7.3.8.8.2 Abstract equation

$$\text{HTTP data transfer cut-off ratio}\,[\%] = \frac{\text{incomplete data transfers}}{\text{successfully started data transfers}} \times 100$$

7.3.8.8.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started data transfer	Start: Web page download starts.	Start method A: Reception of the first data packet containing the content. Start method B: Sending of the first GET command.
Complete data transfer	Stop: Web page download is successfully completed.	Stop: Reception of the last data packet containing the content.
Incomplete data transfer	Stop trigger point not reached.	

Remark:

- The mobile station is already attached (see clause 7.2.3), a PDP context is activated (see clause 7.2.5) and a service was accessed successfully (see service non-accessibility).

7.3.9 Web radio

7.3.9.1 General

Web radio is a term used for different types of audio streaming. Most popular, according to current perception, is the proprietary but de-facto-standard Shoutcast (www.shoutcast.com) type which is used by WinAmp and America online (AOL). There is an open source variant (Icecast, www.icecast.org). The following descriptions refer to Shoutcast, if not mentioned otherwise.

A typical web radio basic scenario starts with starting up the respective client's web radio functionality.

First step is retrieval of an electronic programme guide (EPG), typically in the form of a station list naming station name, genre of content offered by this station, and stream rate (which gives users a hint on expected audio quality). This EPG is typically retrieved from a fixed-URL server, e.g., www.shoutcast.com.

NOTE – Remark: For other clients and types of web radio, clients such as Microsoft Windows media player or Apple's iTunes accessing respective portals are used.

Next step is selection of a station from the list. This triggers an attempt to open the respective stream and start to receive content. Typically, before audio reproduction starts, the client will do some seconds of buffering.

7.3.9.2 Preconditions

With reference to the technical description above, the following key performance indicator

(KPI) belong to the basic scenario only which is characterized as follows:

- EPG retrieval is not part of the scenario (because in typical listening situations this is done once for multiple-station access). It is assumed that the station ID is already known.
- EPG retrieval can be seen, however, as a kind of scenario extension.

7.3.9.3 Special remarks on Internet radio audio playback and buffering

Characteristic for Internet radio audio playback is the fact that with a typical client application, no quality impairment other than gaps in reproduction occur. In other words, there is no poor MOS value or other continuous quality indicator, but simply "silence" for a period of time which cannot be estimated by the user. This fact is important when it comes to the definition of a useful KPI for audio quality.

Since the service is TCP-based and uses buffering, playback will continue until the buffer is empty. The buffer has a fixed maximum size, equalling a constant maximum playback time. If the buffer is full, the whole mechanism can be modelled by a simple differential model where the new data flows in with a network-dependent data rate and flows out with a constant rate (playback stream rate).

In the stationary case with buffer completely filled, incoming throughput is equal to playback stream rate, independent of the maximum throughput the network can deliver. If the buffer is less than full due to a previous drop in incoming data rate, incoming data rate will be higher (at the maximum throughput the network/IP level chain can deliver at this time) until the buffer is full again.

7.3.9.4 Transaction definition from user's perspective

A web radio transaction consists of a single tune-in to a selected station, followed by music playback for a given time.

- It is assumed that all servers being accessed (tune-in information server, stream server) are basically accessible and have sufficient downstream bandwidth.
- It is assumed that the length of the tune-in list is not relevant for KPI precision under given conditions (time effects caused by different lengths of tune-in list to be negligible).

7.3.9.5 Result definition

With respect to the technical description, a full web radio transaction has one of the following results.

Result	Definition
Successful	At least one packet of content was successfully received, and no timeout condition occurred up to the end of the scheduled playback time.
Dropped	The conditions for successful transaction were not met. Examples: Unsuccessful access to the tune-in or stream server, loss of Internet connection during playback, or a gap in playback longer than a pre-defined timeout value.

It shall be noted that according to this definition, a web radio transaction where effectively no useable audio playback was possible is still considered to be technically successful. It is assumed that the fact that the transaction was useless and probably most annoying to the user is reflected in another QoS describing subjective quality. Therefore, the situation is qualitatively equivalent to a technically stable speech telephony call with extremely poor audio MOS score.

There is no "Failed" result because it is assumed that all phases of the transaction are part of service usage, and the impact of unsuccessful phases is equally negative in the user's perception. Failure therefore is always attributed to the earlier phase such as establishment of basic Internet access, or DNS access.

7.3.9.6 QoS parameter overview

The following shows phases in web radio usage and the coverage by the defined QoS parameters.

Phase (user perspective)	Retrieve EPG	Select the station	Listen to the selected station	
Phase (KPI coverage)	EPG Retrieval	Tune-in	Reproduction set-up	
			Reproduction	

Please note that for the sake of "user perspective" reproduction set-up and reproduction are not seamlessly connected. Reproduction setup QoS parameters are provided for diagnostic purposes.

7.3.9.7 Web radio EPG retrieval failure ratio [%]

7.3.9.7.1 Abstract definition

The web radio EPG retrieval failure ratio parameter is the probability that a subscriber cannot access the web radio EPG successfully.

7.3.9.7.2 Abstract equation

$$\text{Web radio EPG retrieval failure ratio}\,[\%] = \frac{\text{unsuccessful attempts to access the EPG}}{\text{all attempts to access the EPG}} \times 100$$

7.3.9.7.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
EPG retrieval attempt	Start: User accesses the web radio EPG.	Start: HTTP GET on EPG URL.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful attempt	Stop: EPG content is successfully received.	Stop: Successful reception of the EPG content (HTTP 200 OK, eventually followed by additional blocks).
Unsuccessful attempt	Stop trigger point not reached.	

7.3.9.8 Web radio EPG retrieval time [s]

7.3.9.8.1 Abstract definition

The web radio EPG retrieval time parameter describes the time period needed to access the web radio EPG successfully.

7.3.9.8.2 Abstract equation

$$\text{Web radio EPG retrieval time}\,[\text{s}] = (t_{\text{Stop\_ER}} - t_{\text{Start\_ER}})\,[\text{s}]$$

7.3.9.8.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t_{Start_ER}: Time of EPG retrieval attempt	Start: User accesses the web radio EPG.	Start: Time of sending the HTTP GET on the EPG URL.
t_{Stop_ER}: Time of successful EPG retrieval attempt	Stop: EPG content is successfully received.	Stop: Time of successful reception of the EPG content (HTTP 200 OK, eventually followed by additional blocks).

7.3.9.9 Web radio tune-in failure ratio [%]

7.3.9.9.1 Abstract definition

The web radio tune-in failure ratio parameter is the probability that a subscriber cannot obtain the tune-in information for a web radio streaming server successfully.

7.3.9.9.2 Abstract equation

$$\text{Web radio tune-in failure ratio}\,[\%] = \frac{\text{unsuccessful tune-in attempts}}{\text{all tune-in attempts}} \times 100$$

7.3.9.9.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Tune-in attempt	Start: Attempt to retrieve the tune-in information.	Start: Obtain the tune-in information via a HTTP GET to a location obtained from the EPG.
Successful attempt	Stop: Receive the tune-in information.	Stop: Successful reception of the tune-in information (HTTP 200 OK, eventually followed by additional blocks).
Unsuccessful attempt	Stop trigger point not reached.	

7.3.9.10 Web radio tune-in time [s]

7.3.9.10.1 Abstract definition

The web radio tune-in time parameter describes the time period needed to obtain the tune-in information for a web radio streaming server successfully.

7.3.9.10.2 Abstract equation

$$\text{Web radio tune-in time}[s] = (t_{Stop\_TI} - t_{Start\_TI})[s]$$

7.3.9.10.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t_{Start_TI}: Time of the tune-in attempt	Start: Attempt to retrieve the tune-in information.	Start: Time when HTTP GET is issued to a location obtained from the EPG.
t_{Stop_TI}: Time of successful tune-in attempt	Stop: Receive the tune-in information.	Stop: Time of successful reception of the tune-in information (HTTP 200 OK, eventually followed by additional blocks).

7.3.9.11 Web radio reproduction set-up failure ratio [%]

7.3.9.11.1 Abstract definition

The web radio reproduction set-up failure ratio parameter is the probability that a subscriber cannot successfully start listening to a given web radio station.

7.3.9.11.2 Abstract equation

$$\text{Web radio reproduction set-up failure ratio}[\%] = \frac{\text{unsuccessful reproduction set-up attempts}}{\text{all reproduction set-up attempts}} \times 100$$

7.3.9.11.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Reproduction set-up attempt	Start: Attempt to retrieve audio stream.	Start: Attempt to retrieve audio content from stream server listed in the tune-in information (HTTP GET).
Successful reproduction set-up attempt	Stop: Indication that player starts buffering (may not be visible in all players).	Stop: Reception of the first block of content (audio data).
Unsuccessful attempt	Stop trigger point not reached.	

7.3.9.12 Web radio reproduction set-up time [s]

7.3.9.12.1 Abstract definition

The web radio reproduction set-up time parameter describes the time period from request of audio stream from stream server to reception of the first data packet of audio content.

Remark:

- Actual start of reproduction from user's point of view will be this time plus the buffer-fill time which may be specific to a web radio client application.

7.3.9.12.2 Abstract equation

$$\text{Web radio reproduction set-up time}[s] = (t_{Stop\_RP} - t_{Start\_RP})[s]$$

7.3.9.12.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t_{Start_RP}: Time of stream reproduction attempt	Start: Indication that the audio stream is requested from the server.	Start: Time when the HTTP GET message is issued to the stream server.
t_{Stop_RP}: Time of reception of the first data packet of the audio content	Stop: Indication that buffering of the content begins.	Stop: Receive the first encoded audio data (client application is buffering).

Remark:

- Indicators listed under "user's point of view", may not be shown by actual web radio client applications.

7.3.9.13 Web radio reproduction cut-off ratio [%]

7.3.9.13.1 Abstract definition

The web radio reproduction cut-off ratio parameter is the probability that a subscriber cannot successfully complete stream reproduction from a given web radio station for a given period of time.

Remark:

- Typically, web radio client applications use buffering; therefore actual audible reproduction will start at a certain time after the reception of the first data packet. This parameter covers the whole reproduction time, starting from reception of the first data packet to avoid making assumptions for the buffer length.

7.3.9.13.2 Abstract equation

$$\text{Web radio reproduction cut-off ratio}[\%] = \frac{\text{unsuccessful listening attempts}}{\text{all listening attempts}} \times 100$$

7.3.9.13.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Listening attempt	Start: Attempt to retrieve the audio stream.	Start: Attempt to retrieve the audio content from the stream server listed in the tune-in information (HTTP GET).
Successful listening attempt	Stop: Reach the end of intended stream playback time without break in IP connection.	Stop: Reach the end of intended stream playback time without break in IP connection.
Unsuccessful attempt	Stop trigger point not reached.	

7.3.9.14 Web radio audio quality

Due to the nature of web radio which is using TCP connections, expected degradation effects are audio "gaps" (silence) only, resulting in buffer-empty condition resulting from insufficient bandwidth.

At this point in time, no commonly accepted definition of perceived audio quality under these conditions exists. It is clear that for such a perceptual measure, all aspects of possible audio gaps need to be taken into account, namely:

- gap duration;
- frequency of gaps;
- time between gaps.

For the time being, it is recommended to report the basic data on gaps on an event basis only.

However, codec and stream rate (encoded bit rate) needs to be part of the measurement definition since it will have decisive impact on results.

7.3.10 WLAN service provisioning with HTTP based authentication

7.3.10.1 Generic signal flow

Figure 7-29 shows general signal flow for WLAN service provisioning with HTTP based authentication.

KPI legend:

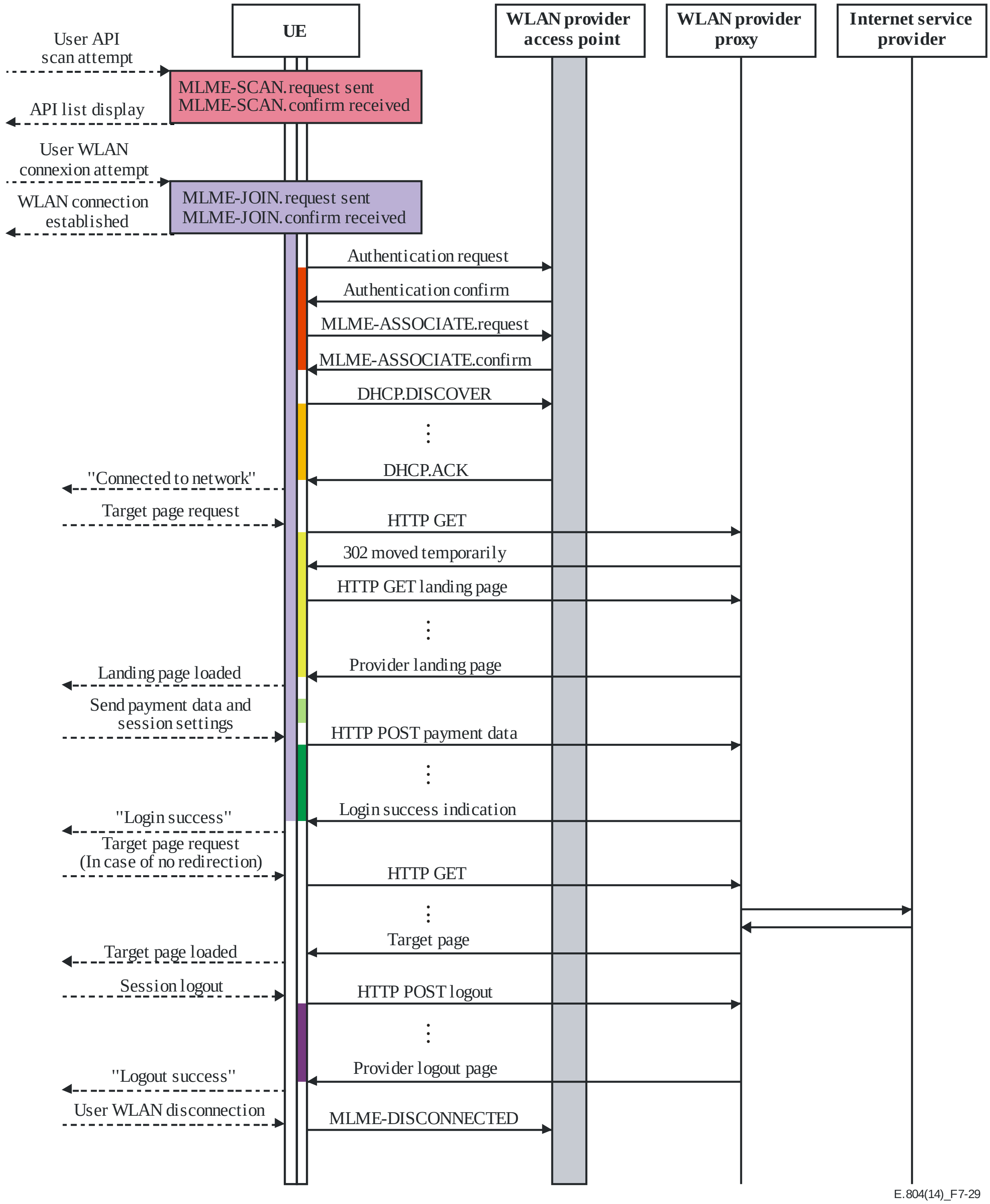


Figure 7-29 – Generic signal flow

7.3.10.2 WLAN scan failure ratio [%]

7.3.10.2.1 Abstract definition

The WLAN scan failure ratio is the probability that no desired active access points (APs) could be found in an area where WLAN is present.

7.3.10.2.2 Abstract equation

$$\text{WLAN scan failure ratio}\,[\%] = \frac{\text{unsuccessful scan attempts}}{\text{total attempts to scan WLAN APs}} \times 100$$

7.3.10.2.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Scan attempt	Start: User attempts to scan for available APs	Start: First "MLME-SCAN.request" containing the target service set identifier (SSID) is sent
Successful scan attempt	Stop: List of available APs is displayed including desired SSID	Stop: "MLME-SCAN.confirm containing the target SSID received
Unsuccessful scan attempt	Stop trigger not reached	

Preconditions for measurement:

- It is possible that a scan to all access points in the area (broadcast) is answered by an access point other than the desired one. To make sure that only the correct access point answers, the scan request shall contain the desired service set identifier (SSID).
- Usually, operating systems keep a list of preferred access points and sporadically scan for these access points automatically. These automated scans shall be deactivated and the list shall be kept empty.

For further study: It should be analysed if the time to scan can vary depending on the applied scan method, i.e., if an aimed scan with the target operator's SSID leads to faster/slower confirmation than a broadcast scan to all access points in the area.

7.3.10.3 WLAN time to scan [s]

7.3.10.3.1 Abstract definition

The WLAN time to scan denotes the time it takes to scan for available access points.

Figure 7-30 shows scan signal flow.

7.3.10.3.2 Abstract equation

$$\text{WLAN time to scan}\,[\text{s}] = \left(t_{\text{Scan result received}} - t_{\text{Scan started}}\right)[\text{s}]$$

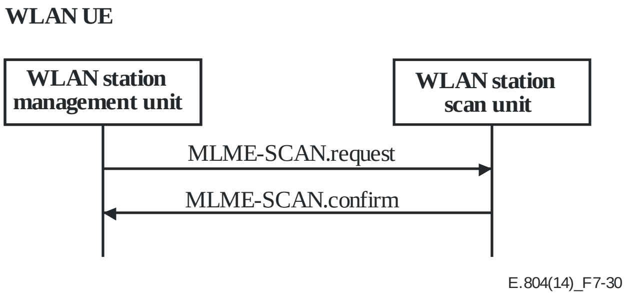


Figure 7-30 – Scan signal flow

7.3.10.3.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{Scan\ started}$: Time of scan attempt	Start: User attempts to scan for available APs.	Start: First "MLME-SCAN.request" containing the target SSID is sent.
$t_{Scan\ result\ received}$: Time of successful scan attempt	Stop: List of available APs is displayed, including the target SSID.	Stop: "MLME-SCAN.confirm" containing the target SSID is received.

Preconditions for measurement:

- It is possible that a scan to all access points in the area (broadcast) is answered by another access point than the desired one. To make sure that only the correct access point answers, the scan request shall contain the desired SSID.
- Usually, operating systems keep a list of preferred access points and sporadically scan for these access points automatically. These automated scans shall be deactivated and the list shall be kept empty.

NOTE – The authorization time that is consumed for entering and receiving the password has an effect on the time to scan.

For further study: It should be analysed if the time to scan can vary depending on the applied scan method, i.e., if an aimed scan with the target operator's SSID leads to faster/slower confirmation than a broadcast scan to all access points in the area.

7.3.10.4 WLAN PS data service provisioning failure ratio [%]

7.3.10.4.1 Abstract definition

The WLAN PS data service provisioning failure ratio is the probability that a user cannot get in position to access services in a WLAN area.

7.3.10.4.2 Abstract equation

$$\text{WLAN PS data service provisioning failure ratio}[\%] = \frac{\text{unsuccessful connect attempts}}{\text{all connect attempts}} \times 100$$

7.3.10.4.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Connect attempt	Start: User attempts to connect to the wireless network.	Start: First "MLME-JOIN.request" is sent.
Successful connect attempt	Stop: Authorization confirmed by receiving the login success indication.	Stop: Reception of the first data packet of a page indicating login success.
Unsuccessful connect attempt	Stop trigger not reached.	

NOTE 1 – After authorization, some operators will automatically redirect the user to the uniform resource locator (URL) that was entered in the initial portal access attempt which led to the landing page redirection. Other operators display a login success page of sorts and do not redirect users to their initially entered URL.

NOTE 2 – The implicit authorization failure ratio also depends on the authorization method, e.g., voucher received by SMS versus credit card. Thus, measurements based on different authorization method cannot be compared.

7.3.10.5 WLAN PS data service provisioning time [s]

7.3.10.5.1 Abstract definition

The WLAN PS data service provisioning time denotes the time it takes until the user is authorized in WLAN and in position to access services.

7.3.10.5.2 Abstract equation

$$\text{WLAN PS data service provisioning time}[s] = \left(t_{\text{Target URL received}} - t_{\text{Connect option selected}}\right)[s]$$

Figures 7-31 and 7-32 show signal flows for Join and WLAN PS data service provisioning respectively.

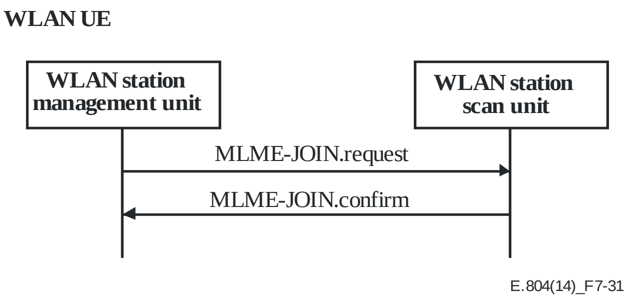


Figure 7-31 – Join signal flow

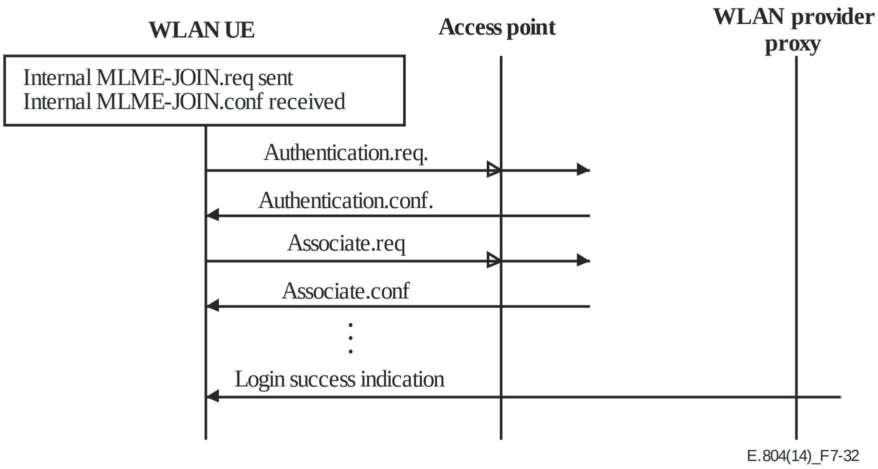


Figure 7-32 – WLAN PS data service provisioning signal flow

7.3.10.5.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{Connect option selected}}$: Time of connect attempt	Start: User attempts to connect to wireless network.	Start: First "MLME-JOIN.request" is sent.
$t_{\text{Target URL received}}$: Time of successful connect attempt	Stop: Authorization confirmed by receiving the login success indication.	Stop: Reception of the first data packet of a page indicating the login success.

NOTE 1 – After authorization, some operators will automatically redirect the user to the URL that was entered in the initial portal access attempt which led to the landing page redirection. Other operators display a login success page of sorts and do not redirect users to their initially entered URL.

NOTE 2 – The implicit authorization time also depends on the authorization method, e.g., voucher received by SMS versus credit card. Thus, measurements based on different authorization method cannot be compared.

NOTE 3 – The implicit authorization time that is consumed for entering and receiving the password has an effect on the PS data service provisioning time.

7.3.10.6 WLAN association failure ratio [%]

7.3.10.6.1 Abstract definition

The WLAN association failure ratio is the probability that a user cannot establish a radio link with the chosen access point.

7.3.10.6.2 Abstract equation

$$\text{WLAN association failure ratio}[\%] = \frac{\text{unsuccessful association attempts}}{\text{all association attempts}} \times 100$$

7.3.10.6.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Association attempt	Start: User attempts to connect to the wireless network.	Start: First "MLME-JOIN.request" is sent.
Successful association attempt	Stop: Connection to the access point is established and displayed.	Stop: "MLME-ASSOCIATE.confirm" received with status code "success".
Unsuccessful association attempt	Stop trigger not reached.	

7.3.10.7 WLAN association time [s]

7.3.10.7.1 Abstract definition

The WLAN association time denotes the time it takes to associate with the chosen access point.

Figure 7-33 shows WLAN association signal flow.

7.3.10.7.2 Abstract equation

$$\text{WLAN association time}[s] = \left(t_{\text{Successful association}} - t_{\text{Association start}}\right)[s]$$

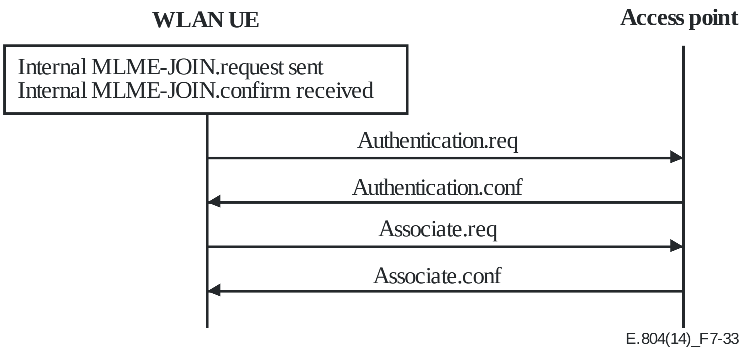


Figure 7-33 – WLAN association signal flow

7.3.10.7.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{Association start}}$: Time of association attempt	Start: User attempts to connect to the wireless network.	Start: First "MLME-JOIN.request" is sent.
$t_{\text{Successful association}}$: Time of successful association attempt	Stop: Connection to access point is established and displayed.	Stop: "MLME-ASSOCIATE.confirm" received with status code "success".

NOTE – The authorization time that is consumed for entering and receiving the password has an effect on the association time.

7.3.10.8 WLAN IP address allocation failure ratio [%]

7.3.10.8.1 Abstract definition

The WLAN IP address allocation failure ratio is the probability that a user is not allocated an IP address by the access point.

7.3.10.8.2 Abstract equation

$$\text{WLAN IP address allocation failure ratio}[\%] = \frac{\text{unsuccessful attempts to allocate IP address}}{\text{all IP address allocation requests}} \times 100$$

7.3.10.8.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
IP address allocation request	Start: Attempt to acquire the network address and display of the status.	Start: First "DHCP.DISCOVER" is sent.
Successful attempt to allocate the IP address	Stop: Connection to the network is established and displayed.	Stop: "DHCP.ACK" received with valid IP address.
Unsuccessful attempt to allocate the IP address	Stop trigger not reached.	

7.3.10.9 WLAN IP address allocation time [s]

7.3.10.9.1 Abstract definition

The WLAN IP address allocation time denotes the time it takes the access point to allocate an IP address to the user's system.

7.3.10.9.2 Abstract equation

$$\text{WLAN IP address allocation time}[s] = (t_{\text{IP reception}} - t_{\text{IP allocation start}})[s]$$

7.3.10.9.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{IP allocation start}}$: Time of the IP address allocation request	Start: Attempt to acquire the network address and display of the status.	Start: First "DHCP.DISCOVER" is sent.
$t_{\text{IP reception}}$: Time of successful attempt to allocate IP address	Stop: Connection to the network is established and displayed.	Stop: "DCHP.ACK" received with valid IP address.

NOTE – The authorization time that is consumed for entering and receiving the password has an effect on the IP address allocation time.

7.3.10.10 WLAN landing page download failure ratio [%]

7.3.10.10.1 Abstract definition

The WLAN landing page download failure ratio is the probability that the landing page to which a user will be redirected for login to the WLAN cannot be successfully downloaded after requesting the target page.

7.3.10.10.2 Abstract equation

$$\text{WLAN landing page download failure ratio}[\%] = \frac{\text{unsuccessful landing page down download attempts}}{\text{all landing page download attempts}} \times 100$$

7.3.10.10.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Landing page download attempt	Start: User enters the target URL and requests the desired page.	Start: "HTTP_GET" for the target page is sent.
Successful landing page download attempt	Stop: Landing page download is finished.	Stop: Last HTTP data packet of the landing page is received.
Unsuccessful landing page download attempt	Stop trigger not reached.	

Preconditions for measurement:

- The measurement system shall be disconnected from the WLAN prior to each measurement cycle.
- The cache shall be emptied prior to each measurement cycle and keep alive shall be deactivated/suppressed.

7.3.10.11 WLAN landing page download time [s]

7.3.10.11.1 Abstract definition

The WLAN landing page download time denotes the time it takes for redirection and download of the landing page provided to login to the WLAN successfully, after the user has tried to access some webpage.

7.3.10.11.2 Abstract equation

$$\text{WLAN landing page download time[s]} = \left(t_{\text{Landing page successfully downloaded}} - t_{\text{Webpage request sent}}\right)[s]$$

7.3.10.11.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{Webpage request sent}}$: Time of the landing page download attempt	Start: User enters the target URL and requests the desired page.	Start: "HTTP_GET" for the target page is sent.
$t_{\text{Landing page successfully downloaded}}$: Time of successful landing page download attempt	Stop: Landing page download is finished.	Stop: Last HTTP data packet of the landing page is received.

Preconditions for measurement:

- The measurement system shall be disconnected from the WLAN prior to each measurement cycle.
- The cache shall be emptied prior to each measurement cycle and keep alive shall be deactivated/suppressed.

NOTE – The authorization time that is consumed for entering and receiving the password has an effect on the landing page download time.

7.3.10.12 WLAN landing page password retrieval failure ratio [%]

7.3.10.12.1 Abstract definition

The WLAN landing page password retrieval failure ratio denotes the probability that the password to get submitted via the landing page is not received by the user.

7.3.10.12.2 Abstract equation

$$\text{WLAN landing page password retrieval failure ratio}\,[\%] = \frac{\text{unsuccessful password retrieval attempts}}{\text{all password retrieval attempts}} \times 100$$

7.3.10.12.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Password retrieval attempt	Start: Authorization form filled in and submitted.	Start: "TCP SYN."
Successful password retrieval attempt	Stop: Depending on the used service, e.g., SMS with the password received successfully.	Stop: Depending on used service, e.g., SMS with the password received successfully.
Unsuccessful password retrieval attempt	Stop trigger not reached.	

NOTE – The password retrieval failure ratio can be neglected when the credit card payment method is used.

7.3.10.13 WLAN landing page password retrieval time [s]

7.3.10.13.1 Abstract definition

The WLAN landing page password retrieval time denotes the time it takes to request and receive a password to get submitted via the landing page.

7.3.10.13.2 Abstract equation

$$\text{WLAN landing page password retrieval time}[s] = \left(t_{\text{Password received}} - t_{\text{Authorization request submitted}}\right)[s]$$

7.3.10.13.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{Authorization request submitted}}$: Time of password retrieval attempt	Start: Authorization form filled in and is submitted.	Start: "TCP SYN".
$t_{\text{Password received}}$: Time of successful password retrieval attempt	Stop: Depending on the used service, e.g., SMS with the password received successfully.	Stop: Depending on the used service, e.g., SMS with the password received successfully.

NOTE 1 – The password retrieval time can be neglected when the credit card payment method is used.

NOTE 2 – The authorization time that is consumed for entering and receiving the password has an effect on the landing page password retrieval time.

7.3.10.14 WLAN landing page authorization failure ratio [%]

7.3.10.14.1 Abstract definition

The WLAN landing page authorization failure ratio is the probability that the user authorization process via the landing page is not successful.

7.3.10.14.2 Abstract equation

$$\text{WLAN landing page authorization failure ratio}\,[\%] = \frac{\text{unsuccessful authorization attempts}}{\text{all authorization attempts}} \times 100$$

7.3.10.14.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Authorization attempt	Start: Password or payment data is submitted.	Start: "HTTP POST" is sent.
Successful authorization attempt	Stop: Authorization confirmed by receiving the login success indication.	Stop: Reception of the first data packet of a page indicating the login success.
Unsuccessful authorization attempt	Stop Trigger not reached.	

NOTE 1 – After authorization, some operators will automatically redirect the user to the URL that was entered in the initial portal access attempt which led to the landing page redirection. Other operators will display a login success page of sorts and not redirect the user to the initially entered URL.

NOTE 2 – The authorization failure ratio also depends on the authorization method, e.g., voucher received by SMS versus credit card. Thus, measurements based on different authorization method cannot be compared.

7.3.10.15 WLAN landing page authorization time [s]

7.3.10.15.1 Abstract definition

The WLAN landing page authorization time denotes the time it takes to perform user authorization via the landing page.

7.3.10.15.2 Abstract equation

$$\text{WLAN landing page authorization time}[s] = (t_{\text{Authorization confirmed}} - t_{\text{Password is submitted}})[s]$$

7.3.10.15.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{Password is submitted}}$: Time of authorization attempt	Start: Password or payment data is submitted.	Start: "HTTP POST" is sent.
$t_{\text{Authorization confirmed}}$: Time of successful authorization attempt	Stop: Authorization confirmed by receiving the login success indication.	Stop: Reception of the first data packet of a page indicating the login success.

NOTE 1 – After authorization, some operators will automatically redirect the user to the URL that was entered in the initial portal access attempt which led to the landing page redirection. Other operators display a login success page of sorts and do not redirect users to their initially entered URL.

NOTE 2 – The authorization time also depends on the authorization method, e.g., voucher received by SMS versus credit card. Thus, measurements based on different authorization method cannot be compared.

NOTE 3 – The authorization time that is consumed for entering and receiving the password has an effect on the landing page authorization time.

7.3.10.16 WLAN re-accessibility failure ratio [%]

7.3.10.16.1 Abstract definition

The WLAN re-accessibility failure ratio is the probability that re-accessing the access point is not successful because of a WLAN failure.

7.3.10.16.2 Abstract equation

$$\text{WLAN re-accessibility failure ratio}[\%] = \frac{\text{unsuccessful attempts reaccess}}{\text{all attempts to reaccess}} \times 100$$

7.3.10.16.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Attempt to reaccess	Start: Access point is displayed in the list of available access points.	Start: First "MLME-ASSOCIATE.request" is sent after radio signal is sufficient again.
Successful attempt to reaccess	Stop: Message that the WLAN adapter is ready (MAC address of AP is available).	Stop: "MLME-ASSOCIATE.confirm" has been received with the status code "success".
Unsuccessful attempt to reaccess	Stop trigger point not reached.	

7.3.10.17 WLAN re-accessibility time [s]

7.3.10.17.1 Abstract definition

The WLAN re-accessibility time denotes the time it takes to re-establish a lost radio link with the access point after the signal strength is sufficient again.

Figure 7-34 shows WLAN re-association signal flow.

7.3.10.17.2 Abstract equation

$$\text{WLAN re-accessibility time}[s] = \left(t_{\text{AP's MAC address is available}} - t_{\text{AP reappears in list}}\right)[s]$$

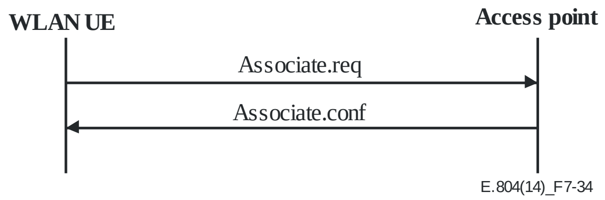


Figure 7-34 – WLAN re-association signal flow

7.3.10.17.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{AP reappears in list}}$: Time of attempt to reaccess	Start: Access point is displayed in the list of available access points.	Start: First "MLME-ASSOCIATE.request" is sent after radio signal is sufficient again.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{AP"s MAC address is available}}$: Time of successful attempt to reaccess	Stop: Message that the WLAN adapter is ready.	Stop: "MLME-ASSOCIATE.confirm" has been received with the status code "success".

NOTE – The authorization time that is consumed for entering and receiving the password has an effect on the re-accessibility time.

7.3.10.18 WLAN logout page download failure ratio [%]

7.3.10.18.1 Abstract definition

The WLAN logout page download failure ratio is the probability that the logout process is not successful.

7.3.10.18.2 Abstract equation

$$\text{WLAN logout page download failure ratio}[\%] = \frac{\text{unsuccessful logout page download attempts}}{\text{all logout page download attempts}} \times 100$$

7.3.10.18.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Logout page download attempt	Start: Decision to logout is submitted.	Start: "HTTP POST" is sent.
Successful logout page download attempt	Stop: Logout is confirmed by receiving the logout page.	Stop: Reception of the first data packet of the logout page.
Unsuccessful logout page download attempt	Stop Trigger not reached.	

7.3.10.19 WLAN logout page download time [s]

7.3.10.19.1 Abstract definition

The WLAN logout page download time denotes the time it takes to perform user logout.

7.3.10.19.2 Abstract equation

$$\text{WLAN logout page download time}[s] = \left(t_{\text{Logout confirmed}} - t_{\text{Logout procedure start}}\right)[s]$$

7.3.10.19.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{Logout procedure start}}$: Time of logout page download attempt	Start: Decision to logout is submitted.	Start: "HTTP POST" is sent.
$t_{\text{Logout confirmed}}$: Time of successful logout page download attempt	Stop: Logout confirmed by receiving the logout page.	Stop: Reception of the first data packet of the logout page.

NOTE – The authorization time that is consumed for entering and receiving the password has an effect on the logout page download time.

7.3.11 Wireless application protocol (WAP)

The wireless application protocol (WAP) is a specification for a set of communication protocols to standardize the way that wireless devices, such as cellular telephones and radio transceivers, can be used for Internet access, including e-mail, the world wide web (WWW), newsgroups, and instant messaging. Devices and service systems that use WAP are able to interoperate.

The WAP layers are:

- Wireless application environment (WAE).
- Wireless session layer (WSL).
- Wireless transport layer security (WTLS).
- Wireless transport layer (WTP).

WAP is a technology designed to allow efficient transmission of optimized Internet content to cellular telephones.

The QoS parameters for WAP are represented in Figure 7-35.

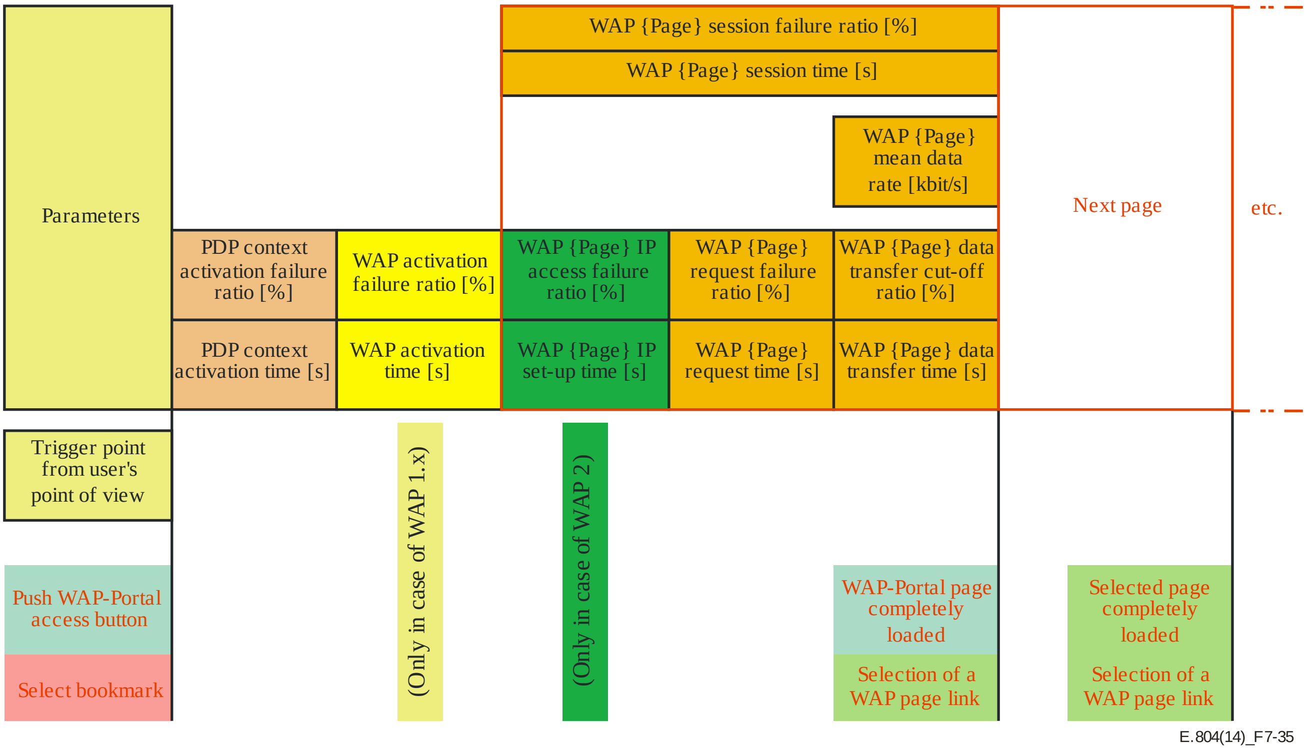


Figure 7-35 – Parameter and service overview

Technical description and protocol part of the parameters for the whole clause are represented in Figure 7-36.

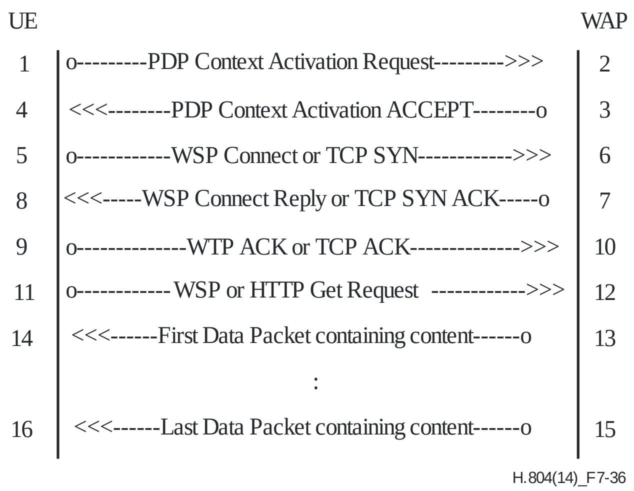


Figure 7-36 – WAP message sequence chart

NOTE – WSP connection usually occurs once per session, TCP connection is more frequent.

7.3.11.1 WAP activation failure ratio [%] (WAP 1.x only)

7.3.11.1.1 Abstract definition

The parameter WAP activation failure ratio is the probability that the WAP session could not be activated in case of WAP 1.x connection-mode session service.

7.3.11.1.2 Abstract equation

$$\text{WAP activation failure ratio}\,[\%] = \frac{\text{unsuccessful WAP activation attempts}}{\text{all WAP activation attempts}} \times 100$$

7.3.11.1.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
WAP activation attempt	Not applicable.	Start: WSP Connect procedure.
Successful WAP activation attempt	Not applicable.	Stop: Reception of the WSP Connect Reply.
Unsuccessful WAP activation attempt	Stop trigger point not reached.	

Remark:

- The bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3).

7.3.11.2 WAP activation time [s] (WAP 1.x only)

7.3.11.2.1 Abstract definition

The parameter WAP activation time describes the time it takes to activate the WAP session in case of WAP 1.x connection-mode session service.

7.3.11.2.2 Abstract equation

$$\text{WAP activation time}[s] = \left(t_{\text{WAPsessionestablished}} - t_{\text{WAPsessionactivationrequest}}\right)[s]$$

7.3.11.2.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{WAP session activation request}}$: Time of WAP session activation request.	Not applicable.	Start: WSP Connect procedure
$t_{\text{WAP session established}}$: Time when WAP session is established.	Not applicable.	Stop: Reception of the WSP Connect Reply

Remark:

- The bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3). Only successful measurements are taken into account to calculate the average time.

7.3.11.3 WAP (page) IP access failure ratio [%] (WAP 2.x only)

7.3.11.3.1 Abstract definition

The parameter WAP (Page) IP access failure ratio denotes the probability that a subscriber cannot establish a TCP/IP connection to the WAP server successfully.

NOTE – This parameter can only be calculated in case of follow up page, if the TCP/IP connection is not persistent.

7.3.11.3.2 Abstract equation

$$\text{WAP (page) IP access failure ratio}\,[\%] = \frac{\text{unsuccessful WAP IP Access attempts}}{\text{all WAP IP Access attempts}} \times 100$$

7.3.11.3.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
WAP IP access attempt	Start: Selecting the link of a WAP page or applying an entered URL	Start: Sending of the first TCP SYN
Successful WAP IP access attempt	Not applicable	Stop: Sending of the first HTTP GET command
Unsuccessful WAP IP access attempt	Stop trigger point not reached	

Remark:

- The bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.11.4 WAP (page) IP access set-up time [s] (WAP 2.x only)

7.3.11.4.1 Abstract definition

The WAP (page) IP access set-up time is the time period needed to establish a TCP/IP connection to the WAP server, from sending the initial query to a server to the point of time when the content is demanded.

NOTE – This parameter can only be calculated in case of follow up page, if the TCP/IP connection is not persistent.

7.3.11.4.2 Abstract equation

$$\text{WAP(page) IP access set - up time}[s] = (t_{\text{WAP IP connection established}} - t_{\text{WAP IP connection request}})[s]$$

7.3.11.4.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{WAP IP connection request}}$: Time of WAP IP connection request.	Start: Selecting the link of a WAP page or applying an entered URL.	Start: Sending of the first TCP SYN.
$t_{\text{WAP IP connection established}}$: Time of WAP IP connection is established.	Not applicable.	Stop: Sending of the first HTTP GET command.

Remark:

- The bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5). Only successful measurements are taken into account to calculate the average time.

7.3.11.5 WAP (page) session failure ratio [%]

7.3.11.5.1 Abstract definition

The parameter WAP (page) session failure ratio is the proportion of unsuccessful WAP page access attempts and sessions that were started successfully.

7.3.11.5.2 Abstract equation

$$\text{WAP (page) session failure ratio}[\%] = \frac{\text{unsuccessful WAP page access attempts}}{\text{all WAP page access attempts}} \times 100$$

7.3.11.5.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
WAP page access attempt	Start: Selecting the link of a WAP page or applying an entered URL.	Start: WAP1.x: Sending of WSP Get Request WAP2.x: a) Sending of the first TCP SYN (if available); or b) Sending of HTTP Get Request (only if the first TCP SYN is not available).
Successful WAP page access attempt	Stop: The requested WAP page is completely loaded.	Stop: WAP1.x/WAP2.x: Reception of the last data packet containing the corresponding content.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Unsuccessful WAP page access attempt	Stop trigger point not reached.	
NOTE – In case of WAP 2.x the start trigger should be the first TCP SYN (a). If the TCP/IP connection is not re-established before the request of the new page (next page part), the start trigger has to be the first respective HTTP Get Request (b).		

Remark:

- The bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.11.6 WAP (page) session time [s]

7.3.11.6.1 Abstract definition

The parameter WAP (Page) session time provides the time in seconds between selection of a specific WAP page and the successful load of the page.

7.3.11.6.2 Abstract equation

$$\text{WAP (page) session time}\,[s] = \left(t_{\text{appearance WAPpage}} - t_{\text{selection WAPpage}}\right)[s]$$

7.3.11.6.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{selection WAP page}}$: Time of selection of the WAP page	Start: Selecting the link of a WAP page or applying an entered URL.	Start: WAP1.x: Sending of the first WSP Get Request. WAP2.x: a) Sending of the first TCP SYN (if available); or b) Sending of HTTP Get Request (only if first TCP SYN is not available).
$t_{\text{appearance WAP page}}$: Time of appearance of the WAP page	Stop: The requested WAP page is completely loaded.	Stop: WAP1.x/WAP2.x: Reception of the last data packet containing the corresponding content.
NOTE – In case of WAP 2.x the start trigger should be the first TCP SYN (a). If the TCP/IP connection is not re-established before the request of the new page (next page part), the start trigger has to be the first respective HTTP Get Request (b).		

Remark:

- The bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5). Only successful measurements are taken into account to calculate the average time.

7.3.11.7 WAP (page) request failure ratio [%]

7.3.11.7.1 Abstract definition

The WAP (page) request failure ratio is the probability that a WAP page request is not successful after a timeout period.

7.3.11.7.2 Abstract equation

$$\text{WAP (page) request failure ratio}[\%] = \frac{\text{unsuccessful WAP page request attempts}}{\text{all WAP page request attempts}} \times 100$$

7.3.11.7.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
WAP page request attempt	Start: Selecting the link of the WAP page.	Start: WAP1.x: Sending of WSP Get Request WAP2.x: Sending of HTTP Get Request.
Successful WAP page request attempt	Stop: Download begins.	Stop: WAP1.x/WAP2.x: Reception of the first data packet containing content.
Unsuccessful WAP page request attempt	Stop trigger point not reached.	

Remark:

- The bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.11.8 WAP (page) request time [s]

7.3.11.8.1 Abstract definition

The parameter WAP (page) request time describes the duration between selection of a specific WAP page and the reception of the first data packet containing WAP page content.

7.3.11.8.2 Abstract equation

$$\text{WAP }(\textit{page})\text{ request time}[s] = \left(t_{\text{first data packet reception}} - t_{\text{selection WAP page}}\right)[s]$$

7.3.11.8.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{selection WAP page}}$: Time of selection of the WAP site.	Start: Selecting the link of the WAP page.	Start: WAP1.x: Sending of the WSP Get Request WAP2.x: Sending of the HTTP Get Request.
$t_{\text{first data packet reception}}$: Time of first data packet reception.	Stop: Download begins.	Stop: WAP1.x/WAP2.x: Reception of the first data packet containing content.

Remark:

- The bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5). Only successful measurements are taken into account to calculate the average time.

7.3.11.9 WAP (page) mean data rate [kbit/s]

7.3.11.9.1 Abstract definition

The WAP (Page) mean data rate is the average data rate (WAP throughput) in kbit/s.

7.3.11.9.2 Abstract equation

$$\text{WAP (page) mean data rate}\,[\text{kbit/s}] = \frac{\text{WAP page size}\,[\text{kbyte}] \times 8}{\left(t_{\text{last data packet reception}} - t_{\text{first data packet reception}}\right)[\text{s}]}$$

7.3.11.9.3 Trigger points

The average throughput is measured from opening the data connection to the end of the successful transfer of the content (file, WAP page).

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{first data packet reception}}$: Time of first data packet reception	Start: Download begins.	Start: WAP1.x/WAP2.x: Reception of the first data packet containing the content.
$t_{\text{last data packet reception}}$: Time of last data packet reception	Stop: Download is completed.	Stop: WAP1.x/WAP2.x: Reception of the last data packet containing the corresponding content.

Remark:

- The bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.11.10 WAP (page) data transfer cut-off ratio [%]

7.3.11.10.1 Abstract definition

The WAP (page) data transfer cut-off ratio is the probability that a data download is incomplete after a timeout period (the download is aborted).

7.3.11.10.2 Abstract equation

$$\text{WAP (page) data transfer cut-off ratio}\,[\%] = \frac{\text{incomplete WAP page transfer attempts}}{\text{all WAP page transfer attempts}} \times 100$$

7.3.11.10.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
WAP page transfer attempt	Start: Download begins.	Start: WAP1.x/WAP2.x: Reception of the first data packet containing the content.
Successful WAP page transfer attempt	Stop: Download is completed.	Stop: WAP1.x/WAP2.x: Reception of the last data packet containing the corresponding content.
Incomplete WAP page transfer attempt	Stop trigger point not reached.	

Remark:

- The bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.3.11.11 WAP (page) data transfer time [s]

7.3.11.11.1 Abstract definition

The parameter WAP (page) data transfer time describes the duration between the reception of the first data packet and the last data packet containing WAP page content.

7.3.11.11.2 Abstract equation

$$\text{WAP (page) data transfer time}\,[s] = \left(t_{\text{last data packet reception}} - t_{\text{first data packet reception}}\right)[s]$$

7.3.11.11.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{first data packet reception}}$: Time of the first data packet reception	Start: Download begins.	Start: WAP1.x/WAP2.x: Reception of the first data packet containing the content.
$t_{\text{last data packet reception}}$: Time of the last data packet reception	Stop: Download is completed.	Stop: WAP1.x/WAP2.x: Reception of the last data packet containing the corresponding content.

Remark:

- The bearer has to be active in the cell used by a subscriber (see clause 7.2.1) and the mobile station has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5). Only successful measurements are taken into account to calculate the average time.

7.3.12 IMS multimedia telephony

This clause describes QoS parameters for the IMS multimedia telephony service (MTSI) as described in [ETSI TS 123 228].

The IMS multimedia service consists of several services, such as video, voice and text. The MTSI parameters are related to the control plane, to real time user services or non-real time user service as shown in Figure 7-37.

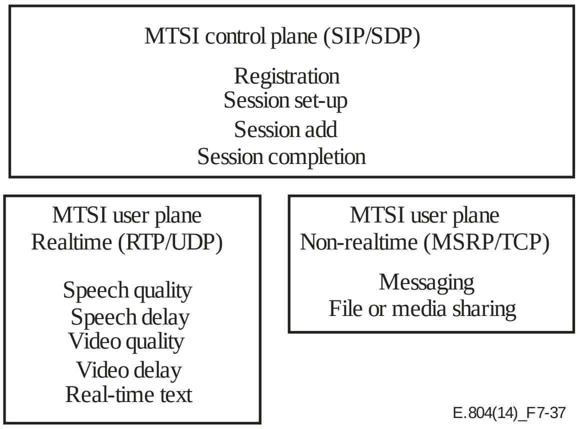


Figure 7-37 – MTSI parameter structure

7.3.12.1 MTSI registration failure ratio [%]

7.3.12.1.1 Abstract definition

The MTSI registration failure ratio is the probability that the terminal cannot register towards IP multimedia subsystem (IMS) when requested.

Remark:

- A successful MTSI registration is required before the terminal can use any MTSI services, and before other terminals can set up MTSI sessions towards it. Even if it is technically possible to wait with the registering until the first use of any MTSI service, it is normally expected that registration is done at terminal power-on.

Figure 7-38 shows message flow for successful MTSI registration example.

7.3.12.1.2 Abstract equation

$$\text{MTSI registration failure ratio}[\%] = \frac{\text{unsuccessful MTSI registration attempts}}{\text{all MTSI registration attempts}} \times 100$$

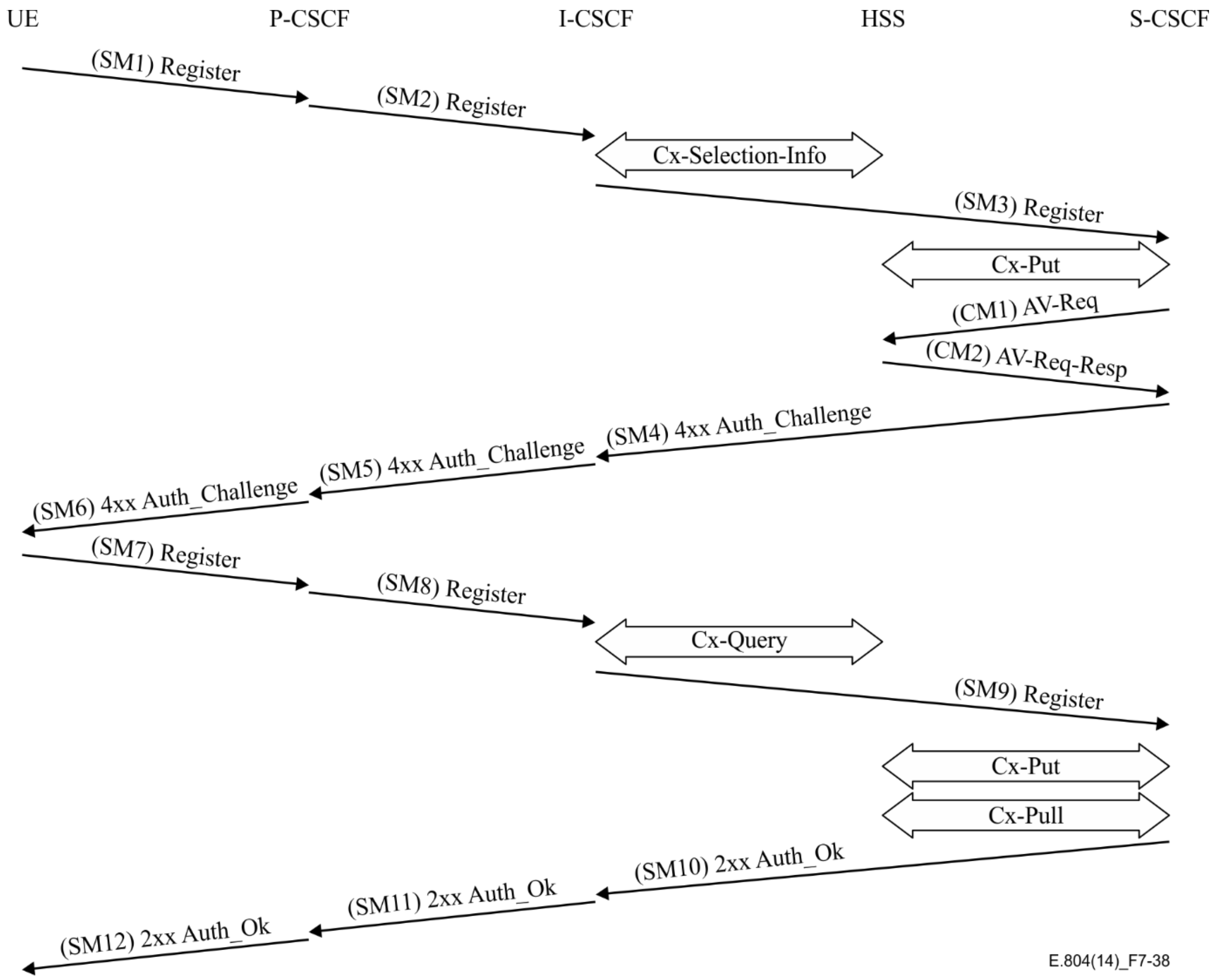


Figure 7-38 – Successful MTSI registration example

Remark:

- The first response to the REGISTER is normally a failure response, indicating that authentication must be done. The UE then makes a second REGISTER completed with the authentication information. After correct authentication, the UE then receives the 200 OK message.

7.3.12.1.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
MTSI registration attempt	Start: Power-on or activation of any MTSI service on the terminal.	Start: Protocol: SIP. First data packet is sent by the terminal containing a "SIP REGISTER" message.
Successful MTSI registration attempt	Stop: MTSI availability is indicated.	Stop: Protocol: SIP. First data packet is received containing a "SIP 200 OK" message.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Unsuccessful MTSI registration attempt	Stop: MTSI availability indication is not given within a pre-determined time.	Stop: Protocol: SIP. Case 1: Second data packet received by the terminal (after sending the "SIP REGISTER" message) containing a message different from "SIP 200 OK". Case 2: First data packet received by the terminal (after the authentication procedure) containing a message different from "SIP 200 OK". Case 3: No message received by the terminal within a pre-determined time.

7.3.12.2 MTSI registration time [s]

7.3.12.2.1 Abstract definition

The MTSI registration time is the time period between the IMS registration request and being registered to IMS.

7.3.12.2.2 Abstract equation

$$\text{MTSI registration time}[s] = (t_{MTSIAvailable} - t_{MTSIActivated})[s]$$

7.3.12.2.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{MTSIActivated}$: Time of MTSI registration attempt	Start: Power-on or activation of any MTSI service on the terminal.	Start: Protocol: SIP. First data packet is sent by the terminal containing a "SIP REGISTER" message.
$t_{MTSIAvailable}$: Time of successful MTSI registration attempt	Stop: MTSI availability is indicated.	Stop: Protocol: SIP. First data packet received containing a "SIP 200 OK" message.

7.3.12.3 MTSI session set-up failure ratio [%]

7.3.12.3.1 Abstract definition

The MTSI session set-up failure ratio is the probability that the terminal cannot set up an MTSI session. An MTSI session is initiated when the user presses the call button and receives a notification that the callee answers within a pre-determined time.

Remarks:

- In a normal SIP call, the user first receives a callee alerted notification; a series of "beep" tones that indicates that the terminating phone is ringing, until the callee answers the phone. However, for drive testing automatic answering will be used and in that case a session set-up notification is received directly instead of the callee alerted notification. The session set-up notification indicates that the other phone accepts the communication.
- An unsuccessful attempt may either be an attempt that is explicitly acknowledged by an error message from the terminating client/network or an attempt that does not results in any responses from the terminating terminal/network at all within a pre-determined time.

7.3.12.3.2 Abstract equation

$$\text{MTSI session set-up failure ratio}\,[\%] = \frac{\text{unsuccessful MTSI session setup attempts}}{\text{all MTSI session setup attempts}} \times 100$$

7.3.12.3.3 Trigger points

Event from abstract equation	Trigger points from user's point of view	Technical description/ protocol part
MTSI session set-up attempt	Start: User initiates session by pushing the call button to make the call.	Start: Protocol: SIP. The trigger from the IMS client that forces the SIP layer of the terminal to create a "SIP INVITE" and send it to the transport layers of the terminal.
Successful MTSI session set-up attempt	Stop: The user hears or sees an indication that the other phone accepts the invitation	Stop: Protocol: SIP. The terminal has received a data packet containing the final "SIP 200 OK (INVITE)" message.
Unsuccessful MTSI session set-up attempt	Stop: The user receives a notification that the session set-up is cancelled, or do not receive any notification at all within a pre-determined time.	Stop: Protocol: SIP. Example of unsuccessful case 1: The terminal informs the IMS client that the SIP session set-up is cancelled after the terminal receives an error, cancel, or redirection message (e.g., a "403 Forbidden" or "488 Not Acceptable Here" message as response to the "SIP INVITE"). Example of unsuccessful case 2: The terminal does not receive any messages to react on within a pre-determined time.

7.3.12.4 MTSI session set-up time [s]

7.3.12.4.1 Abstract definition

The MTSI session set-up time is the time period between initiation of an MTSI session by, e.g., pressing the call button and the reception of a notification that the session has been set up.

Figure 7-39 shows message flow for an implicit initiation of the MTSI session.

7.3.12.4.2 Abstract equation

$$\text{MTSI session set-up time}[s] = \left(t_{\text{user receives notification}} - t_{\text{user initiates session}}\right)[s]$$

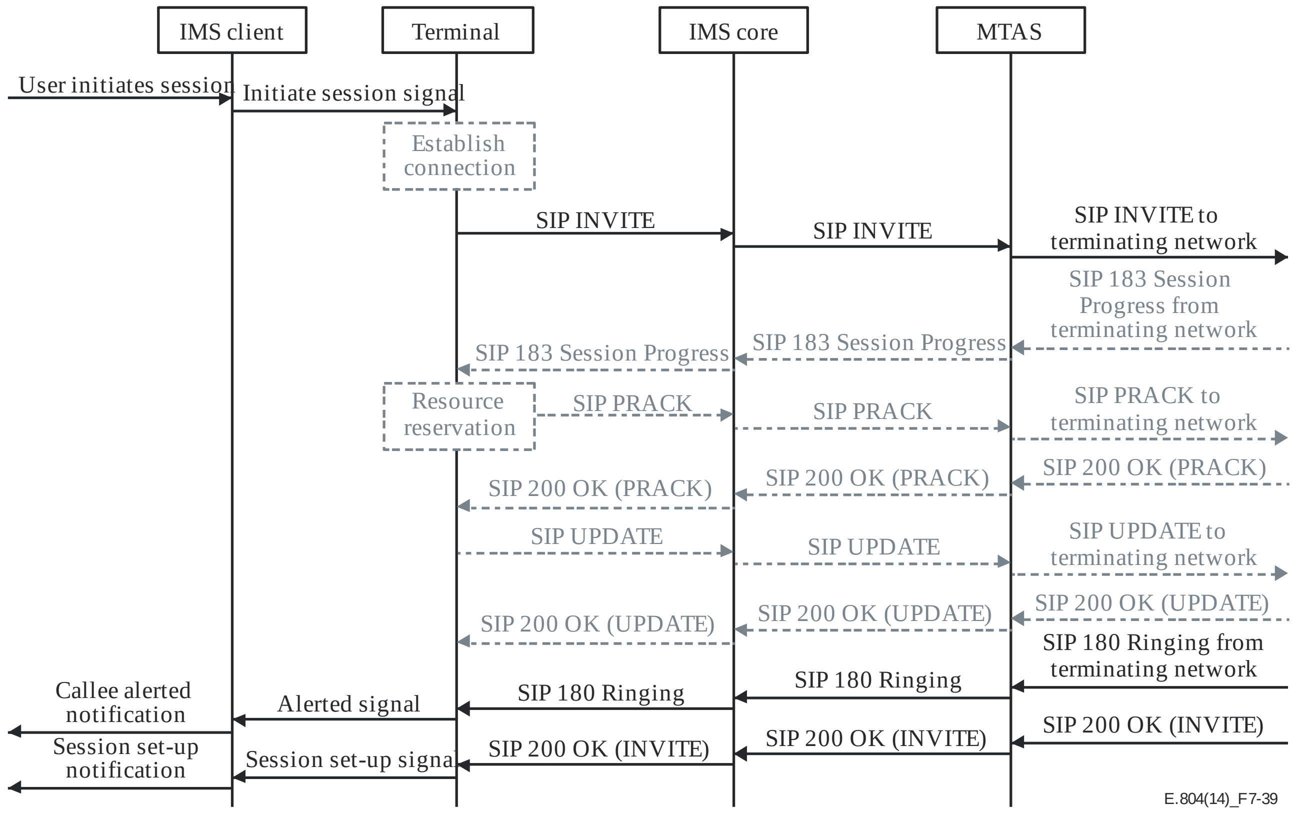


Figure 7-39 – Implicit initiation of the MTSI session

Remarks:

- In a normal SIP call, the user first receives a callee alerted notification; a series of "beep" tones that indicates that the terminating phone is ringing, until the callee answers the phone. However, for drive testing automatic answering will be used and in that case a session set-up notification is received directly instead of the callee alerted notification. The session set-up notification indicates that the other phone accepts the communication.
- In most normal use-cases the originating and terminating mobile terminals are in battery saving mode and do not have any radio bearers established prior the MTSI session set-up. In these cases, the mobile terminal must establish connection to radio access network (RAN) by establishing a radio bearer. The delay contribution to the total MTSI session set-up of this procedure cannot be regarded as insignificant. This is shown in Figure 7-39 by a dashed box labelled "Establish connection".
- All or a subset of the dashed arrows in Figure 7-39 occur in case that the mobile terminals involved in the call need to reserve media resources in the RAN prior to starting the communication. Hence, the session set-up time depends on the resources needed for the media and if any resources was already reserved by the mobile terminals prior to the session set-up.

7.3.12.4.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{user\,initiates\,session}$	Start: User initiates session by pushing the call button.	Start: Protocol: SIP. First data packet is sent by the terminal containing a "SIP INVITE" message.
$t_{user\,receives\,notification}$	Stop: The user receives a notification that the other phone accepts the invitation.	Stop: Protocol: SIP. First data packet is received by terminal containing SIP 200 OK (INVITE).

7.3.12.5 MTSI session add failure ratio [%]

7.3.12.5.1 Abstract definition

The MTSI session add failure ratio is the probability that the terminal cannot add a media component. The change is initiated when the user starts to modify an existing MTSI session by adding a media component. The user then receives a notification that the callee is alerted about the session change within a pre-determined time. Alternatively, the terminating phone can have automatic consent to session changes configured.

Remark:

- The failure ratio can be dependent on the type of the added media component.

7.3.12.5.2 Abstract equation

$$\text{MTSI session add failure ratio}\,[\%] = \frac{\text{unsuccessful MTSI session add attempts}}{\text{all MTSI session add attempts}} \times 100$$

7.3.12.5.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
MTSI session add attempt	Start: User modifies session by pushing appropriate button to add a media component to/in the session.	Start: Protocol SIP. The trigger from the IMS client that forces the SIP layer of the terminal to create a "SIP INVITE" and send it to the transport layers of the terminal.
Successful MTSI session add attempt	Stop: Getting notification that the session change is accepted and, e.g., the new media stream starts (when using automatic consent) or appropriate notification that the other terminal accepts or rejects the session change.	Stop: Protocol SIP. The terminal has received a data packet containing the "SIP 180 Ringing" message or a "SIP 200 OK" message and informs the IMS client that performs a callee alerted notification or a session changed notification.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Unsuccessful MTSI session add attempt	Stop: The user receives a notification that the session change is cancelled, or does not receive any notification at all within a pre-determined time.	Stop: Protocol SIP. Example of unsuccessful case 1: The terminal informs the IMS client that the SIP session change is cancelled after the terminal receives an error, cancel, or redirection message (e.g., a "403 Forbidden" or "488 Not Acceptable Here" message as response to the "SIP INVITE"). Example of unsuccessful case 2: The terminal does not receive any messages to react on within a pre-determined time.

7.3.12.6 MTSI session add time [s]

7.3.12.6.1 Abstract definition

The MTSI session add time is the time period from the start if changing a session (adding a media component) to the reception of a notification that the session has been changed.

Remark:

- The terminals involved must have an MTSI session ongoing before it can be modified.

Figure 7-40 shows message flow for modification of the MTSI session.

7.3.12.6.2 Abstract equation

$$\text{MTSI session add time}\,[s] = t_{\text{User Receives Change Notification}} - t_{\text{User Modifies Session}}$$

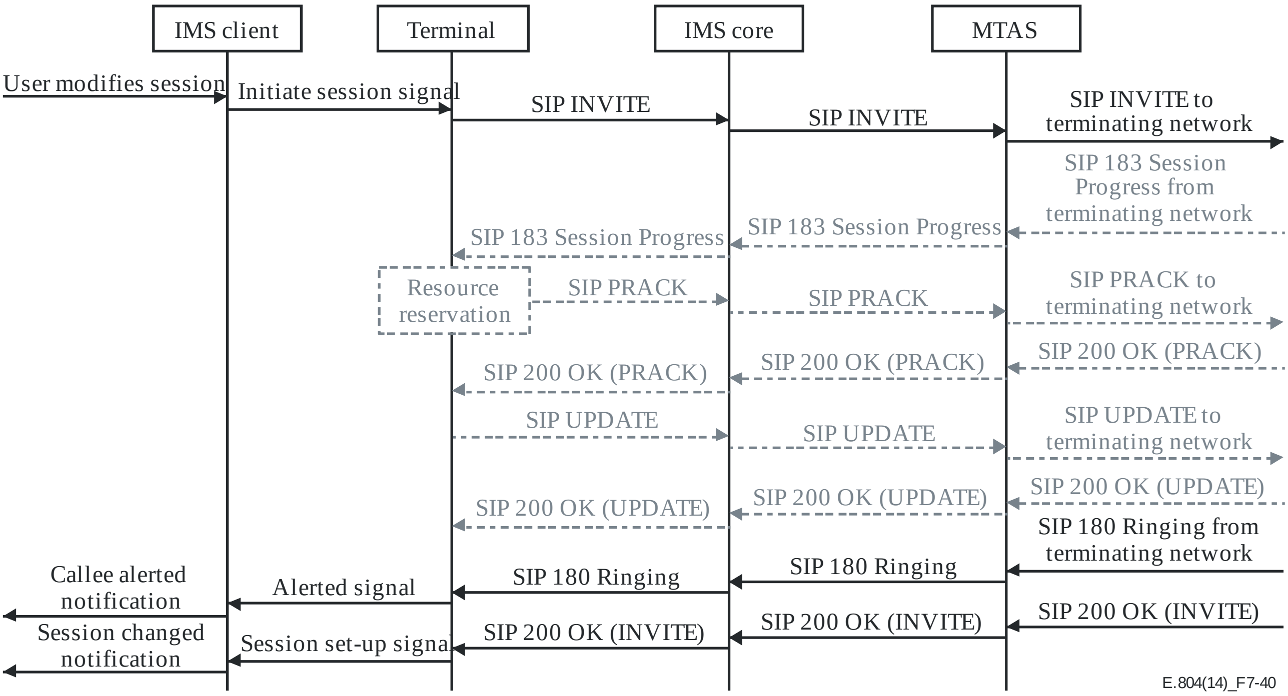


Figure 7-40 – Modification of the MTSI session

Remarks:

- The MTSI session change signalling to add a media component follows the same set of rules as the MTSI session set-up signalling. Therefore, the signalling diagrams in Figures 7-39 and 7-40 are almost identical. The main difference is that the terminals will already have one or more radio bearers established at session change and the radio connection does not need to be established as for the initial session set-up.
- In the case of automatic consent to session changes, the terminating UE may not send any "SIP 180 Ringing" message. In that case the final session change notification (triggered by the "SIP 200 OK (INVITE)" message should be used as the final trigger point for session change latency measurements.
- The dashed arrows and box in Figure 7-40 are optional signals and event that may occur in the case that one or two mobile terminals are involved in the session change.
- All or a sub-set of the dashed arrows in Figure 7-40 occur in case that the terminals involved in the call need to reserve resources in the radio access network (RAN) when adding a new media stream to the MTSI session. Hence, the set-up time depends on the resources needed for the new media stream and the resources reserved by the mobile terminals prior to the session change.

7.3.12.6.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t(user modifies session)	Start: User modifies session by pushing appropriate button to add a media component to/in the session.	Start: Protocol: SIP. First data packet is sent by the terminal containing a "SIP INVITE" message.
t(user receives change notification)	Stop: Getting notification that the session change is accepted.	Stop: Protocol: SIP. First data packet received by terminal containing SIP 200 OK (INVITE).

7.3.12.7 MTSI session remove failure ratio [%]

7.3.12.7.1 Abstract definition

The MTSI session remove failure ratio is the probability that the terminal cannot remove a media component. The removal is initiated when the user starts to modify an existing MTSI session by removing a media component. The user then receives a notification that the callee is alerted about the session change within a pre-determined time. Alternatively, the terminating phone can have automatic consent to session changes configured.

7.3.12.7.2 Abstract equation

$$\text{MTSI session remove failure ratio}[\%] = \frac{\text{unsuccessful MTSI session removal attempts}}{\text{all MTSI session removal attempts}} \times 100$$

7.3.12.7.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
MTSI session removal attempt	Start: User modifies the session by pushing appropriate button to remove a media component to/in the session.	Start: Protocol SIP. The trigger from the IMS client that forces the SIP layer of the terminal to create a "SIP INVITE" and send it to the transport layers of the terminal.
Successful MTSI session remove attempt	Stop: Getting notification that the session change is performed.	Stop: Protocol SIP. The terminal has received a data packet containing the "SIP 180 Ringing" message or a "SIP 200 OK" message and informs the IMS client to perform a session changed notification.
Unsuccessful MTSI session removal attempt	Stop: The user receives a notification that the session change is cancelled, or do not receive any notification at all within a pre-determined time.	Stop: Protocol SIP. Example of unsuccessful case 1: The terminal informs the IMS client that the SIP session change is cancelled after the terminal receives an error message as a response to the "SIP INVITE"). Example of unsuccessful case 2: The terminal does not receive any messages to react on within a pre-determined time

7.3.12.8 MTSI session remove time [s]

7.3.12.8.1 Abstract definition

The MTSI session remove time is the time period from the start of changing a session (removing a media component) to the reception of a notification that the session has been changed.

Remark:

- The terminals involved must have an MTSI session ongoing before it can be modified.

7.3.12.8.2 Abstract equation

$$\text{MTSI session remove time}[s] = t_{\text{UserReceivesChangeNotification}} - t_{\text{UserModifiesSession}}$$

7.3.12.8.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t(user modifies session)	Start: User modifies session by pushing appropriate button to remove a media component from the session.	Start: Protocol: SIP. First data packet is sent by the terminal containing a "SIP INVITE" message.
t(user receives change notification)	Stop: Getting notification that the session change is accepted.	Stop: Protocol: SIP. First data packet received by the terminal containing SIP 200 OK (INVITE).

7.3.12.9 MTSI session completion failure ratio [%]

7.3.12.9.1 Abstract definition

The MTSI session completion failure ratio is the probability that a successfully started MTSI call is ended by a cause other than intentional termination by A- or B-party.

Figure 7-41 shows signalling flow to end an MTSI session.

7.3.12.9.2 Abstract equation

$$\text{MTSI session completion failure ratio}\,[\%] = \frac{\text{unsuccessfully completed MTSI sessions}}{\text{all successfully started MTSI sessions}} \times 100$$

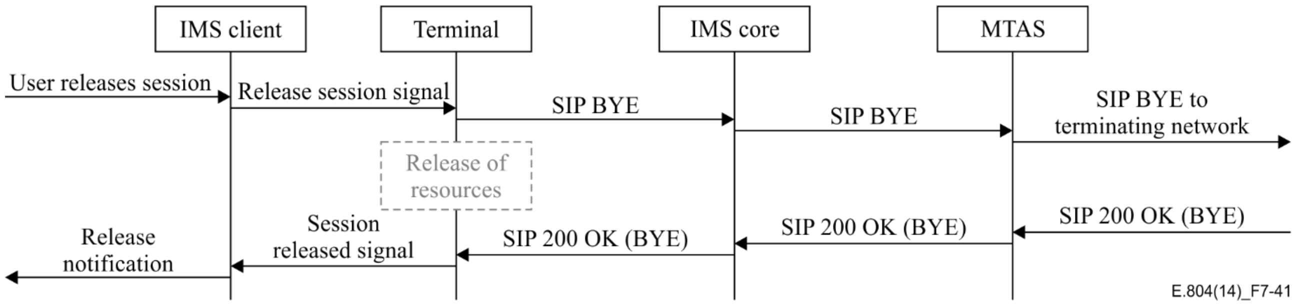


Figure 7-41 – Signalling flow to end an MTSI session

Remark:

- The dashed box is an optional event that typically occurs in the case when a mobile terminal is used. The event is the release of resources that has been reserved in the radio access network (RAN).

7.3.12.9.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started MTSI sessions	Start: User initiates the session by pushing the call button to make the call.	Start: Protocol SIP. The terminal has sent INVITE and received a "SIP 200 OK (INVITE)" message.
Successfully completed MTSI sessions	Stop: The user is notified that the call has ended and that the phone is ready to initiate and receive other calls.	Stop: Protocol SIP. The terminal has received a data packet containing the "SIP 200 OK" message as a response to a "SIP BYE" request and informs the IMS client that performs a release notification.
Unsuccessfully completed MTSI sessions	Stop: Beside the successful release cases described above, some session may be released unexpectedly (i.e., the call is dropped).	Stop: Protocol SIP. Example of unsuccessful case: The terminal loses connectivity and no signalling and/or media can be sent or received.

7.3.12.10 MTSI speech quality

7.3.12.10.1 Abstract definition

The MTSI speech quality represents the end-to-end speech quality of the service.

Remarks:

- The speech quality can be measured for both the caller and the callee.
- The acoustical behaviour of the terminal is not part of this speech quality measurement.
- The speech quality can be measured with a full reference model taking the original speech sample and the degraded sample as input, or with a parametric model taking transport and terminal parameters as input.

7.3.12.10.2 Abstract equation

The validation of the end-to-end quality is made using MOS-LQO scales. These scales describe the opinion of users with speech transmission and its troubles (noise, robot voice, echo, dropouts, time scaling introduced by the jitter buffer, etc.) according to [ITU-T P.863]. The scale used has to be reported. An aggregation of measurement activities or parts of it should be made on speech sample basis.

7.3.12.10.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Not applicable.	Start: Interchange speech samples between A-party and B-party.	Start: Reception of the first RTP packet containing a speech frame
Not applicable.	Stop: Session completion or session change, where the speech service is removed from the session.	Stop case 1: The terminal has received a data packet containing the "SIP 200 OK" message as a response to a "SIP BYE" request and informs the IMS client that perform a release notification. Stop case 2: The terminal has received a data packet containing the "SIP 200 OK" message and informs the IMS client that the speech service is no longer active.

7.3.12.11 MTSI speech transmission delay [s]

7.3.12.11.1 Abstract definition

The MTSI speech transmission delay is the delay between sending speech packets from terminal A to receiving speech packets at terminal B, when the speech is conveyed in the context of an MTSI call.

Figure 7-42 shows the contributing parts to the MTSI speech transmission delay.

7.3.12.11.2 Abstract equation

$$\text{MTSI speech transmission delay}[\text{s}] = t(B\_receives) - t(A\_sends)[\text{s}]$$

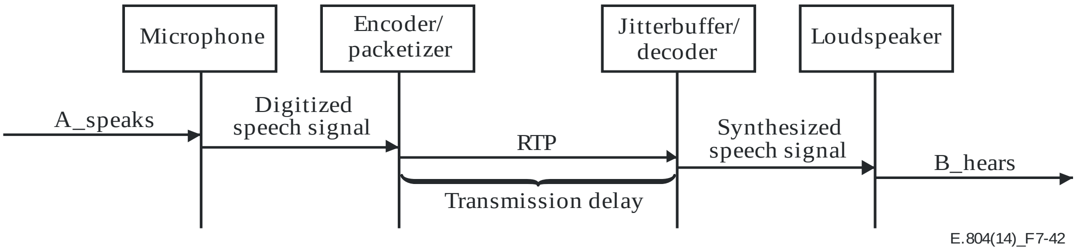


Figure 7-42 – The MTSI speech transmission delay

NOTE 1 – Since the delay can vary for each packet, it is not statistically sufficient to measure the delay only for the first packet.

NOTE 2 – The Speech transmission delay is not exactly the same as perceived by the end user. The speech transmission delay does not include the delay introduce by the jitter buffer and the encoding and decoding delay.

7.3.12.11.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t(A_sends)	Start: Terminal A sends speech.	Start: Protocol: RTP. Data packet sent by terminal A containing speech data.
t(B_receives)	Stop: Speech received by terminal B.	Stop: Protocol: RTP. Corresponding data packet received by terminal B containing speech data.

7.3.12.12 MTSI speech path delay [s]

7.3.12.12.1 Abstract definition

The MTSI speech path delay is the speech delay between reception of speech by the microphone in terminal A to the loudspeaker playing out the speech at terminal B, when the speech is conveyed in the context of an MTSI call.

Figure 7-43 shows the contributing parts to the MTSI speech path delay.

7.3.12.12.2 Abstract equation

$$\text{MTSI Speech Path Delay}\,[\mathrm{s}] = t(B\_hears) - t(A\_speaks)\,[\mathrm{s}]$$

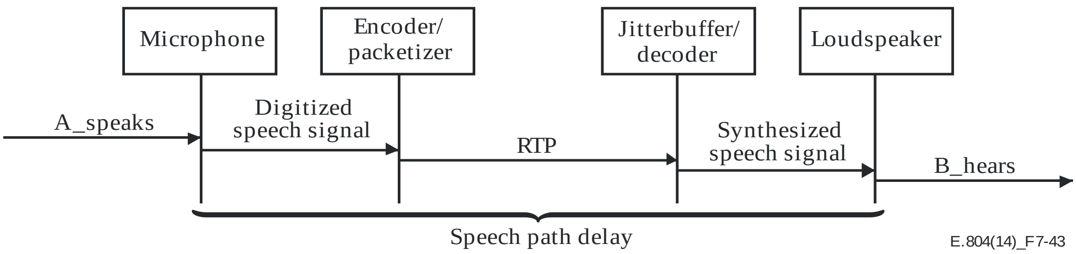


Figure 7-43 – The MTSI speech path delay

NOTE – Since the delay can vary during a call, it is not statistically sufficient to measure the delay only once.

7.3.12.12.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t(A_speaks)	Start: A-party speaks into the microphone	Start: Electrical signal at the microphone. The speech is received at the microphone (acoustical delay not included)
t(B_hears)	Stop: The speaker at the B-party plays the speech	Stop: Electrical signal at the speaker The corresponding speech is played out by the speaker (acoustical delay not included)

7.3.12.13 MTSI video quality

7.3.12.13.1 Abstract definition

The MTSI video quality represents the end-to-end video quality of the service.

Remarks:

- The video quality can be measured for both the caller and the callee.
- The visual behaviour of the terminal's display is not part of this video quality measurement.
- The video quality can be measured with a full reference model taking the original video sample and the degraded sample as input, or with a parametric model taking transport and terminal parameters as input.

7.3.12.13.2 Abstract equation

The validation of the end-to-end quality is made using the MOS scale. This scale describes the opinion of users using the video service with its degradations (blockiness, jerkiness, freezes, etc.). An aggregation of measurement activities or parts of it should be made on video sample basis.

Remark:

- Objective video quality models are to be defined.

7.3.12.13.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Not applicable.	Start: Interchange video between the A-party and the B-party.	Start: Reception of the first RTP packet containing a video frame.
Not applicable.	Stop: Session completion or session change, where the video service is removed from the session.	Stop case 1: The terminal has received a data packet containing the "SIP 200 OK" message as a response to a "SIP BYE" request and informs the IMS client that performs a release notification. Stop case 2: The terminal has received a data packet containing the "SIP 200 OK" message and informs the IMS client that the video service is no longer active.

7.3.12.14 MTSI video transmission delay [s]

7.3.12.14.1 Abstract definition

The MTSI video transmission delay is the delay between sending the video packets from terminal A, and reception of the video packets at terminal B, where the video is transmitted in the context of an MTSI video call.

Figure 7-44 shows the contributing parts to the MTSI video transmission delay.

7.3.12.14.2 Abstract equation

$$\text{MTSI video transmission delay}[\text{s}] = t(B\_receives) - t(A\_sends)[\text{s}]$$

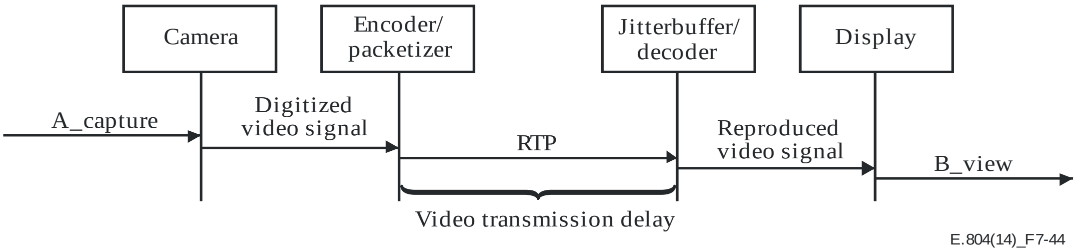


Figure 7-44 – The MTSI video transmission delay

NOTE 1 – Since the delay can vary for each packet, it is not statistically enough to measure only the delay for the first packet.

NOTE 2 – The video transmission delay is not exactly the same as perceived by the end user. The video transmission delay does not include the delay introduce by the jitter buffer and the encoding and decoding delays.

7.3.12.14.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t(A_sends)	Start: Terminal A sends the video.	Start: Protocol: RTP. Data packet is sent by terminal A containing the video data.
t(B_receives)	Stop: Video received at terminal B.	Stop: Protocol: RTP. Corresponding data packet received by terminal B containing the video data.

7.3.12.15 MTSI video path delay [s]

7.3.12.15.1 Abstract definition

The MTSI video path delay is the delay between capturing of video at terminal A and display of the video at terminal B, where the video is transmitted in the context of an MTSI video call.

Figure 7-45 shows the contributing parts to the MTSI video path delay.

7.3.12.15.2 Abstract equation

$$\text{MTSI video path delay}\,[\text{s}] = t(B\_displays) - t(A\_captures)\,[\text{s}]$$

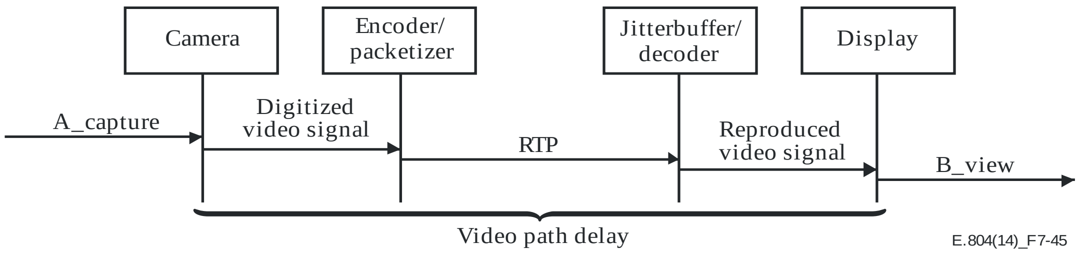


Figure 7-45 – The MTSI video path Delay

NOTE – Since the delay can vary during the session, it is not statistically sufficient to measure the delay only once.

7.3.12.15.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t(A_captures)	Start: Terminal A captures the video	Start: Terminal A captures a video frame
t(B_displays)	Stop: Terminal B displays the video	Stop: Terminal B displays the corresponding video frame

7.3.12.16 MTSI audio/video de-synchronization [%]

7.3.12.16.1 Abstract definition

The MTSI audio/video de-synchronization is the percentage of time that the time differences of the audio and video signal (the "lip sync") at the receiving side is outside two thresholds, in the context of an MTSI combined audio/video call.

The de-synchronization impacts the perceived quality of the service. For broadcasting purposes, [ITU-R BT.1359-1] defines detectability and acceptability thresholds for lip synchronization. Figure 7-46 describes these thresholds. Note that the curve is not symmetrical around zero, as it is more annoying if the speech is played out too early than too late.

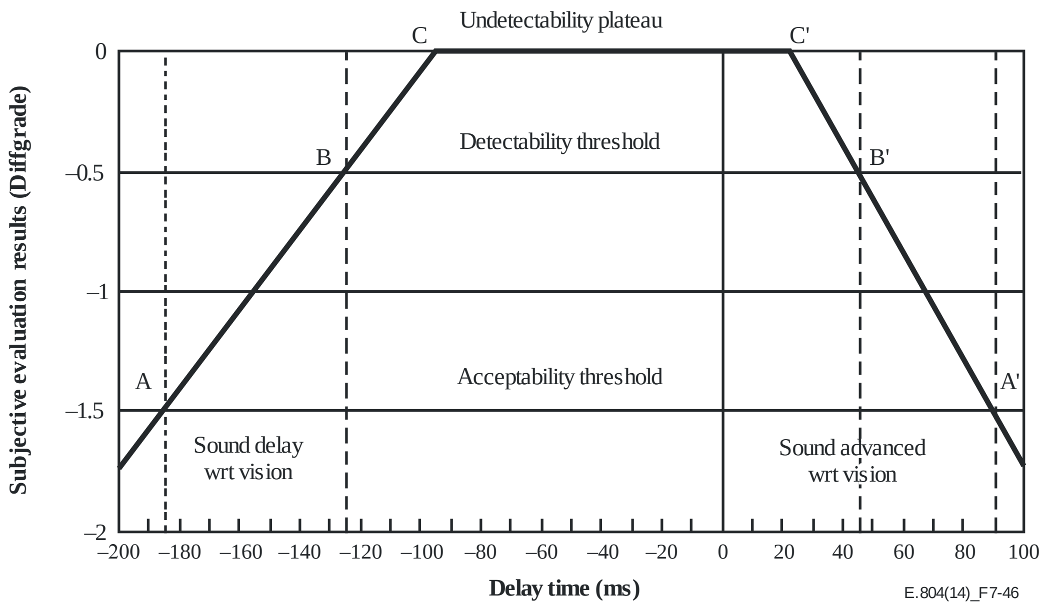


Figure 7-46 – The impact of audio video de-synchronization on perceived quality

7.3.12.16.2 Abstract equation

NOTE – The equation below only calculates the lip sync at a certain position in the video transmission. The measurement frequency to get useful measurement results is still to be defined.

$$\text{MTSI audio video de-synchronization} = \text{video path delay versus speech path delay } [s] = t(B\_view) - t(B\_hear)$$

7.3.12.16.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t(B_hear)	The loudspeaker at B-party plays the speech	Electrical signal at the speaker playing a speech frame The speech is played out by the speaker (acoustical delay not included)
t(B_view)	The display at B-party displays the video corresponding to the speech	The rendering of the video frame corresponding to the speech frame

7.3.12.17 MTSI real time text failure ratio [%]

7.3.12.17.1 Abstract definition

The MTSI real time text failure ratio is the proportion of not displayed letters and total number of letters sent in a successfully started MTSI real time text session.

Remark:

- Real time text is a real time communication method and it is important that the end-to-end delay is low. Therefore, when measuring the success ratio, letters that are received with a delay longer than a pre-determined time should be regarded as lost.

7.3.12.17.2 Abstract equation

$$\text{MTSI real-time text failure ratio} = \frac{\text{Number of not displayed letters in real time text session}}{\text{Number of typed letters in real time text session}} \times 100$$

7.3.12.17.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Started MTSI real time text session	Start: User A initiates/modifies an MTSI session with user B so it includes real time text. This is indicated to the users, and they start to communicate using text.	Start: The first typed real time text is captured and is sent to the transport layers of the terminal. The real time text protocol stack may use redundancy (i.e., the letters are sent multiple times) to make the communication more robust to loss of data packets.
Completed MTSI real time text session	Stop: One of the users pushes the end/modify call button to end the MTSI real time text communication. The session ends or is modified and this is indicated to the users.	Stop: The last part of the real time text conversation is captured and sent by the terminal. Followed by the release or modification (drop of the real time text media) of the SIP session.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Number of not displayed letters	During the real time text communication, some letters may be lost or delayed which leads to impairments of the text communication.	Example of unsuccessful case 1: data packets containing the text are lost or received too late and even if redundancy was applied, parts of the typed text string are lost and cannot be displayed correctly or displayed in time. Example of unsuccessful case 2: The transport of real time text data stops unexpectedly.

7.3.12.18 MTSI real time text delivery time [s]

7.3.12.18.1 Abstract definition

The MTSI real time text delivery time is the delay between sending a character from terminal A and reception of the same character in terminal B.

Remarks:

- The recommendation is to buffer text input 300 ms before sending the typed characters, and the maximum allowed buffering time is 500 ms. This means that normally only one or a few characters are typically transmitted to the other end in each RTP packet.
- The default redundancy scheme is to send the last two text packets together with the most recent text packet. In this way up to two consecutive RTP packets can be lost without losing any characters. However, other redundancy schemes can be used, and it is up to the terminal vendor to select an appropriate scheme depending on the current channel conditions.

7.3.12.18.2 Abstract equation

NOTE – Since the delay can vary for each packet, it is not statistically enough to measure only the delay for the first packet.

$$\text{MTSI real-time text delivery time} = t_{B\_receive} - t_{A\_send}$$

7.3.12.18.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t (A_send)	Start: User A writes a character.	Start: Protocol: RTP. Data packet is sent by terminal A containing the typed character.
t(B_receive)	Stop: User B receives the character on its screen.	Stop: Protocol: RTP. Corresponding data packet is received by terminal B containing the same character.

7.3.12.19 MTSI messaging failure ratio [%]

7.3.12.19.1 Abstract definition

The MTSI messaging failure ratio is the proportion of not received messages and sent messages in an MTSI messaging session.

Figure 7-47 shows signal flow for messaging using MSRP.

7.3.12.19.2 Abstract equation

$$\text{MTSI messaging failure ratio} = \frac{\text{Number of not received messages}}{\text{Total number of sent messages}} \times 100$$

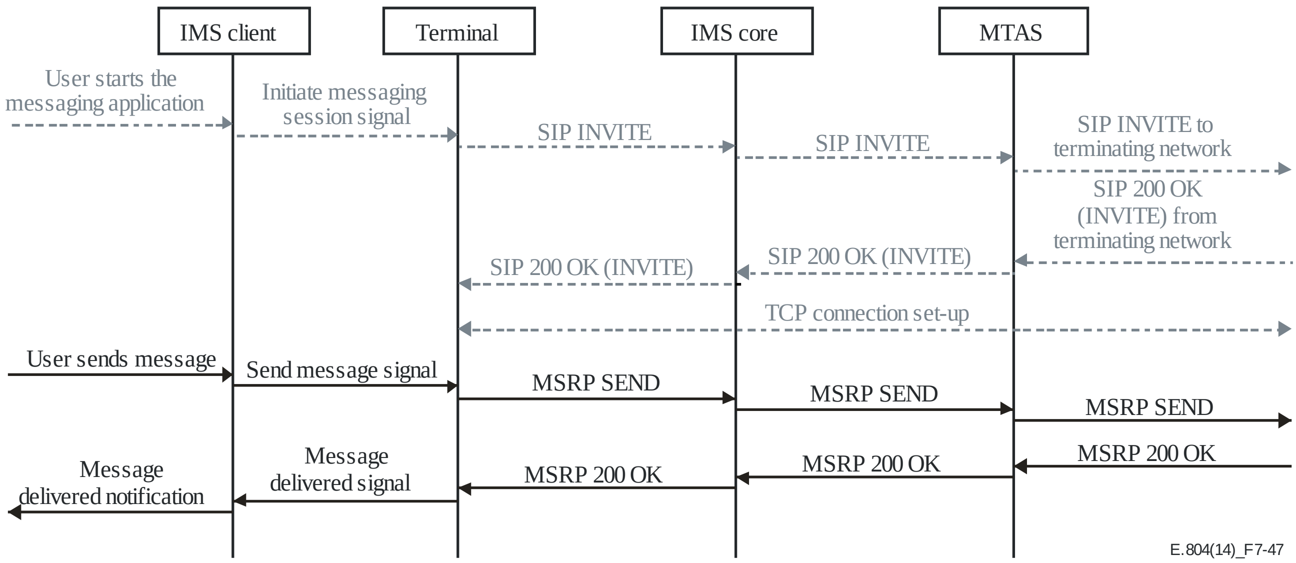


Figure 7-47 – Messaging using MSRP

Remark:

- Before a message can be sent, an MTSI session must either be established or modified so it contains messaging. Furthermore, a TCP connection for message session relay protocol (MSRP) transfer must be established between the two terminals. Typically, the MTSI session and the TCP connection is established or modified when an end user opens up the messaging application on its phone, e.g., during a call. The message is sent using MSRP in a later stage that happens when the user has typed the message using the messaging application and has pressed the "send button".

7.3.12.19.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Start of MTSI messaging session	Start: User A initiates/modifies an MTSI session with user B so it includes messaging. This is indicated to the users and they send messages.	Start: The trigger from the IMS client that starts the messaging session set-up that is followed by a number of message transmissions.
Completed MTSI messaging session	Stop: One of the users pushes the end/modify call button to end the MTSI messaging exchange. The session ends or is modified and this is indicated to the users	Stop: The messaging communication ends and is followed by the release or modification (drop of the messaging service) of the SIP session.
Received messages	Messages are delivered to user B.	Successful case: The terminal receives the "MSRP 200 OK", on time and acknowledges the reception of the message.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
		This is indicated to the IMS client which notifies the user.
Not received messages	Messages either are not delivered to user B, or they are not delivered within a pre-determined time	Example of unsuccessful case 1: The terminal receives an error message (i.e., a "MSRP 4xx or MSRP 5xx message), which is indicated to the IMS client. Example of unsuccessful case 2: The connectivity is lost by one or both of the terminals and no MSRP messages is sent/received by the terminal within a pre-determined time

7.3.12.20 MTSI messaging delivery time [s]

7.3.12.20.1 Abstract definition

The MTSI messaging delivery time is the time difference between sending a message from terminal A and reception of the same message in terminal B, where the terminals are involved in an MTSI messaging communication.

7.3.12.20.2 Abstract equation

$$\text{MTSI messaging delivery time}[s] = t_{\text{Message\_received}} - t_{\text{Message\_sent}}$$

7.3.12.20.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
t(Message_sent)	Start: User A sends a message.	Start: Protocol: MSRP The message is sent using MSRP SEND.
t(Message_received)	Stop: User B receives the message.	Stop: Protocol: MSRP The corresponding MSRP SEND message is received at terminal B.

NOTE – An alternative method is to measure the time between MSRP SEND and MSRP 200 OK, which then can be measured in the same terminal. However, the reception of MSRP 200 OK is not necessarily shown to the end user (depending on terminal implementation).

7.3.12.21 MTSI file/media sharing failure ratio [%]

7.3.12.21.1 Abstract definition

The MTSI file/media sharing failure ratio is the proportion of uncompleted file/media sharing sessions and sessions that were started successfully.

Remark:

- The files can either be a generic file, or a file with a predetermined file and media format.

7.3.12.21.2 Abstract equation

$$\text{MTSI file/media sharing failure ratio} = \frac{\text{uncompleted file/media sharing sessions}}{\text{successfully started file/media sharing sessions}} \times 100$$

7.3.12.21.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started file/media sharing session	Start: User A initiates/modifies a MTSI session with user B so it includes file/media sharing. This is indicated to the users, and they send files.	Start: The trigger from the IMS client that starts the file/media sharing session set-up that is followed by the file transmission.
Completed file/media sharing sessions	Stop: One of the users pushes the end/modify call button to end the MTSI file/media sharing. The session ends or is modified and this is indicated to the users	Stop: The file/media sharing ends and is followed by the release or modification (removal of the file/media sharing) of the SIP session.
Total number of sent files	Start: User A initiates/modifies a MTSI session with user B so it includes file/media sharing. This is indicated to the users, and they send files. Stop: One of the users pushes the end/modify call button to end the MTSI file/media sharing. The session ends or is modified and this is indicated to the users	Start: The trigger from the IMS client that starts the file/media sharing session set-up that is followed by the file transmission. Stop: The file/media sharing ends and is followed by the release or modification (removal of the file/media sharing) of the SIP session.
Uncompleted file/media sharing sessions	Files either are not delivered to user B, or they are not delivered within a pre-determined time.	The terminal does not receive the "MSRP 200 OK" (that acknowledges the reception of a file) within a pre-defined time. Example 1: The terminal receives an error message (i.e., a "MSRP 4xx or MSRP 5xx message) Example 2: The connectivity is lost by one or both of the terminals and no MSRP messages is sent/received by the terminal within a pre-determined time

7.3.12.22 MTSI file/media sharing mean data rate [kbit/s]

7.3.12.22.1 Abstract definition

The MTSI file/media sharing mean data rate is the average data transfer rate measured from a successful transfer of a file or pre-determined media type.

Figure 7-48 shows signal flow for file/media sharing using MTSI.

7.3.12.22.2 Abstract equation

$$\text{MTSI file/media sharing mean data rate}[\text{kbps}] = \frac{\text{Amount of user data transferred}[\text{kb}]}{\text{t(ContentSent) - t(ConnectionEstablished)}}$$

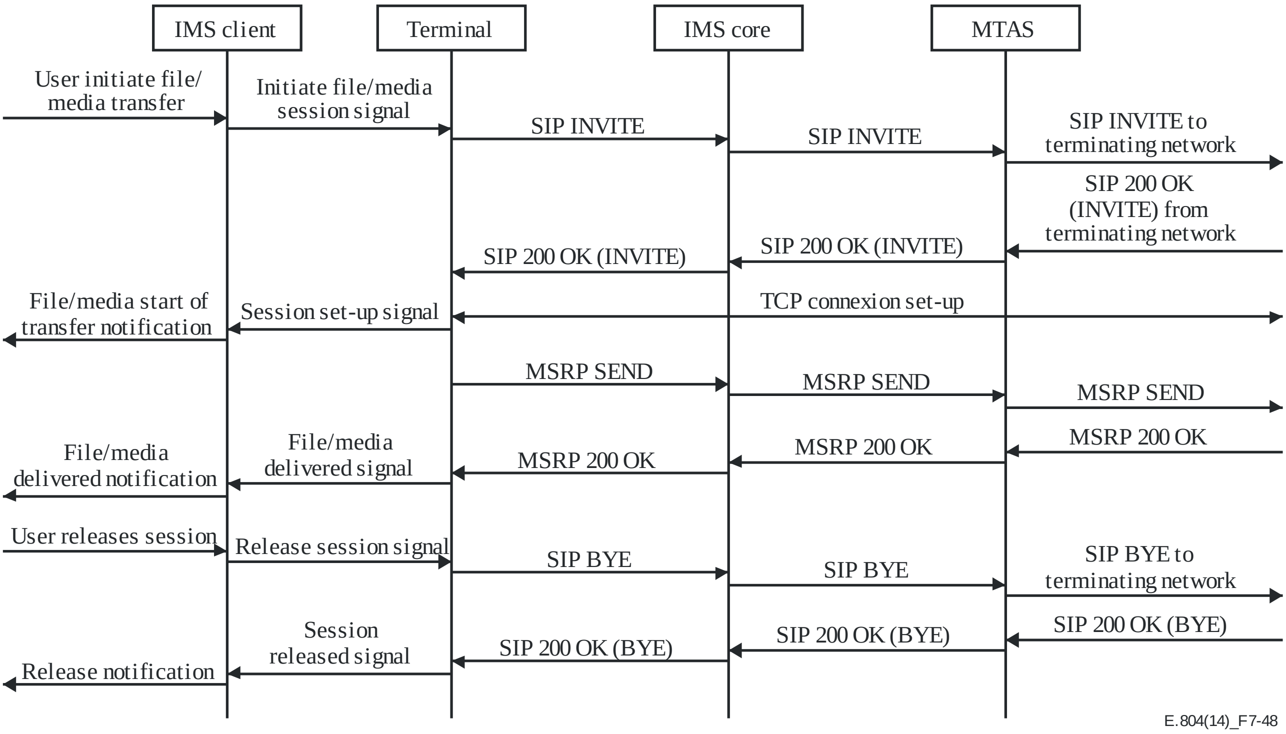


Figure 7-48 – File/media sharing using MTSI

Remarks:

- MTSI file/media sharing uses the same user plane protocol suite as MTSI messaging. Thus, the two methods of communication follow the same set of rules but with one exception. The exception is that for file/media sharing only one MSRP transaction is allowed per established or modified SIP session. Hence, after the file is successfully transferred the MTSI session is either terminated or modified to not contain file/media sharing.
- The time it takes the user to initiate the file/media transfer until receiving the file/media delivered notification can be divided into two parts. The first part is the access time, which is marked in Figure 7-48 as the time between "User initiate file/media transfer" until "File/media start transfer notification". The second part is the transfer time that is the time between the "File/media start transfer notification" and the "File/media delivered notification". This KPI aims to measure the average data rate during the transfer time.
- In file/media sharing the content is usually several maximum transmission units (MTUs) large, therefore the MSRP SEND message that contains the payload is segmented into a number of data packets.

7.3.12.22.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Amount of user data transferred (in kbit)	The users use the file/media sharing enabler to send a file with known size.	User A sends a file with known size to user B.
t(connection established)	Start: When the actual transmission of the file/media starts. At this moment, the user	Start: Protocol: MSRP The MSRP SEND message containing the file data is transmitted.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
	is given a file/media start of transfer notification.	
t(content sent)	Stop: The successful reception of the file, which results in a file/media delivered notification.	Stop: Protocol: MSRP The terminal receives the "MSRP 200 OK" that acknowledges the reception of the file.

7.3.12.23 MTSI media set-up time [s]

7.3.12.23.1 Abstract definition

The MTSI media set-up time is the (non-negative) time period between the successful set-up of the signalling part of the MTSI call set-up and the receipt of the first packet containing valid (i.e., expected) media payload.

Figure 7-49 shows message flow for MTSI media set-up, delayed due to NAT devices.

7.3.12.23.2 Abstract equation

$$\text{MTSI media set-up time} = \text{Max}\left[\left(t_{\text{first valid media packet received}} - t_{\text{successful signalling setup}}\right), 0\right]$$

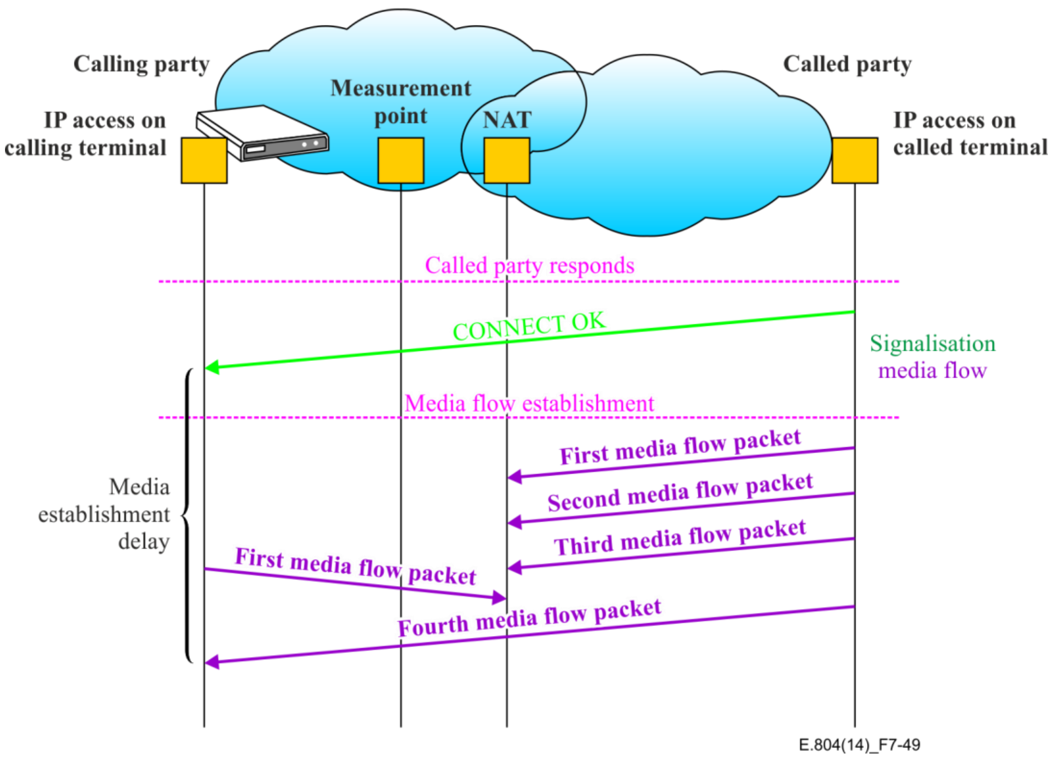


Figure 7-49 – MTSI media set-up

Remarks:

- In most cases the media path will be opened at the same time as the signalling path; for instance when there are no network address translator (NAT) devices in the call path, or when the NAT devices are managed by the operator (for instance the session border controller (SBC)) and opened up automatically during the signalling phase. In such cases the media delay might be zero, or even negative (any negative values should however be set to zero for this parameter).

- If non-managed NAT devices are present in the call path, it is the responsibility of the terminals to open these by sending media or by using protocols such as ICE [b-IETF RFC 5245] or STUN [b-IETF RFC 5389]. In such cases the media set-up time might be substantially larger than zero, depending on the methods used to open the NAT pinholes.

7.3.12.23.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{successful signalling setup}}$	Start: The user receives a notification that the other phone accepts the invitation.	Start: Protocol: SIP. First SIP 200 OK received after initiating a session.
$t_{\text{first valid media packet received}}$	Stop: The media is played out to the user.	Stop: Protocol: SIP. First valid media packet received.

7.3.12.24 MTSI media add time [s]

7.3.12.24.1 Abstract definition

The MTSI media add time is the (non-negative) time period between the successful change of a session (adding a media component), and the receipt of the first packet containing valid (i.e., expected) payload for the new media component.

Remark:

- The terminals involved must have an MTSI session ongoing before it can be modified.

7.3.12.24.2 Abstract equation

$$\text{MTSI media add time} = \text{Max}\left[\left(t_{\text{first valid media packet received}} - t_{\text{successful signalling setup}}\right), 0\right]$$

Remark:

- The MTSI media add time is similar to the MTSI media set-up time, except that the terminals will already have at least one media session open. Depending on the NAT structure in the call path, the time until the first media packet might be zero or even negative (when NATs are already open due to the existing media session) or significant (when NAT pinholes need to be opened by the terminals). Any negative values should be set to zero for this parameter.

7.3.12.24.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{successful signalling setup}}$	Start: The user receives a notification that the other phone accepts the added media invitation.	Start: Protocol: SIP. First SIP 200 OK is received after initiating the session change.
$t_{\text{first valid media packet received}}$	Stop: The added media is played out to the user.	Stop: Protocol: SIP. First valid media packet for the added media is received.

7.3.13 E-mail

Please refer to clause 7.2, as the parameters described there are usable for direct service as well if notification is disabled on the e-mail server.

All QoS parameters from clause 7.2 can be used with the exception of those dealing with notification (see clauses 7.2.10 and 7.2.11).

7.3.14 Group call

7.3.14.1 Group call service non-accessibility [%]

7.3.14.1.1 Abstract definition

The group call service non-accessibility is the probability that the end user cannot access the group call service when requested by pushing the push to talk (PTT) button.

Figure 7-50 provides group call set-up procedure.

7.3.14.1.2 Abstract equation

$$\text{Group call service non-accessibility}\left[\%\right]=\frac{\text{unsuccessful group call attempts}}{\text{all group call attempts}}\times 100$$

7.3.14.1.3 Trigger points

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Group call attempt	Start: Push the PTT button.	Start: Layer 3 (CMCE): The "U-SETUP" message is sent from the A-party. AT: The "ATD <dial string>" command is sent from the A-party, where <dial string> provides a unique identification of the desired group. A preceding "AT+CTSDC" command is used to set the correct parameters for the dial command.
Successful group call attempt	Stop: The acoustic and/or optical indication is given to the A-party user that the group call is established.	Stop: Layer 3 (CMCE): The "D-CONNECT" message is sent from the SwMI to the A-party. AT: The "AT+CTCC" indication is received by the A-party.
Unsuccessful call attempt	Stop trigger point not reached.	
NOTE – For the group call service non-accessibility, it is not necessary to check the possibly involved B-parties (other group members) for a set-up indication, e.g., a "D-SETUP" message, because the group call is actually established towards the network, i.e., the SwMI – no matter if there is any B-party connected to the group call or not.		

Preconditions for measurement:

Precondition	Covered by	Reference document
CS network available	Radio network unavailability	
CS attach successful		
No active group call		

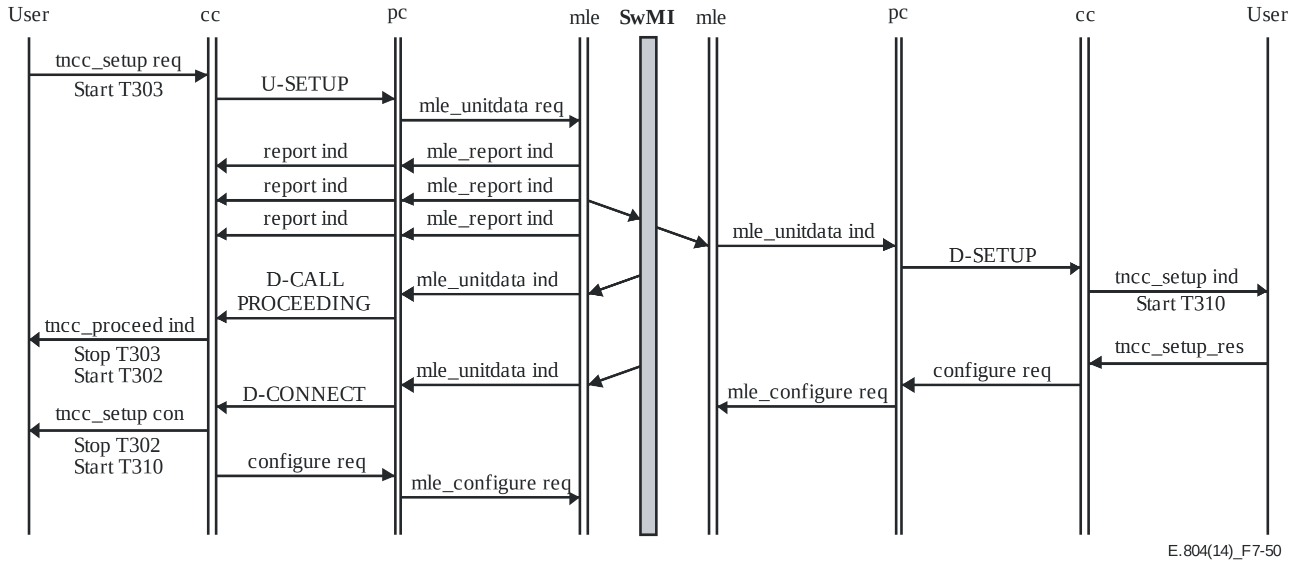


Figure 7-50 – Group call set-up procedure

7.3.14.2 Group call set-up time [s]

7.3.14.2.1 Abstract definition

The group call set-up time is the time period between pushing the push to talk (PTT) button at the UE and receipt of the call set-up notification by an acoustical and/or optical indication at the UE that the group call is successfully established.

7.3.14.2.2 Abstract equation

$$\text{Group call set-up time}\,[s] = \left(t_{\text{connection established}} - t_{\text{user pressed button}}\right)[s]$$

7.3.14.2.3 Trigger points

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{\text{user pressed button}}$: Time of call attempt	Start: Push the PTT button.	Start: Layer 3 (CMCE): The "U-SETUP" message is sent from the A-party. AT: The "ATD <dial string>" command is sent from the A-party, where <dial string> provides a unique identification of the desired group. A preceding "AT+CTSDC" command is used to set the correct parameters for the dial command.
$t_{\text{connection established}}$: Time when connection is established (i.e., successful call attempt)	Stop: The acoustic and/or optical indication is given to the A-party user that the group call is established.	Stop: Layer 3 (CMCE): The "D-CONNECT" message is sent from the SwMI to the A-party. AT: The "AT+CTCC" indication is received by the A-party.

Preconditions for measurement:

Precondition	Covered by	Reference document
CS network available	Radio network unavailability	
CS attach successful		
CS service access successful	Group call service non-accessibility	

7.3.14.3 Group call speech quality on call basis

7.3.14.3.1 Abstract definition

The group call speech quality on call basis is an indicator representing the quantification of the end-to-end speech transmission quality of the group call service. This parameter computes the speech quality on the basis of completed calls.

NOTE 1 – The acoustic behaviour of the terminals is not part of this speech quality measurement.

NOTE 2 – The speech quality in group calls is measured at any receiving B-party, i.e., at every group member in the receiving state. Thus, the overall speech quality for one group call may vary among the receiving B-party UEs. It is up to the following analysis to aggregate and evaluate the different results.

7.3.14.3.2 Abstract equation

The applicability of a suitable speech quality evaluation method for the narrow-band speech codec within TETRA networks is for further study.

7.3.14.3.3 Trigger points

The group call speech quality on call basis is derived from speech transmission during the duration of the entire group call. Trigger points are therefore not defined for the speech quality on call basis itself but for the group call duration according to the definitions for an intentionally terminated group call in clause 7.3.14.5.3.

7.3.14.4 Group call speech quality on sample basis

7.3.14.4.1 Abstract definition

The group call speech quality on sample basis is an indicator representing the quantification of the end-to-end speech transmission quality of the group call service. This parameter computes the speech quality on a sample basis.

NOTE 1 – The acoustic behaviour of terminals is not part of this speech quality measurement.

NOTE 2 – The speech quality in group calls is measured at any receiving B-party, i.e., at every group member in the receiving state. Thus, speech quality for one audio sample may vary among the receiving B-party UEs. It is up to the following analysis to aggregate and evaluate the different results.

7.3.14.4.2 Abstract equation

The applicability of a suitable speech quality evaluation method for the narrow-band speech codec within TETRA networks is for further study.

7.3.14.4.3 Trigger points

The group call speech quality on sample basis is derived from the speech samples transmitted during the duration of the entire group call. Trigger points are therefore not defined for the speech quality on sample basis itself but for the group call duration according to the definitions for an intentionally terminated group call in clause 7.3.14.5.3.

7.3.14.5 Group call cut-off call ratio [%]

7.3.14.5.1 Abstract definition

The group call cut-off ratio is the probability that a successful call attempt is ended by a cause other than the intentional termination by the A- or B-party.

NOTE – In TETRA, a B-party may in special situations request a group call disconnection. Those instances should be excluded from the group call cut-off call ratio.

7.3.14.5.2 Abstract equation

$$\text{Group call cut-off call ratio}\,[\%] = \frac{\text{unintentionally terminated group calls}}{\text{all successful group call attempts}} \times 100$$

7.3.14.5.3 Trigger points

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful group call attempt	Start: The acoustic and/or optical indication is given to the A-party user that the group call is established.	Start: Layer 3 (CMCE): The "D-CONNECT" message is sent from the SwMI to the A-party. AT: The "AT+CTCC" indication is received by the A-party.
Intentionally terminated group call	Stop: Final release of the PTT button by any group member (A-party or involved B-parties).	Stop: Layer 3 (CMCE): The last "U-TX CEASED" message is sent by the latest active party. AT: The last "AT+CUTXC=1" command is sent by the latest active party.
Unintentionally terminated group call	A premature call disconnection	Stop trigger not reached.
NOTE 1 – A group call may contain several phases of exchanging speech samples between A-party and B-parties. Within the speech transmission phases, the roles of A-party and involved B-parties vary in terms of speech transmission originating or terminating side. NOTE 2 – For the group call cut-off call ratio all actively involved B-parties, i.e., other group members connected to the established group call, are considered reflecting the end-to-end experience of the participating group call members, i.e., users.		

7.3.14.6 Group call speech transmission delay [s]

7.3.14.6.1 Abstract definition

The group call speech transmission delay describes the time period between a UE sending speech data and the group member UEs receiving the speech data for a unique talk burst or speech sample within a successfully established group call.

NOTE – The speech transmission delay in group calls is measured from the initiating A-party to any receiving B-party, i.e., to every group member in receiving state. Thus, the speech transmission delay for one instance of audio may vary among the receiving B-parties. It is up to the following analysis to aggregate and evaluate the different results.

7.3.14.6.2 Abstract equation

$$\text{Group call speech transmission delay}[s] = (t_{B,listen} - t_{A,speak})[s]$$

7.3.14.6.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{A,\ speak}$: Time of sending speech at the A-party	Start: A-party issues a talk burst.	Start: Audio interface: A unique audio signal is sent by the A-party.
$t_{B,\ listen}$: Time of receiving speech at the B-party	Stop: B-party hears the talk burst.	Stop: Audio interface: The very same audio signal is received by the B-party.
NOTE – Since every audio signal has certain duration and is therefore sent over a period of time, start and stop trigger points should both either refer to the beginning or the end of this audio signal. In case the speech transmission delay is derived from the transmission of speech samples, the same applies to this particular kind of audio signal.		

7.4 Store-and-forward services QoS parameters

The "store-and-forward" concept can be used for every non real time service called "Background Class", which uses the following communication concept. Two clients are assumed and one or more servers in the middle for each service.

- The A-party uploads a message to a server.
- This server forwards the message to another server (this step is optional).
- The server notifies the B-party that a new message is available (this step is optional).
- The B-party downloads the message.

The user's experience is similar for all services which follow the "store-and-forward" approach.

7.4.1 Generic store-and-forward parameters

The QoS parameter concept presented in this clause should be used for all services that work as described in the introduction of clause 7. Services that use proprietary or encrypted communication between the user equipment and the server of the service are predestinated to use the following generic parameter concept.

7.4.1.1 Parameter overview chart

Figure 7-51 gives an overview of the QoS parameters and their trigger points used in this generic parameter concept. The blue part describes the upload part of a message from the A-party to a server. The green part describes the notification part. The B-party will be informed about a new message. At the end, the message will be downloaded at the B-party side from a server, described by the orange boxes. Empty parameter boxes indicate that the parameter is not yet defined.

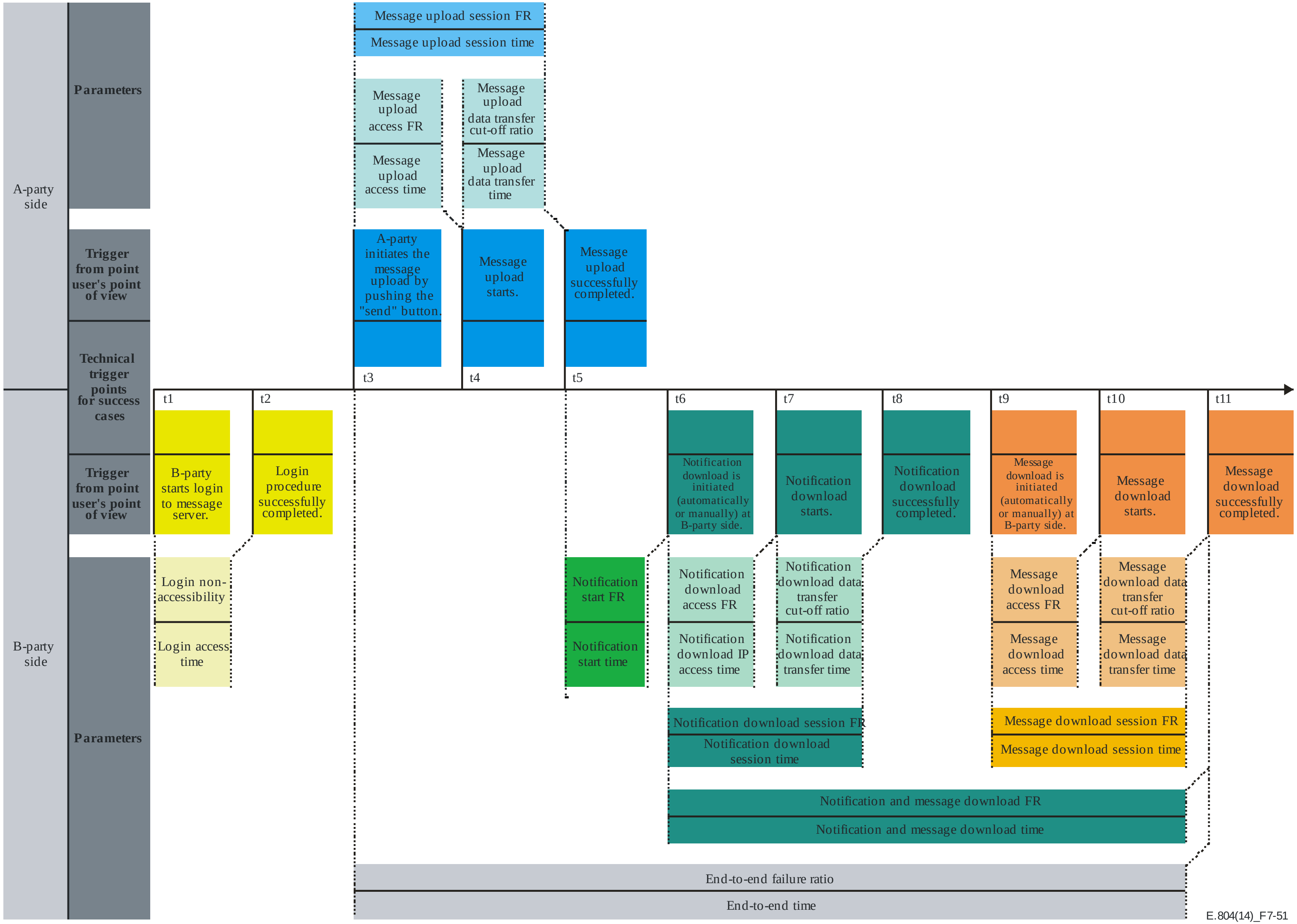


Figure 7-51 – Generic store-and-forward parameter overview

7.4.1.2 (Service) Message upload session failure ratio [%]

7.4.1.2.1 Abstract definition

The message upload session failure ratio is the proportion of unsuccessful message upload sessions and message upload sessions that were started successfully. The upload is successful if the message is marked as sent.

7.4.1.2.2 Abstract equation

$$\text{(Service) Message upload session failure ratio}\,[\%] = \frac{\text{unsuccessful message upload sessions}}{\text{all message upload session start attempts}} \times 100$$

7.4.1.2.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Message upload session start attempt	A-party initiates the message upload by pushing the "send" message button.
Successful message upload session	Message upload is successfully completed.
Unsuccessful message upload session	Stop trigger point not reached.

7.4.1.3 (Service) Message upload session time [s]

7.4.1.3.1 Abstract definition

The message upload session time is the time period needed to successfully complete a message upload session.

7.4.1.3.2 Abstract equation

$$\text{(Service) Message upload session time}\,[s] = \left(t_{\text{successful message upload session}} - t_{\text{message upload session start attempt}}\right)[s]$$

7.4.1.3.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Message upload session start attempt	A-party initiates the message upload by pushing the "send" message button.
Successful message upload session	Message upload is successfully completed.

Precondition for measurement: Message upload shall be successful.

7.4.1.4 (Service) Message upload access failure ratio [%]

7.4.1.4.1 Abstract definition

The message upload access failure ratio is the probability that the user cannot successfully establish a data connection to the message server to upload the messages.

7.4.1.4.2 Abstract equation

$$\text{(Service) Message upload access failure ratio}\,[\%] = \frac{\text{unsuccessful message upload accesses}}{\text{all message upload access attempts}} \times 100$$

7.4.1.4.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Message upload access attempt	A-party initiates the message upload by pushing the "send" message button.
Successful message upload access	Message upload starts.
Unsuccessful message upload access	Stop trigger point not reached.

7.4.1.5 (Service) Message upload access time [s]

7.4.1.5.1 Abstract definition

The message upload access time is the time period needed to establish a data connection to the message server, from sending the initial query to the message server to the point of time when the message upload starts.

7.4.1.5.2 Abstract equation

$$\text{(Service) Message upload access time}\,[\mathrm{s}] = \left(t_{\text{successful message upload access}} - t_{\text{message upload access attempt}}\right)[\mathrm{s}]$$

7.4.1.5.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Message upload access attempt	A-party initiates the message upload by pushing the "send" message button.
Successful message upload access	Message upload starts.

Precondition for measurement: Message upload access shall be successful.

7.4.1.6 (Service) Message upload data transfer cut-off ratio [%]

7.4.1.6.1 Abstract definition

The message upload data transfer cut-off ratio describes the proportion of unsuccessful message uploads and message uploads that were started successfully.

7.4.1.6.2 Abstract equation

$$\text{(Service) Message upload data transfer cut-off ratio}\,[\%] = \frac{\text{unsuccessful message uploads}}{\text{all successfully started message uploads}} \times 100$$

7.4.1.6.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Successfully started message upload	Message upload starts at the A-party side.
Successful message upload	Message upload is successfully completed.
Unsuccessful message upload	Stop trigger point not reached.

7.4.1.7 (Service) Message upload data transfer time [s]

7.4.1.7.1 Abstract definition

The message upload data transfer time is the time period from the start to the end of the complete message upload.

7.4.1.7.2 Abstract equation

$$\text{(Service) Message upload data transfer time}[s] = \left(t_{\text{successful message upload}} - t_{\text{successfully started message upload}}\right)[s]$$

7.4.1.7.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Successfully started message upload	Message upload starts at A-party side.
Successful message upload	Message upload successfully completed.

Precondition for measurement: Message upload data transfer shall be successful.

7.4.1.8 (Service) Notification start failure ratio [%]

7.4.1.8.1 Abstract definition

The notification start failure ratio is the probability that the notification download by the B-party is not successfully initiated after the successful upload of the message by the A-party.

7.4.1.8.2 Abstract equation

$$\text{(Service) Notification start failure ratio}[\%] = \frac{\text{unsuccessful notification download attempts by B-party}}{\text{all successful message uploads by A-party}} \times 100$$

7.4.1.8.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Successful message upload by A-party	Message upload successfully completed by A-party.
Notification download attempt by B-party	Notification download is initiated (automatically or manually) at the B-party side.
Unsuccessful notification download attempt by B-party	Stop trigger point not reached.

7.4.1.9 (Service) Notification start time [s]

7.4.1.9.1 Abstract definition

The notification start time is the time period from the successful message upload by the A-party to the start of the notification download attempt by the B-party.

7.4.1.9.2 Abstract equation

$$\text{(Service) Notification start time}[s] = \left(t_{\text{notification download attempt by B-party}} - t_{\text{successful message upload by A-party}}\right)[s]$$

7.4.1.9.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Successful message upload by A-party	Message upload successfully completed by the A-party.
Notification download attempt by the B-party	Notification download is initiated (automatically or manually) at the B-party side.

Precondition for measurement: Notification download attempt shall be successful.

7.4.1.10 (Service) Notification download session failure ratio [%]

7.4.1.10.1 Abstract definition

The notification download session failure ratio is the proportion of unsuccessful notification downloads and notification downloads that were started successfully.

7.4.1.10.2 Abstract equation

$$\text{(Service) Notification download session failure ratio}\,[\%] = \frac{\text{unsuccessful notification download sessions}}{\text{all notification download session start attempts}} \times 100$$

7.4.1.10.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Notification download session start attempt	Notification download is initiated (automatically or manually) at B-party side.
Successful notification download session	Notification download successfully completed.
Unsuccessful notification download session	Stop trigger point not reached.

7.4.1.11 (Service) Notification download session time [s]

7.4.1.11.1 Abstract definition

The notification download session time is the time period needed to successfully complete a notification download session.

7.4.1.11.2 Abstract equation

$$\text{(Service) Notification download session time}\,[s] = \left(t_{\text{successful notification download session}} - t_{\text{notification download session start attempt}}\right)[s]$$

7.4.1.11.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Notification download session start attempt	Notification download is initiated (automatically or manually) at the B-party side.
Successful notification download session	Notification download successfully completed.

Precondition for measurement: Message notification download shall be successful.

7.4.1.12 (Service) Notification download access failure ratio [%]

7.4.1.12.1 Abstract definition

The notification download access failure ratio is the probability that the user cannot successfully establish a data connection to the message server to download the notification of a new message.

7.4.1.12.2 Abstract equation

$$\text{(Service) Notification download access failure ratio}\,[\%] = \frac{\text{unsuccessful notification download accesses}}{\text{all notification download access attempts}} \times 100$$

7.4.1.12.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Notification download access attempt	Notification download is initiated (automatically or manually) at the B-party side.
Successful notification download access	Notification download starts.
Unsuccessful notification download access	Stop trigger point not reached.

7.4.1.13 (Service) Notification download access time [s]

7.4.1.13.1 Abstract definition

The notification download access time is the time period needed to establish the data connection to the message server, from sending the initial query to the message server to the point of time when the notification download starts.

7.4.1.13.2 Abstract equation

$$\text{(Service) Notification download access time}[s] = \left(t_{\text{successful notification download access}} - t_{\text{notification download access attempt}}\right)[s]$$

7.4.1.13.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Notification download access attempt	Notification download is initiated (automatically or manually) at the B-party side.
Successful notification download access	Notification download starts.

Precondition for measurement: Notification download access shall be successful.

7.4.1.14 (Service) Notification download data transfer cut-off ratio [%]

7.4.1.14.1 Abstract definition

The notification download data transfer cut-off ratio is the proportion of unsuccessful notification downloads and notification downloads that were started successfully.

7.4.1.14.2 Abstract equation

$$\text{(Service) Notification download data transfer cut-off ratio}[\%] = \frac{\text{unsuccessful notification downloads}}{\text{all successfully started notification downloads}} \times 100$$

7.4.1.14.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Successfully started notification download	Notification download starts at the B-party side.
Successful notification download	Notification download is successfully completed.
Unsuccessful notification download	Stop trigger point not reached.

7.4.1.15 (Service) Notification download data transfer time [s]

7.4.1.15.1 Abstract definition

The notification download data transfer time describes the time period from the start to the end of the complete notification download.

7.4.1.15.2 Abstract equation

$$\text{(Service) Notification data transfer time}[s] = \left(t_{\text{successful notification download}} - t_{\text{successfully started notification download}}\right)[s]$$

7.4.1.15.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Successfully started notification download	Notification download starts at the B-party side.
Successful notification download	Notification download is successfully completed.

Precondition for measurement: Notification data transfer shall be successful.

7.4.1.16 (Service) Message download session failure ratio [%]

7.4.1.16.1 Abstract definition

The message download session failure ratio is the proportion of unsuccessful message download sessions and message download sessions that were started successfully.

7.4.1.16.2 Abstract equation

$$\text{(Service) Message download session failure ratio}[\%] = \frac{\text{unsuccessful message download sessions}}{\text{all message download session start attempts}} \times 100$$

7.4.1.16.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Message download session start attempt	Message download is initiated (automatically or manually) at the B-party side.
Successful message download session	Message download is successfully completed.
Unsuccessful message download session	Stop trigger point not reached.

7.4.1.17 (Service) Message download session time [s]

7.4.1.17.1 Abstract definition

The message download session time describes the time period needed to successfully complete a message download session.

7.4.1.17.2 Abstract equation

$$\text{(Service) Message download session time}[s] = \left(t_{\text{successful message download session}} - t_{\text{message download session start attempt}}\right)[s]$$

7.4.1.17.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Message download session start attempt	Message download is initiated (automatically or manually) at the B-party side.
Successful message download session	Message download is successfully completed.

Precondition for measurement: Message download shall be successful.

7.4.1.18 (Service) Message download access failure ratio [%]

7.4.1.18.1 Abstract definition

The message download access failure ratio is the probability that the user cannot successfully establish a data connection to the message server to download messages.

7.4.1.18.2 Abstract equation

$$\text{(Service) Message downoad access failure ratio}[\%] = \frac{\text{unsuccessful message download accesses}}{\text{all message download access attempts}} \times 100$$

7.4.1.18.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Message download access attempt	Message download is initiated (automatically or manually) at the B-party side.
Successful message download access	Message download starts.
Unsuccessful message download access	Stop trigger point not reached.

7.4.1.19 (Service) Message download access time [s]

7.4.1.19.1 Abstract definition

The message download access time is the time period needed to establish a data connection to the message server, from sending the initial query to the message server to the point of time when the message download starts.

7.4.1.19.2 Abstract equation

$$\text{(Service) Message download access time}[s] = \left(t_{\text{successful message download access}} - t_{\text{message download access attempt}}\right)[s]$$

7.4.1.19.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Message download access attempt	Message download is initiated (automatically or manually) at the B-party side.
Successful message download access	Message download starts.

Precondition for measurement: Message download access shall be successful.

7.4.1.20 (Service) Message download data transfer cut-off ratio [%]

7.4.1.20.1 Abstract definition

The message download data transfer cut-off ratio is the proportion of unsuccessful message downloads and message downloads that were started successfully.

7.4.1.20.2 Abstract equation

$$\text{(Service) Message download data transfer cut-off ratio}[\%] = \frac{\text{unsuccessful message downloads}}{\text{all successfully started message downloads}} \times 100$$

7.4.1.20.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Successfully started message download	Message download starts at the B-party side.
Successful message download	Message download is successfully completed.
Unsuccessful message download	Stop trigger point not reached.

7.4.1.21 (Service) Message download data transfer time [s]

7.4.1.21.1 Abstract definition

The message download data transfer time is the time period from the start to the end of the complete message download.

7.4.1.21.2 Abstract equation

$$\text{(Service) Message download data transfer time}[s] = \left(t_{\text{successful message download}} - t_{\text{successfully started message download}}\right)[s]$$

7.4.1.21.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Successfully started message download	Message download starts at the B-party side.
Successful message download	Message download is successfully completed.

Precondition for measurement: Message download data transfer shall be successful.

7.4.1.22 (Service) Notification and message download failure ratio [%]

7.4.1.22.1 Abstract definition

The notification and message download failure ratio is the probability that the user cannot download first the notification and thereafter the complete message with the UE. User reaction times are not considered.

7.4.1.22.2 Abstract equation

$$\text{(Service) Notification and message download failure ratio}[\%] = \frac{\text{unsuccessful notification and message downloads}}{\text{all notification and message download attempts}} \times 100$$

7.4.1.22.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Notification and message download attempt	Notification download is initiated (automatically or manually) at the B-party side.
Successful notification and message download	Message download is successfully completed.
Unsuccessful notification and message download	Stop trigger point not reached.

7.4.1.23 (Service) Notification and message download time [s]

7.4.1.23.1 Abstract definition

The notification and message download time is the time period from the start of the notification download to the end of the reception of the whole message content. User reaction times are not considered.

7.4.1.23.2 Abstract equation

$$\text{(Service) Notification and message download time}[s] = (t_{\text{successful notification and message download}} - t_{\text{notification and message download attempt}})[s]$$

7.4.1.23.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Notification and message download attempt	Notification download is initiated (automatically or manually) at the B-party side.
Successful notification and message download	Message download is successfully completed.

Precondition for measurement: Notification and the message download shall be successful.

7.4.1.24 (Service) End-to-end failure ratio [%]

7.4.1.24.1 Abstract definition

The end-to-end failure ratio is the probability that the complete service usage from the start of the message upload at the A-party to the complete message download at the B-party cannot be completed successfully. This transmission is unsuccessful if the message upload, the notification (if possible) or the message download fails.

7.4.1.24.2 Abstract equation

$$\text{(Service) End-to-end failure ratio}[\%] = \frac{\text{unsuccessful message downloads by B-party}}{\text{all message upload attempts by A-party}} \times 100$$

7.4.1.24.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Message upload attempt	A-party initiates the message upload by pushing the "send" message button.
Successful message download	Message download is successfully completed at B-party side.
Unsuccessful message download	Stop trigger point not reached.

7.4.1.25 (Service) End-to-end time [s]

7.4.1.25.1 Abstract definition

The end-to-end time is the time period needed for the complete service usage, from the start of the message upload at the A-party to the complete message download at the B-party.

7.4.1.25.2 Abstract equation

$$\text{(Service) End-to-end time}[s] = (t_{\text{successful message download}} - t_{\text{message upload attempt}})[s]$$

7.4.1.25.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Message upload attempt	A-party initiates the message upload by pushing the "send" message button.
Successful message download	Message download is successfully completed at the B-party side.

Precondition for measurement: End-to-end service usage shall be successful.

7.4.1.26 (Service) Login non-accessibility [%]

7.4.1.26.1 Abstract definition

The login non-accessibility is the probability of a login failure between the message client and the message server. The login is needed to prepare the client of the B-party to be able to receive new notifications or messages. The parameter does not consider an actual message transfer.

7.4.1.26.2 Abstract equation

$$\text{(Service)Login non-accessibility}[\%] = \frac{\text{unsuccessful logins}}{\text{all login attempts}} \times 100$$

7.4.1.26.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Login attempt	B-party starts login to the message server.
Successful login	Login procedure is successfully completed.
Unsuccessful login	Stop trigger point not reached.

7.4.1.27 (Service) Login access time [s]

7.4.1.27.1 Abstract definition

The login access time is the time period from starting the login procedure to the point of time when the login procedure is successfully completed and the client can receive notifications or messages at the B-party side.

7.4.1.27.2 Abstract equation

$$\text{(Service) Login access time}\,[s] = \left(t_{\text{successfullogin}} - t_{\text{loginattempt}}\right)\,[s]$$

7.4.1.27.3 Trigger points

Event from abstract equation	Trigger point from user's point of view
Login attempt	B-party starts login to the message server.
Successful login	Login procedure is successfully completed.

Precondition for measurement: Login shall be successful.

7.4.2 E-mail

7.4.2.1 Parameter overview chart

Figures 7-52 to 7-55 give an overview of the QoS parameters used in the e-mail concept based on the SMTP, IMAP4 and POP3 protocol.

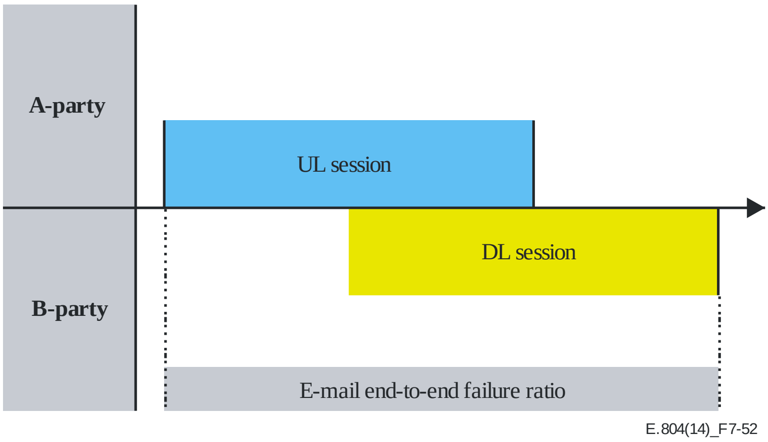


Figure 7-52 – End-to-end session overview

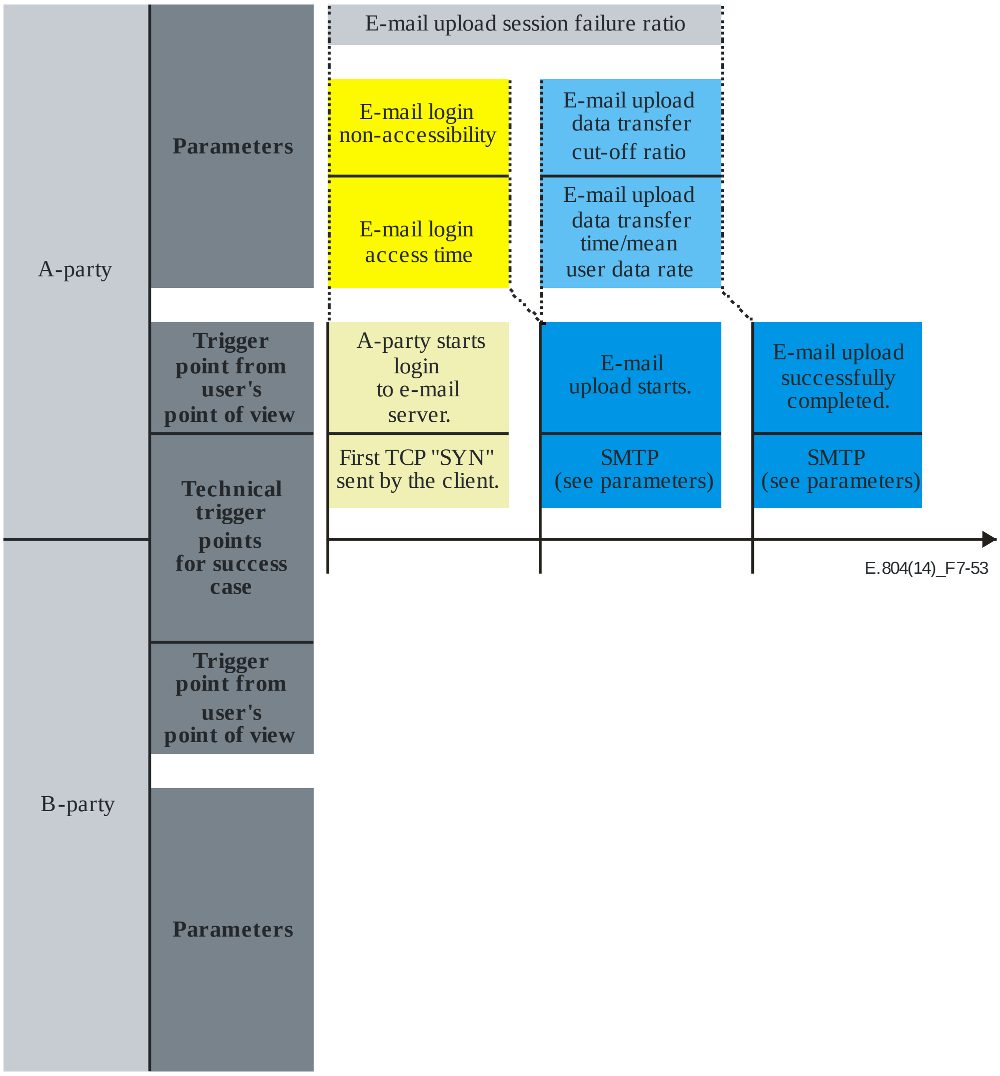


Figure 7-53 – SMTP Overview

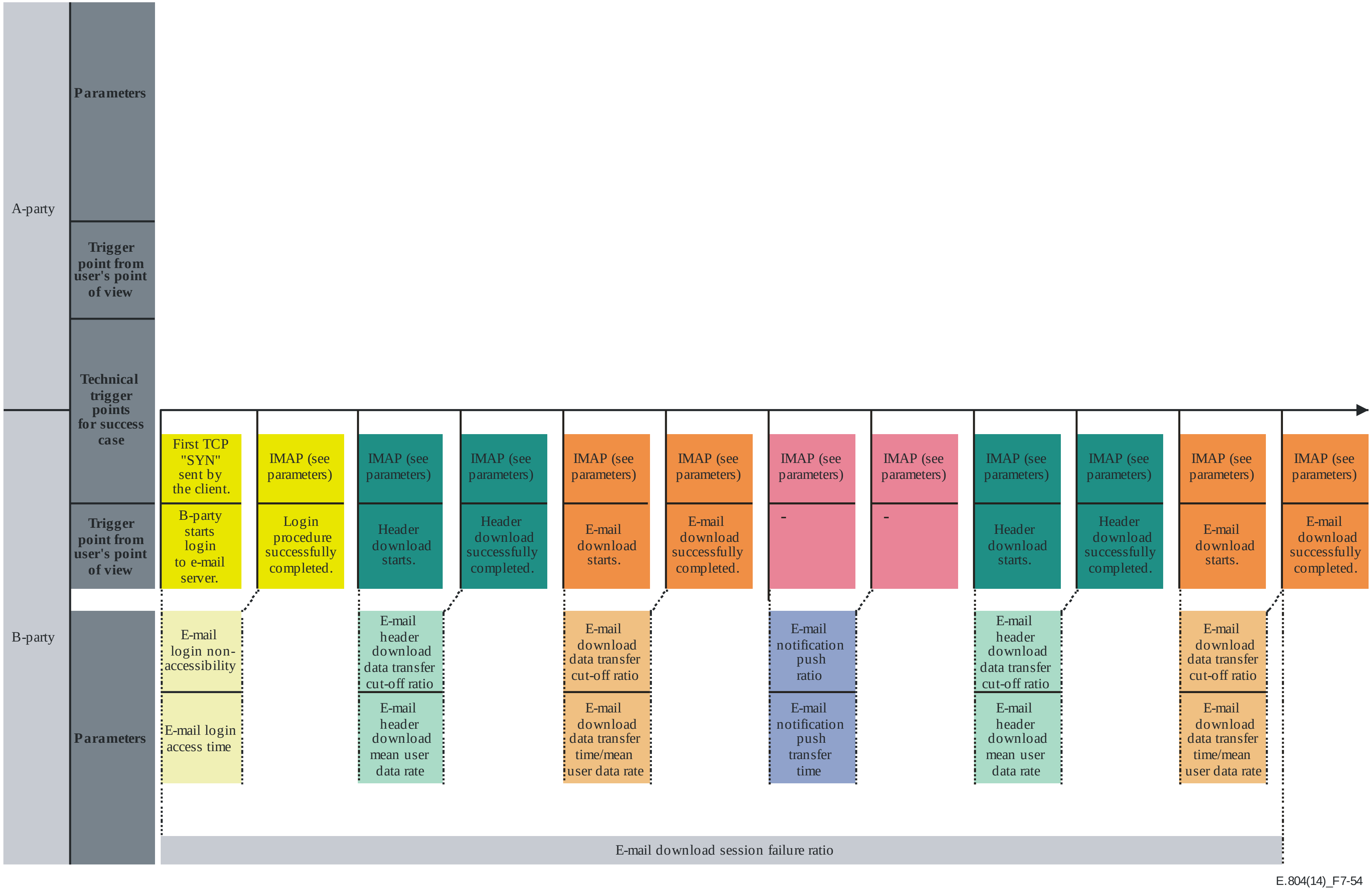


Figure 7-54 – IMAP 4 (including idle feature) parameter overview

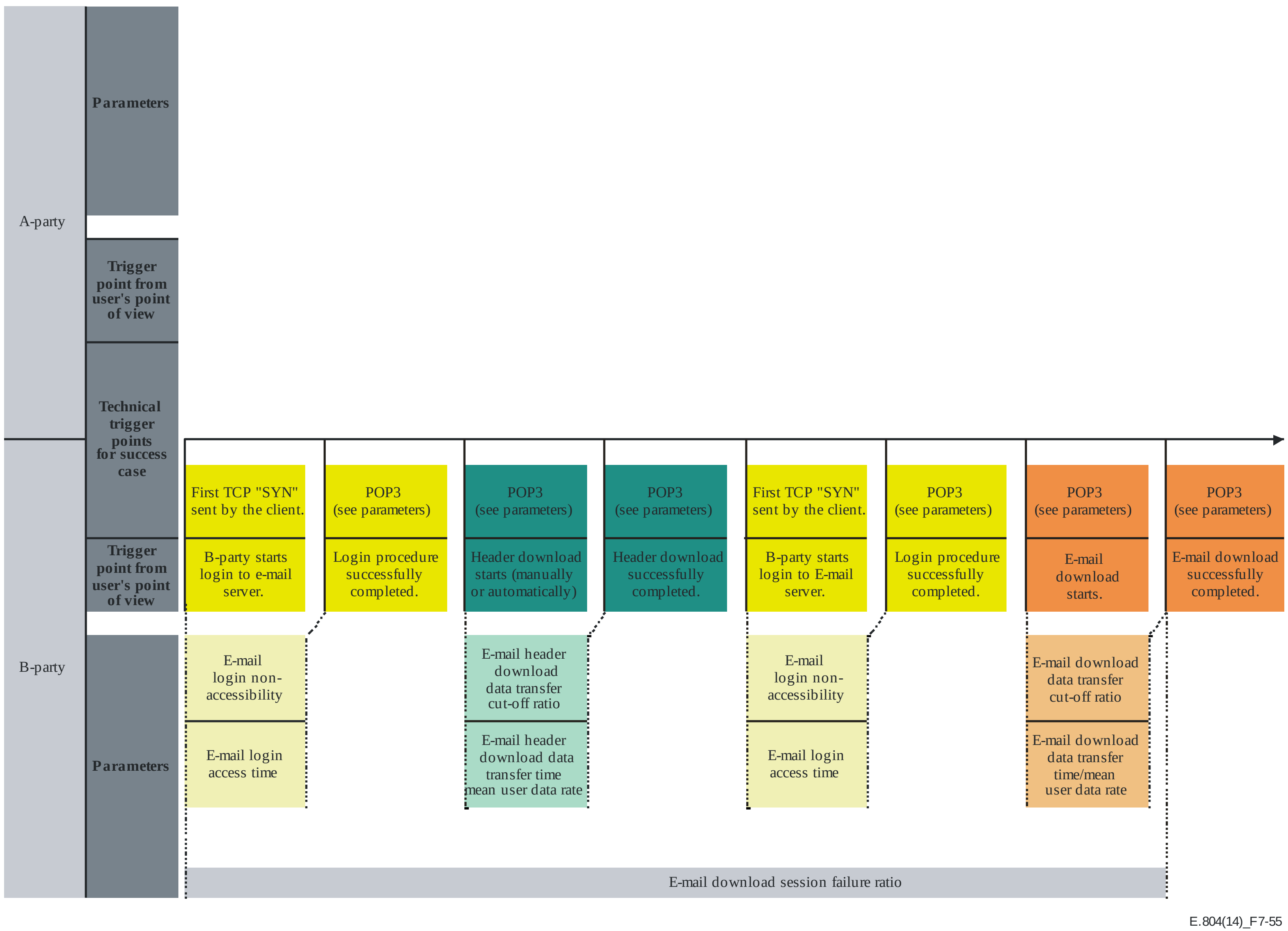


Figure 7-55 – POP3 parameter overview

7.4.2.2 E-mail {download|upload} service non-accessibility [%]

This parameter was removed due to major changes in the e-mail QoS concept.

7.4.2.3 E-mail {download|upload} set-up time [s]

This parameter was removed due to major changes in the e-mail QoS concept.

7.4.2.4 E-mail {download|upload} IP-service access failure ratio [%]

This parameter was replaced by the "login non-accessibility" parameter specified in clause 7.4.2.11.

7.4.2.5 E-mail {download|upload} IP-service set-up time [s]

This parameter was replaced by the "login non-accessibility" parameter specified in clause 7.4.2.11.

7.4.2.6 E-mail {download|upload} session failure ratio [%]

7.4.2.6.1 Abstract definition

The e-mail {download|upload}session failure ratio is the proportion of unsuccessful sessions and sessions that were started successfully.

7.4.2.6.2 Abstract equation

$$\text{E-mail}\ \{\text{download} \mid \text{upload}\}\ \text{session failure ratio}\ [\%] = \frac{\text{unsuccessful sessions}}{\text{all session start attempts}} \times 100$$

7.4.2.6.3 Trigger points

Upload:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Session start attempt	Start: A-party starts login to the e-mail server.	Start: TCP: First "SYN" is sent by the client.
Successful session	Stop: E-mail upload is successfully completed by the A-party.	Stop: SMTP: Reply code "250 message accepted" received by the client. An e-mail upload session can consist of several uploads.
Unsuccessful session	Stop trigger point not reached.	

Download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Session start attempt	Start: B-party starts login to the e-mail server.	Start: First TCP "SYN" is sent by the client.
Successful session	Stop: E-mail download successfully completed by the B-party.	Stop: POP3: Termination sequence <CRLF.CRLF> received by the client as an answer to the "RETR" command. IMAP4: "OK FETCH completed" received by the client. An e-mail download session can consist of several FETCH/RETR/TOP requests (body and/or header downloads). All successful requests shall be confirmed accordingly.
Unsuccessful session	Stop trigger point not reached.	

Remark:

- The PS bearer has to be active in the cell used by a user (see clause 7.2.1) and the UE has to be attached (see clause 7.2.3) as well as the respective PDP context has to be activated (see clause 7.2.5).

7.4.2.7 E-mail {upload|header download|download} session time [s]

This parameter was removed due to the fact that the significance of the parameter is weak due to the following factors:

- Different e-mail client implementations behave quite differently during a session with respect to the POP3/IMAP4 commands they send to the e-mail server.
- In certain use cases (e.g., header download first) user interaction is required to resume the session.

Both points have considerable influence on the measured results.

7.4.2.8 E-mail {upload|header download|download} mean data rate [kbit/s]

7.4.2.8.1 Abstract definition

The e-mail mean data rate is the average data transfer rate measured throughout the entire connect time to the e-mail service. The data transfer shall be successfully terminated.

7.4.2.8.2 Abstract equation

$$\text{E-mail}\{\text{upload}|\text{ download}\}\text{ mean data rate}[\text{kbit/s}]=\frac{\text{user data transfered}[\text{kbit}]}{\left(t_{\text{successful data transfer}}-t_{\text{successfully started data transfer}}\right)[\text{s}]}$$

7.4.2.8.3 Trigger points

Upload:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started data transfer	Start: E-mail upload starts.	Start: SMTP: "MAIL FROM" is sent by the client.
Successful data transfer	Stop: E-mail upload is successfully completed by the A-party.	Stop: SMTP: Reply "250 message accepted" received by the client.

Header download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started data transfer	Start: Header download starts.	Start: POP3: "TOP" command is sent by the client. IMAP4: "UID FETCH" command is sent by the client to request the header.
Successful data transfer	Stop: Header download is successfully completed by the B-party.	Stop: POP3: Termination sequence <CRLF.CRLF> received by the client. IMAP4: "OK FETCH completed" is received by the client. A header download can consist of several FETCH/TOP requests. All successful requests shall be confirmed accordingly.

Download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started data transfer	Start: E-mail download starts.	Start: POP3: "RETR" command is sent by the client. IMAP4: "UID FETCH" command is sent by the client to request header and body.
Successful data transfer	Stop: E-mail download is successfully completed by the B-party.	Stop: POP3: Termination sequence <CRLF.CRLF> is received by the client. IMAP4: "OK FETCH completed" received by the client. An e-mail download can consist of several FETCH/RETR requests. All successful requests shall be confirmed accordingly.

Preconditions for measurement: The PS bearer has to be active in the cell used by a user (see clause 7.2.1), the UE has to be attached (see clause 7.2.3), the respective PDP context has to be activated (see clause 7.2.5) and the login to the e-mail server was successful (see clause 7.4.2.11).

7.4.2.9 E-mail {upload|header download|download} data transfer cut-off ratio [%]

7.4.2.9.1 Abstract definition

The e-mail data transfer cut-off ratio is the proportion of unsuccessful data transfers and data transfers that were started successfully.

7.4.2.9.2 Abstract equation

$$\text{E-mail}\left\{\text{upload}\,\middle|\,\text{header download}\,\middle|\,\text{download}\right\}\text{ data transfer cut-off ratio}[\%]=\frac{\text{unsuccessful data transfers}}{\text{all successfully started data transfers}}\times 100$$

7.4.2.9.3 Trigger points

Upload:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started data transfer	Start: E-mail upload starts.	Start: SMTP: "MAIL FROM" is sent by the client.
Successful data transfer	Stop: E-mail upload is successfully completed by the A-party.	Stop: SMTP: Reply "250 OK, message accepted" is received by the client.
Unsuccessful data transfer	Stop trigger point not reached.	

Header download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started data transfer	Start: Header download starts.	Start: POP3: "TOP" command is sent by the client. IMAP4: "UID FETCH" command is sent by the client to request the header.
Successful data transfer	Stop: Header download is successfully completed by the B-party.	Stop: POP3: Termination sequence <CRLF.CRLF> is received by the client as an answer to the "TOP" command. IMAP4: "OK Fetch complete" is received by the client. A header download can consist of several FETCH/TOP requests. All successful requests shall be confirmed accordingly.
Unsuccessful data transfer	Stop trigger point not reached.	

Download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started data transfer	Start: E-mail download starts.	Start: POP3: "RETR" command is sent by the client. IMAP4: "UID FETCH" command is sent by the client to request the header and the body.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful data transfer	Stop: E-mail download is successfully completed by the B-party.	Stop: POP3: Termination sequence <CRLF.CRLF> is received by the client as an answer to the "RETR" command. IMAP4: "OK Fetch complete" is received by the client. An e-mail download can consist of several fetch requests. All successful requests shall be confirmed by "OK Fetch completed". An e-mail download can consist of several FETCH/RETR requests. All successful requests shall be confirmed accordingly.
Unsuccessful data transfer	Stop trigger point not reached.	

Preconditions for measurement: The PS bearer has to be active in the cell used by a user (see clause 7.2.1), the UE has to be attached (see clause 7.2.3), the respective PDP context has to be activated (see clause 7.2.5) and the login to the e-mail server was successful (see clause 7.4.2.11).

7.4.2.10 E-mail {upload|header download|download} data transfer time [s]

7.4.2.10.1 Abstract definition

The e-mail data transfer time is the time period from the start to the end of the complete transfer of e-mail content.

7.4.2.10.2 Abstract equation

$$\text{E-mail}\left\{\text{upload}\middle|\ \text{header download}\middle|\ \text{download}\right\}\text{data transfer time}[s] = \left(t_{\text{successful data transfer}} - t_{\text{successfully started data transfer}}\right)[s]$$

7.4.2.10.3 Trigger points

Upload:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started data transfer	Start: E-mail upload starts.	Start: SMTP: "MAIL FROM" is sent by the client.
Successful data transfer	Stop: E-mail upload is successfully completed the A-party.	Stop: SMTP: Reply "250 message accepted" is received by the client.

Header download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started data transfer	Start: Header download starts.	Start: POP3: "TOP" command sent by the client. IMAP4: "UID FETCH" command is sent by the client to request the header.
Successful data transfer	Stop: Header download is successfully completed by the B-party.	Stop: POP3: Termination sequence <CRLF.CRLF> is received by the client. IMAP4: "OK Fetch completed" is received by the client. A header download can consist of several FETCH/TOP requests. All successful requests shall be confirmed accordingly.

Download:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully started data transfer	Start: E-mail download starts.	Start: POP3: "RETR" command is sent by the client. IMAP4: "UID FETCH" command is sent by the client to request the header and the body.
Successful data transfer	Stop: E-mail download is successfully completed by the B-party.	Stop: POP3: Termination sequence <CRLF.CRLF> is received by the client as an answer to the "RETR" command. IMAP4: "OK Fetch completed" is received by the client. An e-mail download can consist of several FETCH/RETR requests. All successful requests shall be confirmed accordingly.

Preconditions for measurement: The PS bearer has to be active in the cell used by a user (see clause 7.2.1), the UE has to be attached (see clause 7.2.3), the respective PDP context has to be activated (see clause 7.2.5) and the login to the e-mail server was successful (see clause 7.4.2.11).

7.4.2.11 E-mail login non-accessibility [%]

7.4.2.11.1 Abstract definition

The e-mail login non-accessibility is the probability that the e-mail client is not able to get access to the e-mail server.

7.4.2.11.2 Abstract equation

$$\text{E-mail Login non-accessibility}[\%] = \frac{\text{unsuccessful logins}}{\text{all login attempts}} \times 100$$

7.4.2.11.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Login attempt	Start: User starts login to the e-mail server.	Start: TCP: First "SYN" is sent by the client.
Successful login	Stop: Login procedure is successfully completed.	Stop: SMTP: Reply "235 Authentication successful" is received by the client as an answer to the authentication request. IMAP4: Reply "OK AUTHENTICATE successful" is received by the client as an answer to the authentication request. POP3: "+OK" is received by the client as an answer to the authentication request.
Unsuccessful login	Stop trigger point not reached.	

Preconditions for measurement: The PS bearer has to be active in the cell used by a user (see clause 7.2.1) and the UE has to be attached (see clause 7.2.3).

7.4.2.12 E-mail login access time [s]

7.4.2.12.1 Abstract definition

The e-mail login access time is the time period from starting the login procedure to the point of time when the client is authenticated.

7.4.2.12.2 Abstract equation

$$\text{E-mail login access time}\,[s] = \left(t_{\text{successful login}} - t_{\text{login attempt}}\right)[s]$$

7.4.2.12.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Login attempt	Start: User starts login to the e-mail server.	Start: TCP: First "SYN" is sent by the client.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successful login	Stop: Login procedure successfully completed.	Stop: SMTP: Reply "235 Authentication successful" is received by the client as an answer to the authentication request. IMAP4: Reply "OK AUTHENTICATE successful" is received by the client as an answer to the authentication request. POP3: "+OK" is received by the client as an answer to the authentication request.

Preconditions for measurement: The PS bearer has to be active in the cell used by a user (see clause 7.2.1) and the UE has to be attached (see clause 7.2.3).

7.4.2.13 E-mail notification push failure ratio [%]

7.4.2.13.1 Abstract definition

The e-mail notification push failure ratio is the probability that the notification announcement was not successfully conveyed to the B-party.

7.4.2.13.2 Abstract equation

$$\text{E-mail notification push failure ratio}\,[\%] = \frac{\text{unsuccessful attempts to push the notification to the B-party}}{\text{all attempts to push the notification to the B-party}} \times 100$$

7.4.2.13.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Notification push attempt	Start: Not applicable.	Start: IMAP4: "EXISTS" command is received by the client.
Successful idle complete	Stop: Not applicable.	Stop: IMAP4: "OK IDLE complete" is received by the client.
Unsuccessful idle complete	Stop trigger point not reached.	

Preconditions for measurement: The PS bearer has to be active in the cell used by a user (see clause 7.2.1), the UE has to be attached (see clause 7.2.3), the respective PDP context has to be activated (see clause 7.2.5), the login to the e-mail server was successful (see clause 7.4.2.11) and the e-mail upload was successful (see clause 7.4.2.6).

7.4.2.14 E-mail notification push transfer time [s]

7.4.2.14.1 Abstract definition

The e-mail notification push transfer time is the time period from starting the notification push to the successful confirmation of the e-mail server of the end of the idle period.

7.4.2.14.2 Abstract equation

$$\text{E-mail notification push transfer time}[s] = \left(t_{\text{successful idle complete}} - t_{\text{notification push attempt}}\right)[s]$$

7.4.2.14.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Notification push attempt	Start: Not applicable.	Start: IMAP4: "EXISTS" command is received by the client.
Successful idle complete	Stop: Not applicable.	Stop: IMAP4: "OK IDLE complete" is received by the client.

Preconditions for measurement: The PS bearer has to be active in the cell used by a user (see clause 7.2.1), the UE has to be attached (see clause 7.2.3), the respective PDP context has to be activated (see clause 7.2.5), the login to the e-mail server was successful (see clause 7.4.2.11) and the e-mail upload was successful (see clause 7.4.2.6).

7.4.2.15 E-mail end-to-end failure ratio [%]

7.4.2.15.1 Abstract definition

The e-mail end-to-end failure ratio is the probability that the complete service usage from the start of the e-mail upload at the A-party to the complete e-mail download at the B-party with an e-mail client cannot be completed successfully. This transmission is unsuccessful if the e-mail upload, the header download (if applicable) or the e-mail download fails.

7.4.2.15.2 Abstract equation

$$\text{E-mail end-to-end failure ratio}[\%] = \frac{\text{unsuccessful e-mail downloads by the B-party}}{\text{all e-mail upload attempts by the A-party}} \times 100$$

7.4.2.15.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
E-mail upload attempt by the A-party	Start: The A-party starts login to the e-mail server.	Start: TCP: First "SYN" sent by the client.
Successful e-mail download by the B-party	Stop: E-mail download is successfully completed by the B-party.	Stop: POP3: Termination sequence <CRLF.CRLF> is received by the client. IMAP4: "OK FETCH completed" is received by the client. An e-mail download can consist of several FETCH/RETR requests. All successful requests shall be confirmed accordingly.
Unsuccessful e-mail download by B-party	Stop trigger point not reached.	

Preconditions for measurement: The PS bearer has to be active in the cell used by a user (see clause 7.2.1) and the UE has to be attached (see clause 7.2.3).

7.4.2.16 Exemplary signal flow

The following signal flows are examples. The signalling between client and server can differ. It depends on the used client and the server type.

7.4.2.16.1 SMTP e-mail upload

Event	Trigger	Client	Server
TCP connection set-up	1	SYN	
	2		SYN, ACK
	3	ACK	
Request for capabilities	4	EHLO	
	5		250 Hello [...capability list…]
Authentication	6	AUTH [...]	
	7		334
		…authentication challenge between client and server…	
	8		235 Authentication successful

Event	Trigger	Client	Server
E-mail upload	9	MAIL FROM:<name@domain.com>	
	10		250 OK
	11	RCPT TO:<name2@domain.com>	
	12		250 OK
	13	DATA	
	14		354 Start mail input
		... header and body data is sent from client to server…	
	15	<CRLF>.<CRLF>	
	16		250 OK, message accepted
Logout	17	QUIT	
	18		221 Closing connection

7.4.2.16.2 IMAP4 idle header and e-mail download

Event	Trigger	Client	Server
TCP connection set-up	1	SYN	
	2		SYN, ACK
	3	ACK	
	4		* OK *IMAP server ready*
Request for capabilities	5	001 CAPABILITY	
	6		* CAPABILITY […capability list…]
	7		001 OK CAPABILITY complete
Authentication	8	002 AUTHENTICATE [...]	
	9		+ Go ahead
		…authentication challenge between client and server…	
	10		002 OK *AUTHENTICATE successful*
Synchronization	11	003 LIST "" ""	
	12		* LIST […]
	13		003 OK LIST completed
	14	004 SELECT "INBOX"	
	15		* 2 EXISTS
	16		* 0 RECENT
	17		* FLAGS (\Seen [..])
	18		[…]
	19		004 OK *SELECT complete*

Event	Trigger	Client	Server
Activation idle mode	20	005 IDLE	
	21		+ *IDLE*
		…time passes; new mail arrives at server…	
New e-mail arrived at the server	22		* 3 EXISTS
	23	DONE	
	24		005 OK *IDLE complete*
Request for UID number, method differs	25	006 FETCH 3 (UID)	
	26		* 3 FETCH (UID 4711)
	27		006 OK *FETCH complet*e
Header download	28	007 UID FETCH 4711 BODY[HEADER]	
	29		* 4711 FETCH (BODY[HEADER] {123}
	30		Date: […] From: […] Subject: […] To: […] cc: […] Message-Id: […]
	31		007 OK *FETCH completed*
E-mail header and body download	32	008 UID FETCH 4711 (UID FLAGS BODY.PEEK[])	
	33		* 1 FETCH (UID 4711 FLAGS (\Recent) Body [] {123456}
	34		Return-Path: name@domain.com
		…header and body data is sent from server to client…	
	35		008 OK *FETCH completed*
Delete	36	009 UID STORE 4711 +flags \deleted	
	37		* 3 FETCH (FLAGS (\Seen \Deleted))
	38		009 OK +*FLAGS completed*
	39	010 EXPUNGE	
	40		010 OK *Expunge completed*
Logout	41	011 LOGOUT	
	42		* Bye
	43		011 OK *LOGOUT completed*

7.4.2.16.3 POP3 header download

Event	Trigger	Client	Server
TCP connection set-up	1	SYN	
	2		SYN, ACK
	3	ACK	
	4		+OK POP3 server ready
Request for capabilities	5	AUTH	
	6		+OK List of supported SASL authentication methods follows: […authentication mechanism list…]
	7	CAPA	
	8		+OK Capability list follows: […capability list…]
Authentication	9	AUTH [...]	
		…authentication challenge between client and server…	
	10		+OK 1 message, 1500 octets
Synchronization	11	STAT	
	12		+OK 1 1500
	13	LIST	
	14		+OK Scan list follows 1 1500 <CRLF>.<CRLF>
E-mail header download	15	TOP 1 0	
	16		+OK Message top follows
		…header data is sent from server to client…	
	17		<CRLF>.<CRLF>
Logout	18	QUIT	
	19		+OK

7.4.2.16.4 POP3 e-mail download

Event	Trigger	Client	Server
TCP connection set-up	1	SYN	
	2		SYN, ACK
	3	ACK	
	4		+OK Server ready

Event	Trigger	Client	Server
Request for capabilities	5	AUTH	
	6		+OK List of supported SASL authentication methods follows: […authentication mechanism list…]
	7	CAPA	
	8		+OK Capability list follows: […capability list…]
Authentication	9	AUTH [...]	
		…authentication challenge between client and server…	
	10		+OK 1 message, 1500 octets
Synchronization	11	STAT	
	12		+OK 1 1500
	13	LIST	
	14		+OK Scan list follows 1 1500 <CRLF>.<CRLF>
	15	UIDL	
	16		+OK Scan list follows 1 12 <CRLF>.<CRLF>
E-Mail header and body download	17	RETR 1	
	18		+OK 1500 octets
		…header and body data is sent from server to client…	
	19		<CRLF>.<CRLF>
Delete	20	DELE 1	
	21		+OK Message deleted
Logout	22	QUIT	
	23		+OK Closing connection

7.4.3 Multimedia messaging service (MMS)

NOTE 1 – It is important to keep in mind that measurement equipment and techniques used can affect the data collected. The measurement equipment and techniques should be defined and their effects documented for all tests. One example of this is the effect of Windows remote access service (RAS) on the set-up of PDP Context.

NOTE 2 – The underlying transport mechanism can be either WAP1.x or WAP2.0. Figure 7-56 provides MMS parameter overview with trigger points.

7.4.3.1 Parameter overview chart

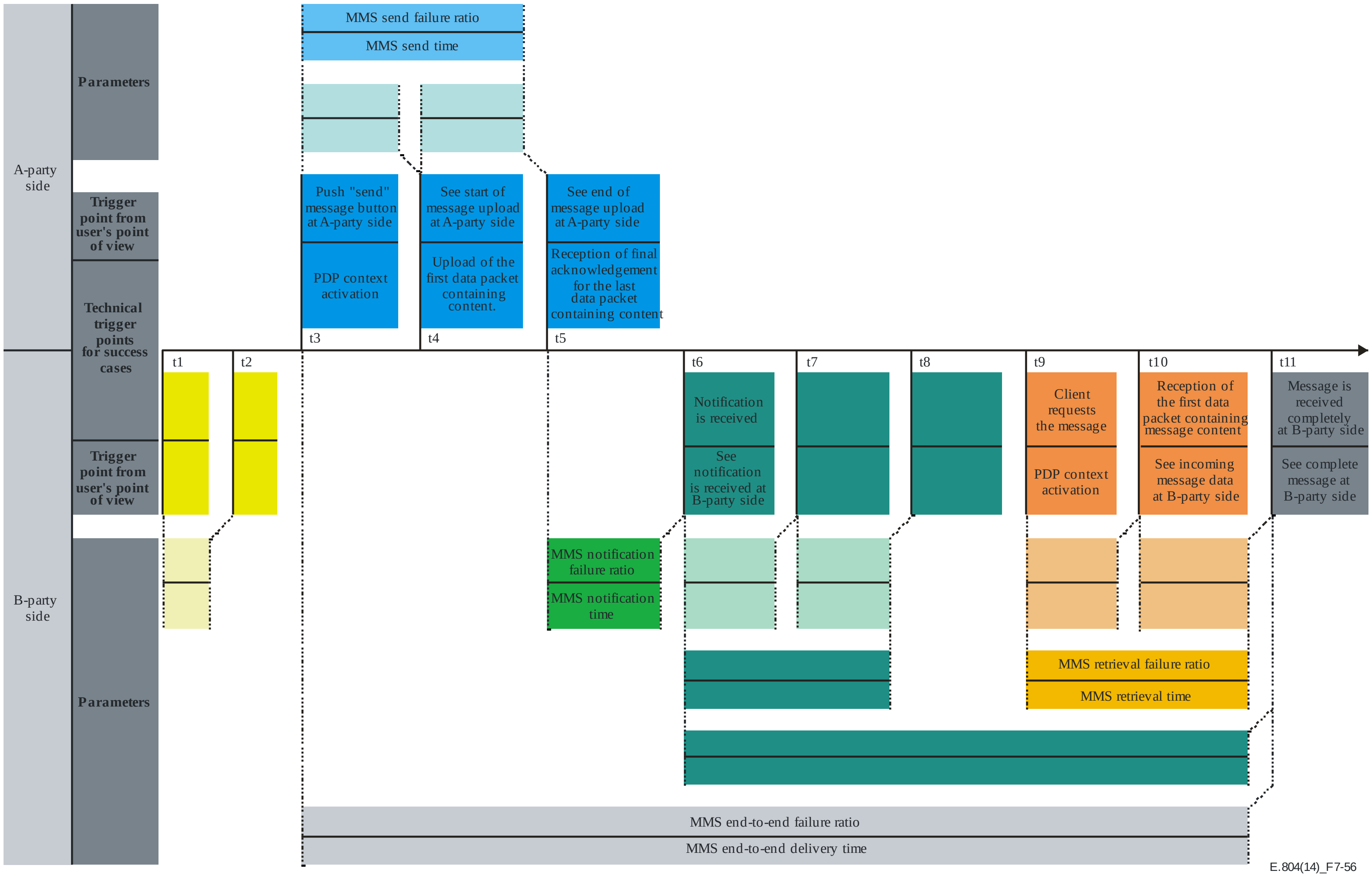


Figure 7-56 – MMS parameter overview with trigger points

7.4.3.2 MMS send failure ratio [%]

7.4.3.2.1 Abstract definition

The parameter MMS send failure ratio is the probability that a MMS message cannot be sent by the subscriber, although it has been requested to do so by pushing the "send button".

7.4.3.2.2 Abstract equation

$$\text{MMS send failure ratio}[\%] = \frac{\text{unsuccessful MMS send attempts}}{\text{all MMS send attempts}} \times 100$$

7.4.3.2.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
MMS send attempt	Pushing of the send button.	The send button initiates the *PDP context activation* of the MS (MO), followed by a connection to the WAP gateway, and to the MMSC. (See trigger 1 in Figure 7-57).
Unsuccessful MMS send attempt	Do not see "Message sent".	The *m-send.conf* (see [b-WAP Forum]) (where Response Status: $80 = M_RS_OK) is not received by the MS (MO). (See trigger 18 in Figure 7-57). (See Notes 1 to 3). "MMS unsuccessful send attempt timeout".
NOTE 1 – The phase where the WAP session (WAP1.x)/TCP connection (WAP 2.0) will be deactivated is not covered by this indicator. Some mobile devices may not support the sending/receiving of the next MMS unless the WAP session (WAP1.x)/TCP connection (WAP 2.0) is disconnected properly. NOTE 2 – A forwarding of a MMS without reception of a positive m-send.conf (where Response Status: $80 = M_RS_OK) shall be counted as failure. NOTE 3 – Only MMS sent within the timeouts will be considered.		

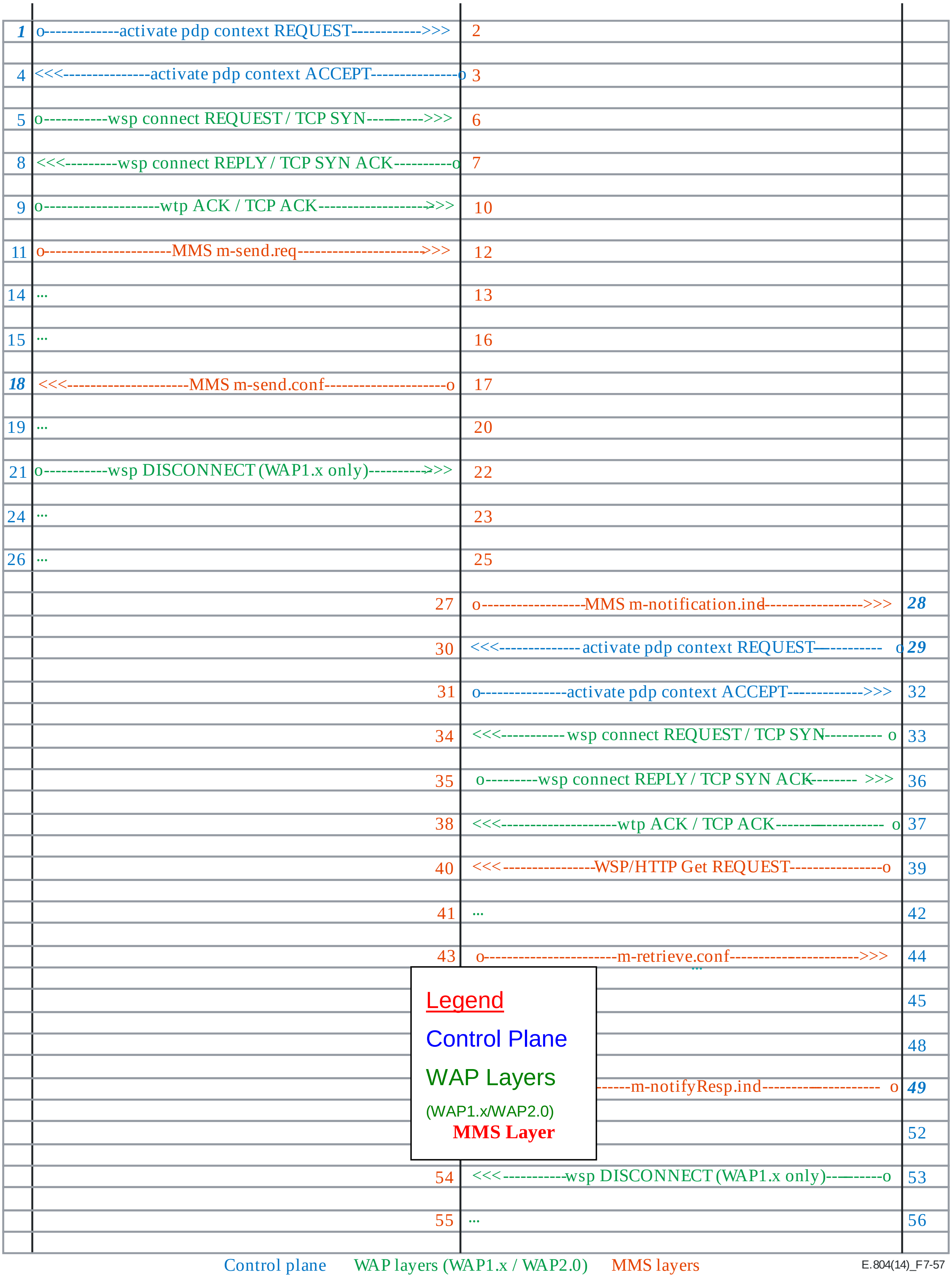


Figure 7-57 – MMS transaction flow (immediate retrieval)

NOTE – In Figure 7-57 only the transaction flow for immediate retrieval is shown. Refer to Figure 5 in [b-WAP Forum] for the delayed retrieval transaction flow.

7.4.3.3 MMS retrieval failure ratio [%]

7.4.3.3.1 Abstract definition

The parameter MMS retrieval failure ratio is the probability that the MMS message cannot be downloaded by the MT mobile device which received a MMS notification before.

Remark:

- The MMS notification is a push-message. This message either initiates the download of the MMS content by starting a "WAP Get Request" (when the mobile device is switched to automatic mode) or enables the user to manually start this "Wap Get Request" (when the mobile device is switched to manual mode). The measurements will be done either using the setting "Automatic Download" (e.g., the download follows the immediate retrieval transaction flow) or following the delayed retrieval. In case of delayed retrieval, the wait time between the notification response (m-notifyResp.ind) and the WSP/HTTP get request (WSP/HTTP Get.req) must be set to zero. Refer to Figure 5 in [b-WAP Forum] for the delayed retrieval transaction flow.

7.4.3.3.2 Abstract equation

$$\text{MMS delivery failure ratio}[\%] = \frac{\text{unsuccessful MMS delivery attempts}}{\text{all MMS delivery attempts}} \times 100$$

7.4.3.3.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
MMS retrieval attempt (MT)	Start: Initiation of the Wap Get Request MT.	Start: After the *m-Notification.ind*. (see [b-WAP Forum]) has been sent to the MS (MT), the mobile device activates a PDP-context and contacts the MMSC via the WAP Gateway (See trigger 29 in Figure 7-57).
Unsuccessful MMS retrieval attempt (MT)	Stop: No MMS-message is received.	Stop (immediate retrieval): The *m-notifyResp.ind* (see [b-WAP Form]) is not sent by the MS (MT). (See trigger 49 in Figure 7-57). (See Notes 1 and 2). "MMS unsuccessful Retrieval timeout" as specified in clause 10. Stop (deferred retrieval): The *m-acknowledge.ind* is not sent by the MS (MT).
NOTE 1 – The phase where the WAP session (WAP1.x)/TCP connection (WAP 2.0) will be deactivated is not covered by this indicator. Some mobile devices may not support the sending/receiving of the next MMS unless the WAP session (WAP1.x)/TCP connection (WAP 2.0) is disconnected properly. NOTE 2 – Only MMS received within the timeouts will be considered.		

7.4.3.4 MMS send time [s]

7.4.3.4.1 Abstract definition

The time elapsing from pushing the send button after the editing of a MMS message to the completion of the data transfer is given by this parameter. This is when a subscriber uses the multimedia messaging service (as indicated by the network ID on the mobile phone display).

NOTE – Possible measurement scenarios for time indicators of MMS may vary in the number of involved multimedia messaging service centres (MMSCs). With increasing MMS-traffic or internetwork-traffic surveillance, the number of MMSCs involved will also increase. Number of MMSCs involved is therefore a measurement condition to be discussed.

7.4.3.4.2 Abstract equation

$$\text{MMS send time}[s] = \left(t_{\text{MMS to MMS Complete}} - t_{\text{sendButton}}\right)[s]$$

7.4.3.4.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
sendButton	Start: The send button is pushed.	Start: The send button initiates the *PDP context activation* of the MS(MT), followed by a connection to the WAP Gateway (See trigger 1 in Figure 7-57). (See Notes 1 and 2). "MMS unsuccessful send transfer timeout" as specified in clause 10.
MMStoMMSCcomplete	Stop: MMS-message is completely transmitted to MMS-C.	Stop: The *m-send.conf* (see [b-WAP Forum]) (where Response Status: $80 = M_RS_OK) is received by the MS (MO). (See trigger 18 in Figure 7-57).
NOTE 1 – The phase, where the WAP session (WAP1.x)/TCP connection (WAP 2.0) will be deactivated is not covered by this indicator. Some mobile devices may not support the sending/receiving of the next MMS unless the WAP session (WAP1.x)/TCP connection (WAP 2.0) is disconnected properly. NOTE 2 – Only MMS sent within the timeouts will be considered.		

7.4.3.5 MMS retrieval time [s]

7.4.3.5.1 Abstract definition

The reception of a MMS message works as follows: A push-sms is sent to the receiver's mobile device. In automatic mode, the push sms initiates a WAP-connection to download the MMS from the MMS-C. The initiation of the WAP connection is called the "WAP Get Request (WGR)". The time elapsing between the WGR and the completion of the download of the MMS is given by the parameter MMS retrieval time.

Possible measurement scenarios for time indicators of MMS may vary in the number of involved MMSCs. With increasing MMS-traffic or internetwork-traffic surveillance, the number of MMSCs involved will also increase. Number of MMSCs involved is therefore a measurement condition to be discussed.

7.4.3.5.2 Abstract equation

$$\text{MMS delivery time}[s] = \left(t_{\text{MMS from MMS Complete}} - t_{\text{initWGR}}\right)[s]$$

7.4.3.5.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
initWGR	Start: Time when WAP Get Request is initiated.	Start: The *m-Notification.ind* (see [b-WAP Forum] is delivered to the MS (MT). This initiates the *PDP context activation.* (See trigger 29 in Figure 7-57).
MMSfromMMSC complete	Stop: MMS-message is received completely.	Stop (immediate retrieval): The *m-notifyResp.Ind* (see [b-WAP Forum]) is sent by the MS (MT). (See trigger 49 in Figure 7-57). (See Notes 1 and 2) "MMS successful retrieval timeout" as specified in clause 10. Stop (deferred retrieval): The *m-acknowledge.ind* is sent by the MS (MT).
NOTE 1 – The phase, where the WAP session (WAP1.x)/TCP connection (WAP 2.0) will be deactivated is not covered by this indicator. Some mobile devices may not support the sending/receiving of the next MMS unless the WAP session (WAP1.x)/TCP connection (WAP 2.0) is disconnected properly. NOTE 2 – Only MMS received within the timeouts will be considered.		

7.4.3.6 MMS notification failure ratio [%]

7.4.3.6.1 Abstract definition

The parameter MMS notification failure ratio [%] is the probability that the multimedia messaging service (MMS) is not able to deliver the notification of a MMS message to the B-parties mobile devices.

7.4.3.6.2 Abstract equation

$$\text{MMS notification failure ratio}[\%] = \frac{\text{failed MMS-notifications}}{\text{successfully submitted MMS}} \times 100$$

7.4.3.6.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Successfully submitted MMS MO	Start: Reception of the acknowledgement from the MMS-C MO (i.e., "Message sent").	Start: The *m-send.conf* (see [b-WAP Forum]) (where Response Status: $80 = M_RS_OK) is not received by the MS (MO). (See trigger 18 in Figure 7-57). (See Notes 1 and 2)
Failed MMS-Notifications	Stop: Failure delivery (non-delivery) of the Notification-SMS.	Stop: *m-notification.ind* (see [b-WAP Forum]) is not delivered to the MS (MT). (See trigger 28 in Figure 7-57). (See Note 3) "MMS successful notification timeout" as specified in clause 10.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
NOTE 1 – The phase where the WAP session (WAP1.x)/TCP connection (WAP 2.0) will be deactivated is not covered by this indicator. Some mobile devices may not support the sending/receiving of the next MMS unless the WAP session (WAP1.x)/TCP connection (WAP 2.0) is disconnected properly. NOTE 2 – Only the accepted MMS has to be considered (see the response status = $80 in the sendconf) MMS with negative response but delivered can be added alternatively. NOTE 3 – Only Notifications received within the timeouts will be considered as successful.		

7.4.3.7 MMS notification time [s]

7.4.3.7.1 Abstract definition

A subscriber uses the MMS, the time elapsing from the complete submission of the multimedia-message to the MMSC to the reception of the notification (MT) is the *MMS Notification Delay*.

Possible measurement scenarios for time indicators of MMS may vary in the number of involved MMSCs. With increasing MMS-traffic or internetwork-traffic surveillance, the number of MMSCs involved will also increase. Number of MMSCs involved is therefore a measurement condition to be discussed.

7.4.3.7.2 Abstract equation

$$\text{MMSnotification time}s[s] = \left(t_{recNotif} - t_{MMSsubmit}\right)[s]$$

7.4.3.7.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
MMSsubmit	Start: The MMS is submitted successfully.	Start: The *m-send.conf* (see [b-WAP Forum]), (where Response Status: $80 = M_RS_OK) is received by the MS (MO). (See trigger 18 in Figure 7-57). (See Note 1).
recNotif	Stop: Time when the notification is received (MT).	Stop: *M-Notification.ind* (see [b-WAP Forum]) is received by MS (MT) (See trigger 28 in Figure 7-57). (See Note 2). "MMS successful notification timeout" as specified in clause 10.
NOTE 1 – The phase, where the WAP session (WAP1.x)/TCP connection (WAP 2.0) will be deactivated is not covered by this indicator. Some mobile devices may not support the sending/receiving of the next MMS unless the WAP session (WAP1.x)/TCP connection (WAP 2.0) is disconnected properly. NOTE 2 – Only notifications received within the timeouts will be considered as successful.		

7.4.3.8 MMS end-to-end failure ratio [%]

7.4.3.8.1 Abstract definition

The parameter MMS end-to-end failure ratio is the probability that the multimedia messaging service (MMS) is not able to deliver a MMS message after the "send button" has been pushed or the MO party has not received an acknowledgement of the successful transmission from the MMSC.

7.4.3.8.2 Abstract equation

$$\text{MMS end-to-end failure ratio}[\%] = \frac{\text{unsuccessfully delivered MMS-messages}}{\text{all MMS send attempts}} \times 100$$

End-to-end parameter measurement may optionally be derived by concatenating the component measurements.

7.4.3.8.3 Trigger points

Event from abstract equation	**Trigger point from user's point of view**	**Technical description/ protocol part**
MMS send attempt by MS(MO)	Start: Pushing of the send button.	Start: The send button initiates the *PDP context activation* of the MS, followed by a connection to the WAP Gateway. (See trigger 1 in Figure 7-57). (See Note 1).
Unsuccessful MMS retrieval attempt of MS(MT)	Stop: No MMS-message is received (MT) or no acknowledgement from the MMSC is received at the MS (MO).	Stop: The *m-send.conf* (where Response Status: $80 = M_RS_OK) is not received by the MS (MO) or the *m-notifyResp.ind* (in case of immediate retrieval) respectively the *m-acklowledge*.ind (in case of deferred retrieval, see also [b-WAP Forum]) is not sent by the MS (MT). See trigger 18 and 49 in Figure 7-57 and Notes 2 and 3. MMS unsuccessful end-to-end timeout as specified in clause 10.
NOTE 1 – The forwarding of a MMS by the MMSC to the MS (MT) might be possible without the reception of the *m-send.conf* MS (MO) (see [b-WAP Forum]), (where response status is $80 = M RS OK). NOTE 2 – The phase where the WAP session (WAP1.x)/TCP connection (WAP 2.0) will be deactivated is not covered by this indicator. Some mobile devices may not support the sending/receiving of the next MMS unless the WAP session (WAP1.x)/TCP connection (WAP 2.0) is disconnected properly. NOTE 3 – Only MMS received within the timeouts will be considered.		

7.4.3.9 MMS end-to-end delivery time [s]

7.4.3.9.1 Abstract definition

A subscriber uses the MMS (as indicated by the network ID on the mobile phone display). The time elapsing from pushing of the "send button" to the reception of the MMS by the B-parties mobile is the MMS end-to-end delivery time.

This parameter is not calculated if the MO party has not received an acknowledgement of the successful transmission from the MMSC.

The size of a MMS varies. In comparison to SMS, the size has noticeable impact on the submission time. So, a typical size MM should be used for this measurement.

NOTE 1 – Possible measurement scenarios for time indicators of MMS may vary depending on the number of involved MMSCs. With increasing MMS-traffic or internetwork-traffic surveillance, the number of MMSCs involved will also increase. Number of MMSCs involved is therefore a measurement condition to be discussed.

NOTE 2 – End-to-end parameter measurement may optionally be derived by concatenating the component measurements.

7.4.3.9.2 Abstract equation

$$\text{MMS end-to-end delivery time}[s] = (t_{MMSrec} - t_{sendAttempt})[s]$$

7.4.3.9.3 Trigger points

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
sendAttempt	Start: Time when the "send button" is pushed.	Start: The send button initiates the *PDP context activation* of the MS (MO), followed by a connection to the WAP gateway. (See trigger 1 in Figure 7-57). (See Note 1).
MMSrec	Stop: Time when the MMS is received at the B-party's mobile device.	Stop: The M-retrieve.conf (see [b-WAP Forum]) is received completely by the MS (MT), and the MS (MT) sends the m-notifyResp.ind (See trigger 49 in Figure 7-57 in case of immediate retrieval) respectively the *m-acklowledge*.ind (in case of deferred retrieval). (See Notes 2 to 4). "MMS successful end-to-end timeout" as specified in clause 10.
NOTE 1 – The forwarding of a MMS by the MMSC to the MS (MT) might be possible without the reception of the *m-send.conf* MS (MO). NOTE 2 – Parameter not calculated if the m-send.conf (where Response Status: $80 = M_RS_OK) is not received by MS (MO) (See trigger 18 in Figure 7-57). NOTE 3 – The phase where the WAP session (WAP1.x)/TCP connection (WAP 2.0) will be deactivated is not covered by this indicator. Some mobiles devices may not support the sending/receiving of the next MMS unless the WAP session (WAP1.x)/TCP connection (WAP 2.0) is disconnected properly. NOTE 4 – Only MMS received within the timeouts will be considered.		

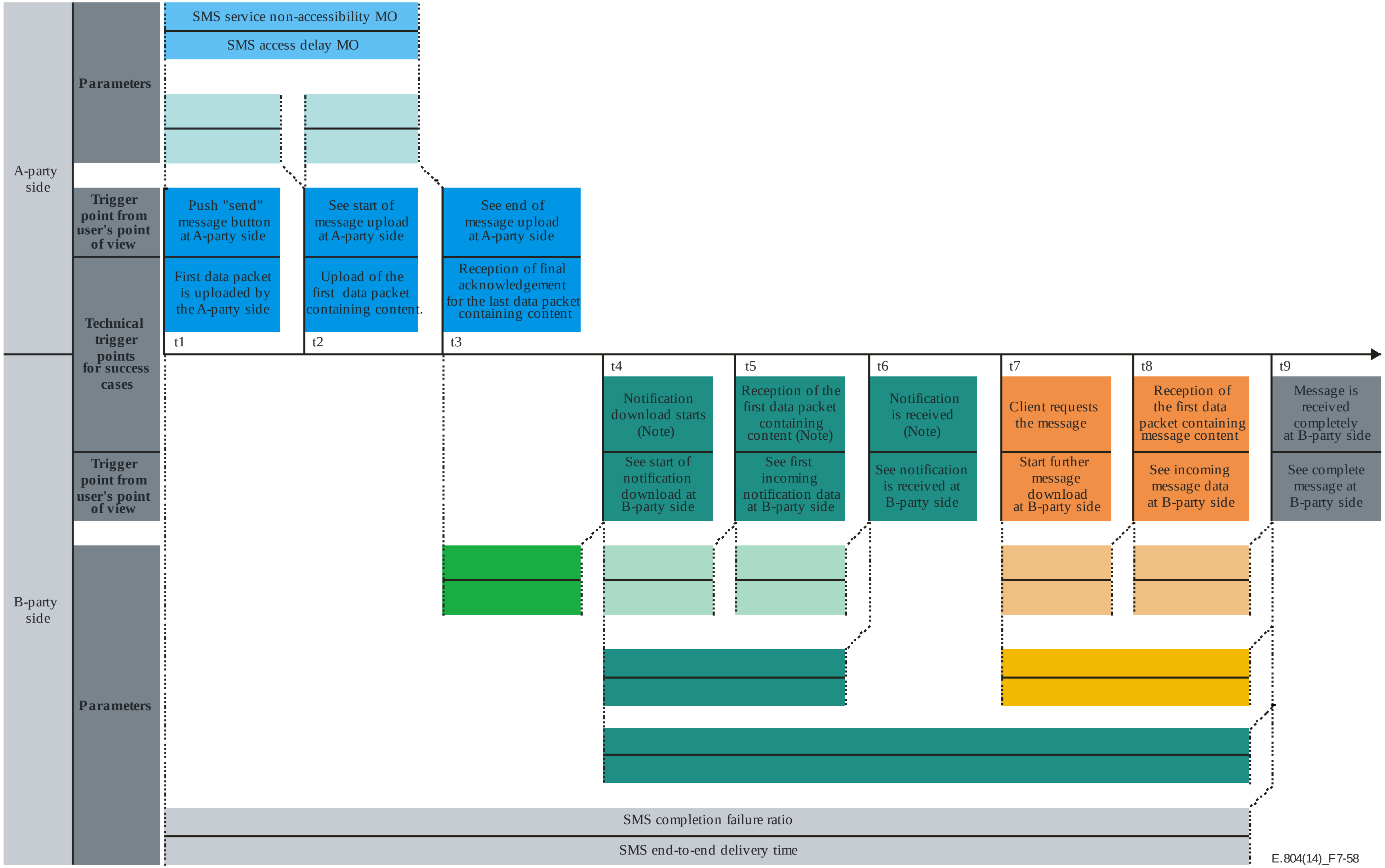
7.4.4 Short message service (SMS) and short data service (SDS)

The short message service (SMS) is available in GSM/UMTS networks, whereas the short data service (SDS) is available in TETRA networks. For both types of services the actual user defined data is referred as short message in the following.

NOTE – Four types of SDS are defined in [ETSI EN 300 392-2], SDS type 1 to SDS type 4. SDS type 1 offers 16 bit user defined data, SDS type 2 offers 32 bit user defined data, SDS type 3 offers 64 bit user defined data, and SDS type 4 offers user defined data bits up to a maximum length of 2039 bit. SDS type 4 also offers an additional SDS transport layer (TL) protocol, which enhances the service provided by the layer 3 SDS protocol to provide protocol mechanisms for end-to-end acknowledgement, store-and-forward and to ensure that applications using this service interpret the user data in the same way.

Figure 7-58 provides SMS parameter overview with trigger points.

7.4.4.1 Parameter overview chart



NOTE – For the SMS service, a paging is preceded within the notification phase.

Figure 7-58 – SMS parameter overview with trigger points

7.4.4.2 {SMS | SDS} service non-accessibility [%]

7.4.4.2.1 Abstract definition

The {SMS | SDS} service non-accessibility is the probability that the end-user cannot access the short message service (SMS) or short data service (SDS) when requested while it is offered by display of the network indicator on the UE.

Figures 7-59 and 7-60 provide SMS transaction flow (originating UE) and SDS signalling flow chart respectively.

7.4.4.2.2 Abstract equation

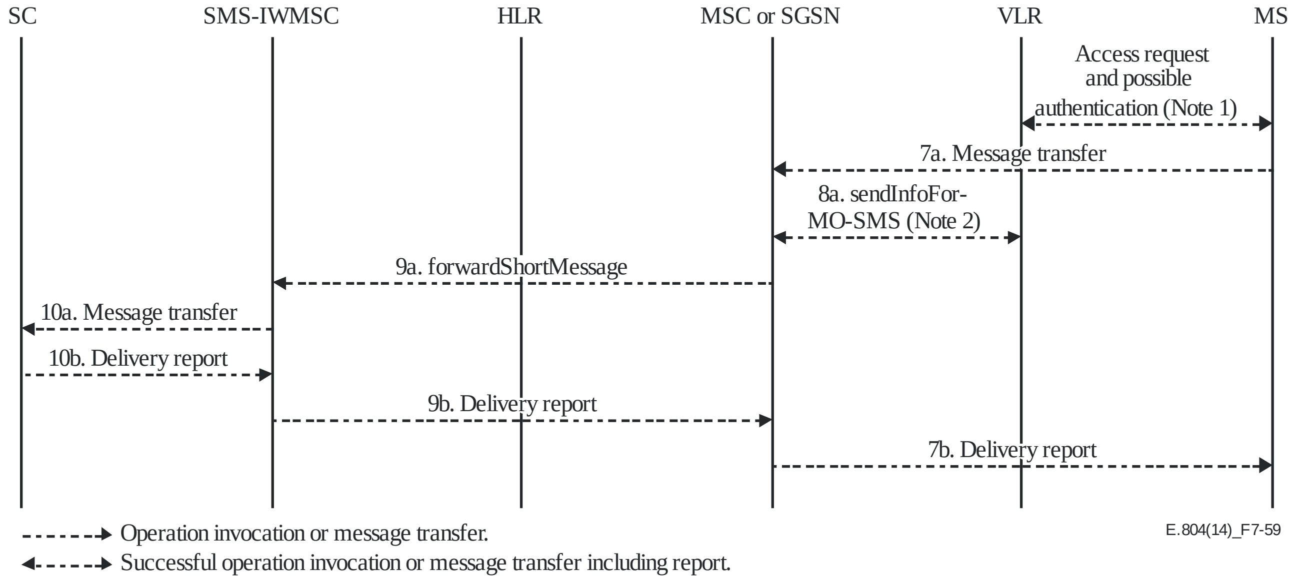
$$\{SMS \mid SDS\}\text{service non-accessibility}[\%] = \frac{\text{unsuccessful }\{SMS \mid SDS\}\text{service attempts}}{\text{all }\{SMS \mid SDS\}\text{service attempts}} \times 100$$

7.4.4.2.3 Trigger points

GSM/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
SMS service attempt	Start: Push the send button (initiate sending an SMS).	Start: Layer 3 (MM): The first "Access request" is sent by the originating UE (Figure 7-59, most upper signalling point). Detailed: CM service request is sent by the originating UE. AT: The "AT+CMGS=<len>" or "AT+CMGS=<MSISDN>" (parameter depends on the "AT+CMGF" setting, PDU or text mode) command is sent by the originating TE.
Successful SMS service attempt	Stop: Receive the acknowledgement from the SMSC at the A-party.	Stop: Layer 3 (SMS): The "Delivery report" is received by the originating UE (Figure 7-59, signalling point number 7b). Detailed: CP_DATA (RP_ACK) is received by the originating UE. AT: "OK" is received by the originating TE.
Unsuccessful SMS service attempt	Stop trigger point not reached.	

Figure 7-59 shows SMS transaction flow – originating UE.



NOTE 1 – Described in [ETSI TS 124 008] and [ETSI TS 129 002].

NOTE 2 – This operation is not used by the SGSN.

Figure 7-59 – SMS transaction flow – Originating UE

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
SDS service attempt	Start: Push the send button (initiate sending an SDS).	Start: Layer 3 (CMCE): The first "U-SDS-DATA" message is sent by the originating UE. AT: The "AT+CMGS="<called party identity>", <length><CR> <LF><user data><CtrlZ>" command is sent by the originating TE, where <called party identity> provides a unique identification of the desired B-party and <length> is the size of the SDS in [bits].
Successful SMS service attempt	Stop: Receive the acknowledgement from the SwMI at the initiating party.	Stop: Layer 2 (LLC): The "BL-ACK" message is received at the originating UE. AT: "OK" is received by the originating TE.
Unsuccessful SMS service attempt	Stop trigger point not reached.	
NOTE – The "BL-ACK" message is related to the logical link control (LLC) protocol whereas the "U-SDS-DATA" message is related to the circuit mode control entity (CMCE) protocol.		

Figure 7-60 provides SDS signalling flow.

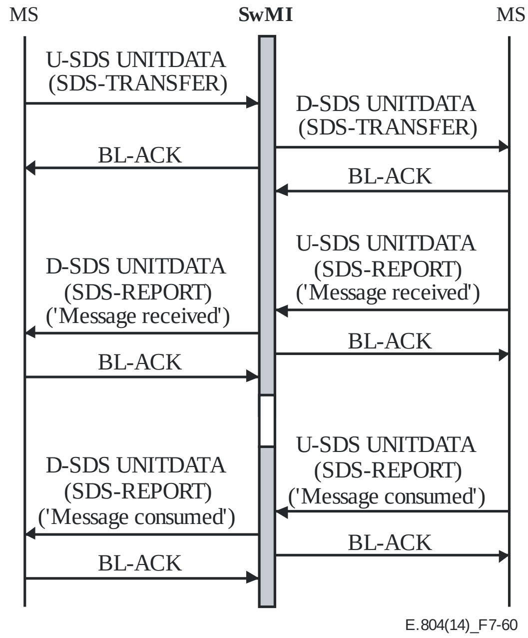


Figure 7-60 – SDS signalling flow chart

Remark:

- In TETRA, the SDS type and parameters relating to the message are set with a previous "AT+CTSDS <AI service>, <area>, <e-to-e encryption>, <access priority>, <called party identity type>" command.

7.4.4.3 {SMS | SDS} access delay [s]

7.4.4.3.1 Abstract definition

The {SMS | SDS} access delay is the time period between sending a short message to the network and receiving a send confirmation from the network at the originating side.

7.4.4.3.2 Abstract equation

$$\{SMS \mid SDS\}\text{access delay }[s] = \left(t_{A,\,receive} - t_{A,\,send}\right)[s]$$

7.4.4.3.3 Trigger points

GSM/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{A,\,send}$	Start: Push the send button (initiate sending an SMS).	Start: Layer 3 (MM): The first "Access request" is sent by the originating UE (Figure 7-59, most upper signalling point). Detailed: CM service request is sent from the originating UE. AT: The "AT+CMGS=<len>" or "AT+CMGS=<MSISDN>" (parameter depends on the "AT+CMGF" setting, PDU or text mode) command is sent by the originating TE.
$t_{A,\,receive}$	Stop:	Stop:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
	Acknowledgement from the SMSC is received at the A-party.	Layer 3 (SMS): The "Delivery report" is received by the originating UE (Figure 7-59, signalling point number 7b). Detailed: CP_DATA (RP_ACK) is received by the originating UE. AT: "OK" is received by the originating TE.

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{A,send}$	Start: Push the send button (initiate sending an SDS).	Start: Layer 3 (CMCE): The first "U-SDS-DATA" message is sent by the originating UE. AT: The "AT+CMGS="<called party identity>", <length><CR> <LF><user data><CtrlZ>" command is sent by the originating TE, where <called party identity> provides a unique identification of the desired B-party and <length> is the size of the SDS in [bits].
$t_{A,receive}$	Stop: Receive the acknowledgement from the SwMI at the initiating party.	Stop: Layer 2 (LLC): The "BL-ACK" message is received at the originating UE. AT: "OK" is received by the originating TE.
NOTE – The "BL-ACK" message is related to the logical link control (LLC) protocol whereas the "U-SDS-DATA" message is related to the circuit mode control entity (CMCE) protocol.		

7.4.4.4 {SMS | SDS} completion failure ratio [%]

7.4.4.4.1 Abstract definition

The {SMS | SDS} completion failure ratio is the ratio of unsuccessfully received and sent messages from one UE to another UE, excluding duplicate received and corrupted messages.

A corrupted SMS (or SDS) is an SMS (or SDS) with at least one bit error in its message part.

Figure 7-61 shows SMS transaction flow (terminating UE).

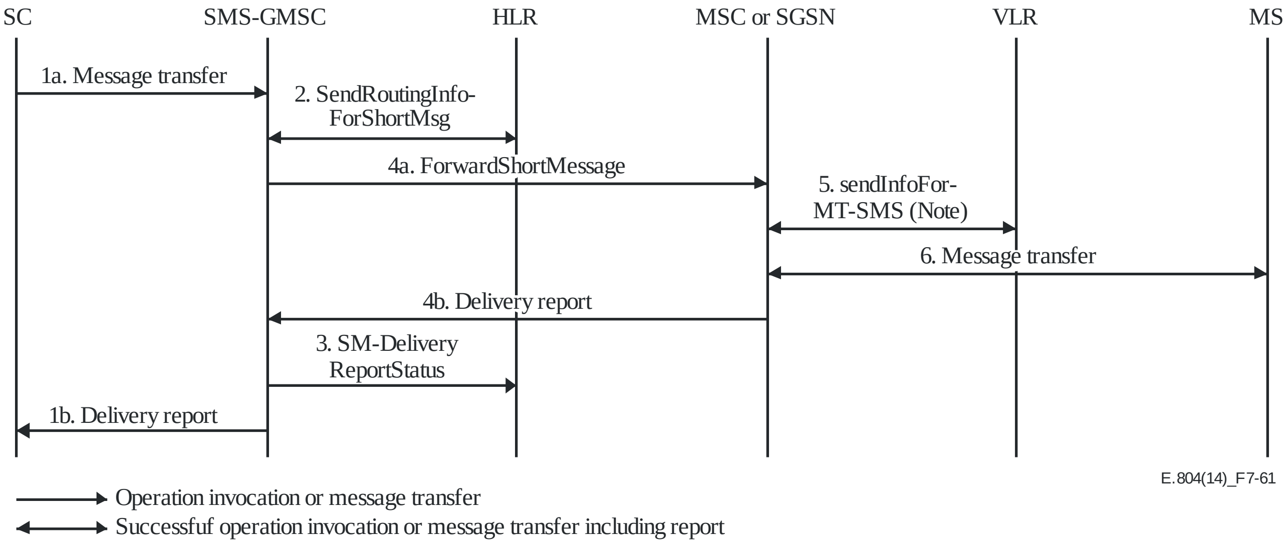
7.4.4.4.2 Abstract equation

$$\{SMS \mid SDS\}\text{completion failure ratio}[\%] = \frac{\text{unsuccessfully received }\{SMS \mid SDS\}}{\text{all }\{SMS \mid SDS\}\text{service attempts}} \times 100$$

7.4.4.4.3 Trigger points

GSM/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
SMS service attempt	Start: Push the send button (Initiate sending an SMS).	Start: Layer 3 (MM): The first "Access request" is sent by the originating UE (Figure 7-59, most upper signalling point). Detailed: CM Service Request is sent from the originating UE. AT: The "AT+CMGS=<len>" or "AT+CMGS=<MSISDN>" (parameter depends on the "AT+CMGF" setting, PDU or text mode) command is sent by the originating TE.
Successfully received SMS	Stop: The short message is received by the B-party's UE.	Stop: Layer 3: The "Message transfer" is received in the terminating UE (Figure 7-61, signalling point number 6). Detailed: CP_DATA (RP_ACK) is sent by the terminating UE. AT: The "+CMTI" event received at the terminating TE.
Unsuccessfully received SMS	Stop trigger point not reached or SMS received is duplicated or corrupted.	



NOTE – This operation is not used by the SGSN.

Figure 7-61 – SMS transaction flow –Terminating UE

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
SDS service attempt	Start: Push the send button (initiate sending an SDS).	Start:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
		Layer 3 (CMCE): The first "U-SDS-DATA" message is sent by the originating UE. AT: The "AT+CMGS="<called party identity>", <length><CR> <LF><user data><CtrlZ>" command is sent by the originating TE, where <called party identity> provides a unique identification of the desired B-party and <length> is the size of the SDS in [bits].
Successfully received test SDS	Stop: The short message is received by the terminating party.	Stop: Layer 3 (CMCE): The corresponding "D-SDS-DATA" message is received by the terminating UE. AT: The "AT+CMTI" new message indication or the "AT+CTSDSR" receive notification for the corresponding message is received at the terminating TE (depending on SDS settings).
Unsuccessfully received test SDS	Stop trigger point not reached or SMS received is duplicated or corrupted.	

Remarks:

- In GSM/UMTS, "CMGR=<n>" gives back the received SMS, or "CMGL="ALL" or "CMGL=4" all of the received ones. In order to receive CMTI events they have to be activated at the B-party with the "AT+CNMI" command.
- The detection of duplicated and corrupted received SMS or SDS is a post processing issue.
- In TETRA, the short data service centre (SDSC) might modify the content of an SDS. The unique identification of an SDS at the receiving UE is up to the following analysis.

7.4.4.5 {SMS | SDS} end-to-end delivery time [s]

7.4.4.5.1 Abstract definition

The {SMS | SDS} end-to-end delivery time is the time period between sending a short message to the network and receiving the very same short message at another UE.

7.4.4.5.2 Abstract equation

$$\{SMS \mid SDS\}\text{end-to-end delivery time}\,[s] = \left(t_{B,receive} - t_{A,send}\right)[s]$$

7.4.4.5.3 Trigger points

GSM/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{A,send}$	Start: Push the send button (Initiate sending an SMS).	Start: Layer 3: The first "Access request" is sent by the originating UE (Figure 7-59, most upper signalling point).

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
		Detailed: CM service request is sent by the originating UE. AT: The "AT+CMGS=<len>" or "AT+CMGS=<MSISDN>" (parameter depends on the "AT+CMGF" setting, PDU or text mode) command is sent by the originating TE.
$t_{B,receive}$	Stop: The short message is received by the B-party's UE.	Stop: Layer 3: The "Message transfer" is received by the terminating UE (Figure 7-61, signalling point number 6). Detailed: CP_DATA (RP_ACK) is sent by the terminating UE. AT: The "CMTI" event received at the terminating TE.

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{A,send}$	Start: Push the send button (initiate sending an SDS).	Start: Layer 3 (CMCE): The first "U-SDS-DATA" message is sent by the originating UE. AT: The "AT+CMGS="<called party identity>", <length><CR> <LF><user data><CtrlZ>" command is sent by the originating TE, where <called party identity> provides a unique identification of the desired B-party and <length> is the size of the SDS in [bits].
$t_{B,receive}$	Stop: The short message is received by the terminating party.	Stop: Layer 3 (CMCE): The corresponding "D-SDS-DATA" message is received by the terminating UE. AT: The "AT+CMTI" new message indication or the "AT+CTSDSR" receive notification for the corresponding message is received at the terminating TE (depending on SDS settings).

7.4.4.6 {SMS | SDS} receive confirmation failure ratio [%]

7.4.4.6.1 Abstract definition

The {SMS | SDS} receive confirmation failure ratio is the probability that the receive confirmation for a sent attempt is not received by the originating UE although requested.

7.4.4.6.2 Abstract equation

$$\{SMS \mid SDS\}\,\text{receive confirmation failure Ratio}\,[\%] = \frac{\text{non-confirmed}\,\{SMS \mid SDS\}\,\text{receptions}}{\text{all}\,\{SMS \mid SDS\}\,\text{service attempts}} \times 100$$

7.4.4.6.3 Trigger points

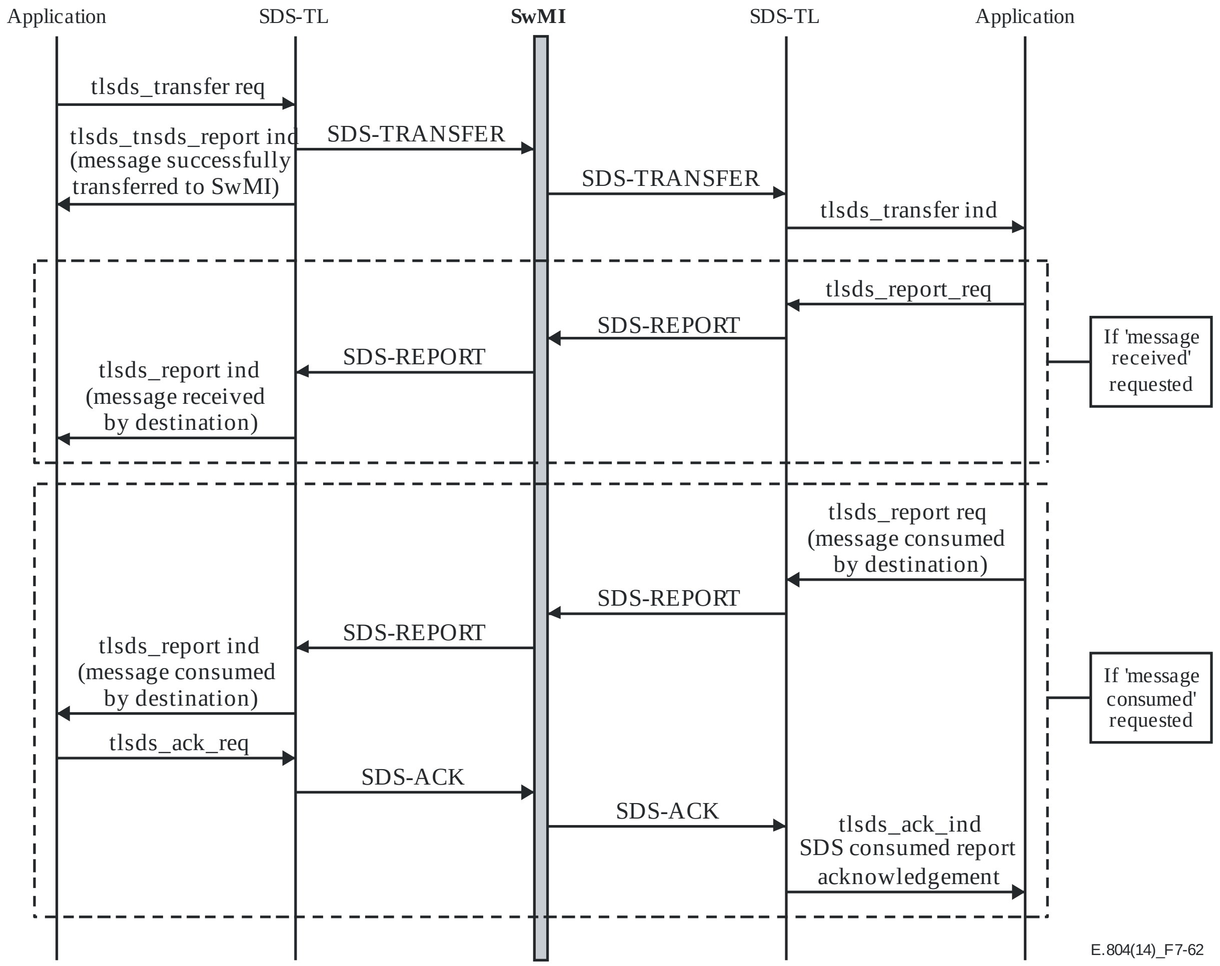
GSM/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
SMS service attempt	Start: Push the send button (initiate sending an SMS).	Start: Layer 3: The first "Access request" is sent by the originating UE (Figure 7-59, most upper signalling point). Detailed: CM service request is sent by the originating UE. AT: The "AT+CMGS=<len>" or "AT+CMGS=<MSISDN>" (parameter depends on the "AT+CMGF" setting, PDU or text mode) command is sent by the originating TE.
Confirmed SMS reception	Stop: Receive the confirmation at the initiating party that the message is received at the terminating party.	Stop: To be defined.
Non-confirmed SMS reception	Stop trigger point not reached.	

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
SDS service attempt	Start: Push the send button (initiate sending an SDS).	Start: Layer 3 (CMCE): The first "U-SDS-DATA" message carrying the "SDS-TRANSFER" message with Delivery report request "Message received report requested" or "Message received and consumed report requested" is sent by the originating UE. AT: The "AT+CMGS="<called party identity>", <length><CR> <LF><user data><CtrlZ>" command is sent by the originating TE.
Confirmed SMS reception	Stop: Receive the confirmation at the initiating party that the message is received at the terminating party.	Stop: Layer 3 (CMCE): The "D-SDS-DATA" message carrying the "SDS-REPORT" message with delivery status "SDS receipt acknowledged by destination" is received by the originating UE. AT: to be defined.
Non-conformed SMS reception	Stop trigger point not reached.	

Figure 7-62 shows SDS signalling flow chart for SDS type 4.



NOTE 1 – SDS-TL protocol according to [ETSI EN 300 392-2], clause 29.3.3 with transparent SwMI transport is used

NOTE 2 – SwMI store-and-forward functionality uses additional SDS-REPORT and SDS-ACK messages

Figure 7-62 – Signalling flow chart for SDS type 4

7.4.4.7 {SMS | SDS} receive confirmation time [s]

7.4.4.7.1 Abstract definition

The {SMS | SDS} receive confirmation time is the time period between sending a short message to the network and receiving the receive confirmation for this message from the network.

7.4.4.7.2 Abstract equation

$$\left\{\mathrm{SMS}\,\middle|\,\mathrm{SDS}\right\}\text{receive confirmation time}\,[\mathrm{s}] = \left(t_{\mathrm{A,receive\ confirmation}} - t_{\mathrm{A,send}}\right)\ [\mathrm{s}]$$

7.4.4.7.3 Trigger points

GSM/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{A,send}$	Start: Push the send button (initiate sending an SMS).	Start: Layer 3: The first "Access request" is sent by the originating UE (Figure 7-59, most upper signalling point). Detailed: CM service request is sent by the originating UE. AT: The "AT+CMGS=<len>" or "AT+CMGS=<MSISDN>" (parameter depends on the "AT+CMGF" setting, PDU or text mode) command is sent by the originating TE.
$t_{A,receive\ confirmation}$	Stop: Receive the confirmation at the initiating party that the message is received at the terminating party.	Stop: To be defined.

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{A,send}$	Start: Push the send button (initiate sending an SDS).	Start: Layer 3 (CMCE): The first "U-SDS-DATA" message carrying the "SDS-TRANSFER" message with delivery report request "Message received report requested" or "Message received and consumed report requested" is sent by the originating UE. AT: The "AT+CMGS="<called party identity>", <length><CR> <LF><user data><CtrlZ>" command is sent by the originating TE.
$t_{A,receive\ confirmation}$	Stop: Receive the confirmation at the initiating party that the message is received at the terminating party.	Stop: Layer 3 (CMCE): The "D-SDS-DATA" message carrying the "SDS-REPORT" message with delivery status "SDS receipt acknowledged by destination" is received by the originating UE. AT: to be defined.

7.4.4.8 {SMS | SDS} consumed confirmation failure ratio [%]

7.4.4.8.1 Abstract definition

The {SMS | SDS} consumed confirmation failure ratio is the probability that the consumed confirmation for a sent attempt is not received by the originating UE although requested.

7.4.4.8.2 Abstract equation

$$\{SMS \mid SDS\}\,consumed\,confirmation\,failure\,ratio[\%] = \frac{non\text{-}confirmed\{SMS \mid SDS\}\,consumptions}{all\,\{SMS \mid SDS\}\,service\,attempts} \times 100$$

7.4.4.8.3 Trigger points

GSM/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
SMS service attempt	Start: Push the send button (initiate sending an SMS).	Start: Layer 3: The first "Access request" is sent by the originating UE (Figure 7-59, most upper signalling point). Detailed: CM service request is sent by the originating UE. AT: The "AT+CMGS=<len>" or "AT+CMGS=<MSISDN>" (parameter depends on the "AT+CMGF" setting, PDU or text mode) command is sent by the originating TE.
Confirmed SMS consumption	Stop: Receive the confirmation at the initiating party that the message is received at the terminating party.	Stop: To be defined.
Non-confirmed SMS consumption	Stop trigger point not reached.	

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
SMS service attempt	Start: Push the send button (initiate sending an SDS).	Start: Layer 3 (CMCE): The first "U-SDS-DATA" message carrying the "SDS-TRANSFER" message with delivery report request "Message consumed report requested" or "Message received and consumed report requested" is sent by the originating UE. AT: The "AT+CMGS="<called party identity>", <length><CR> <LF><user data><CtrlZ>" command is sent by the originating TE.

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
Confirmed SMS consumption	Stop: Receive the confirmation at the initiating party that the message is consumed.	Stop: Layer 3 (CMCE): The "D-SDS-DATA" message carrying the "SDS-REPORT" with delivery status "SDS consumed by destination" is received by the originating UE. AT: To be defined.
Non-confirmed SMS consumption	Stop trigger point not reached.	

7.4.4.9 {SMS | SDS} consumed confirmation time [s]

7.4.4.9.1 Abstract definition

The {SMS | SDS} consumed confirmation time is the time period between sending a short message to the network and receiving the consumed confirmation from the network.

7.4.4.9.2 Abstract equation

$$\{\mathrm{SMS} | \mathrm{SDS}\}\ \text{consumed confirmation time}\,[s] = \left(t_{A,\mathrm{consumedconfirmation}} - t_{A,\mathrm{send}}\right)[s]$$

7.4.4.9.3 Trigger points

GSM/UMTS:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{A,send}$	Start: Push the send button (initiate sending an SMS).	Start: Layer 3: The first "Access request" is sent by the originating UE (Figure 7-59, most upper signalling point). Detailed: CM service request is sent by the originating UE. AT: The "AT+CMGS=<len>" or "AT+CMGS=<MSISDN>" (parameter depends on the "AT+CMGF" setting, PDU or text mode) command is sent by the originating TE.
$t_{A,consumed\ confirmation}$	Stop: Receive the confirmation at the initiating party that the message is consumed at the terminating party.	Stop: To be defined.

TETRA:

Event from abstract equation	Trigger point from user's point of view	Technical description/ protocol part
$t_{A,send}$	Start: Push the send button (initiate sending an SDS).	Start: Layer 3 (CMCE): The first "U-SDS-DATA" message carrying the "SDS-TRANSFER" message with delivery report request "Message consumed report requested" or "Message received and consumed report requested" is sent by the originating UE. AT: The "AT+CMGS="<called party identity>", <length><CR> <LF><user data><CtrlZ>" command is sent by the originating TE.
$t_{A,consumed\ confirmation}$	Stop: Receive the confirmation at the initiating party that the message is consumed at the terminating party.	Stop: Layer 3 (CMCE): The "D-SDS-DATA" message carrying the "SDS-REPORT" with delivery status "SDS consumed by destination" is received by the originating UE. AT: To be defined.

8 Typical procedures for quality of service measurement equipment

8.1 Aim of measurement

The aim of the measurements described in this Recommendation is to assess the network under test for its quality of service (QoS) parameters as defined in clause 7. This is, to determine the network quality for the respective transactions from the user's view.

8.2 Classification of services

8.2.1 Classification guidelines

For the purpose of this Recommendation, services are classified using what is considered to be their dominating property. The first distinction is made between direct and store-and-forward services:

- Direct-transaction services are services where there is – in the user's perception – a direct end-to-end connection.
- Store-and-forward services are services where content is stored in the network and delivered to the recipient at a later point in time.

As a technically usable differentiation, a service is considered to be direct if it is possible to decide on end-to-end content transfer success from the initiating party (A-party) of the connection within the scope of the transaction itself.

NOTE – e-mail is a special case since it has both aspects of direct and of store-and-forward services. More information on store-and-forward services measurement is provided in clause 8.5.

8.2.2 General structure of service descriptions

In the following, each service family description will contain the following structural elements:

- A general part defining:
  - the basic transaction definition and if applicable, transaction types;
  - a description of the transaction phase combined with a table of parameters governing transaction behaviour in this phase;
  - a description of all possible outcomes of a single transaction;
  - a description of content quality measurement definitions (if applicable).
- In case there are service-dependent differences, a service-dependent part having the same structure as above.

8.3 General aspects for all types of services

8.3.1 Set-up and control

Measurements should be conducted in a way that user behaviour is realistically modelled. Parameters and settings which have substantial influence on results need to be under control of the measurement equipment.

The test case design (configuration and user profile) – to the degree necessary to fully reproduce the test – shall be part of the measurement documentation.

It is assumed that for all types of services under test, a test case consists of a number of single identical transactions. The measurement equipment and control must ensure that the starting conditions are the same for each transaction. This includes, among other things, that pause times are sufficiently long that the equipment is in a stable (idle) state again. The parameter "guard time" sets a minimum value for the pause between transactions.

It is assumed that all QoS-relevant transaction parameters are recorded for proper post-processing and are kept constant during measurements. If a measurement contains more than one parameter set, evaluation shall be made for each parameter set separately.

8.3.2 Phase and result classification

In order to ensure common wording, the following clause defines terms and definitions for service measurements.

It is assumed that each transaction can be described at least by one seamless sequence of phases. There may exist several angles of view (AOV), each leading to a different phase description.

Example: Internet services (as described by its QoS parameters defined in clause 7.1.2 and method A and method B). AOV differ here by different assumptions on start of service usage. Each AOV, however, is a consistent description by seamlessly connected phases.

Phases may be further described having sub-phases.

Pauses between transactions are not explicitly mentioned, but are relevant with respect to parameter reporting. Typically, there is a minimum pause (guard time) ensuring that the system under test is in a stable starting condition for the next test. Values are technology-dependent.

8.3.2.1 Phase and result classification for direct services

A direct transaction consists of two top-level phases: service access and service usage.

Table 8-1 shows general phase definitions for direct services while Table 8-2 shows the general result classification for direct services.

Table 8-1 – General phase definitions for direct services

Phase	Sub-phase	Definition
Service access		All steps leading to the technical ability to perform actual user-perspective content transport between A- and B-party. Service access may consist of different sub-phases, e.g., Network access, IP service access and Internet access. The availability of these sub-phases actually depends on the particular service.
	Network access	Basic access to the network under test. Successful network access is assumed when the UE is able to do as much basic communication with the network as is necessary to initiate the next phase in the service access procedure.
	IP service access	Basic access to the generic packet-data transfer capabilities the particular service is based upon.
	Internet access	Basic access to those Internet services the service is meant to provide.
Service usage		Content transfer between the A- party and the B-party.

A direct transaction may have one of the following overall results.

Table 8-2 – General result classification for direct services

Result	Definition
Failed	Phase of service usage not reached. Successful or failed service access may be broken down into diagnostic sub-categories. The general name-forming rule is: <name of sub-phase>result. Example: Network access failed
Completed	Data-transfer transactions: All content intended to be transferred has been successfully transferred. Conversational transactions: The intended transaction duration has been reached.
Dropped/Cut-off	Service usage was ended before completion.
NOTE – If a transaction, while being in the service usage phase, is stopped due to some timeout or due to other criteria by the measurement system, e.g., to enhance test rate, this shall be treated as a dropped transaction. This behaviour has to be recorded by the measurement system.	

8.3.2.2 Phase and result classification for store-and-forward services

A store-and-forward transaction consists of two top-level phases: content sending and content delivery.

Table 8-3 provides general phase definitions for store-and-forward services and Table 8-4 provides the general result classification for store-and-forward services.

Table 8-3 – General phase definitions for store-and-forward services

Phase	Sub-phase	Definition
Content sending		All steps required to transfer the content to the network, up to the point where the network is able to start delivery. This phase is completed when there is nothing more the A-party can/needs to do to transfer the content to the B-party. It is assumed that the A-party gets information sufficient to judge if sending has been successful or not.
Content delivery		All steps to transfer the content from the network to the B-party. Delivery may consist of two main sub-phases: **notification** and **retrieval**.
	Notification (optional)	Information sent to the B-party that content is ready for transfer.
	Retrieval/Delivery	Transport of content from the network to the B-party.

A store-and-forward transaction may have one of the following overall results.

Table 8-4 – General result classification for store-and-forward services

Result	Sub-category	Definition
Completed		Content was successfully transferred from the A- party to the B-party.
Failed		Content was not successfully transferred from the A-party to the B-party. Depending on the particular services and the available information, there may be a number of possible sub-categories for this result.
	Undelivered	Content was successfully sent to the network, but was never delivered or retrieved.
	Send failed	Content was not successfully delivered to the network.
	Lost	Content was successfully sent to the network, but notification was never received by the B-party. Diagnostic sub-category in case notification can/shall be technically identified within the delivery process.
	Timeout	The transaction was completed, but the content delivery time was above a given threshold.

8.4 Telephony measurements

This clause deals with telephony services. In general, the term "content" will be used throughout this clause for the information flow exchanged between participating users during a call. Depending on the type of service, content can be audio or audio and video.

8.4.1 General aspects

8.4.1.1 Transaction definition and transaction types

The basic transaction for telephony testing is equivalent to a single call to a counterpart extension.

Table 8-5 provides an overview of parameters for telephony measurements.

8.4.1.2 Parameter overview

Table 8-5 – Parameter overview for telephony measurements

Phase	Parameters
Service access	Call counterpart. This includes the type of equipment (dedicated unit, unique identifier (e.g., called party number), automatic answer with taped message, etc.).
	Call type.
	Timeout value
Service usage	Call duration.
	Content flow direction: This is an inner parameter for a transaction. Basically, all combinations of uplink/downlink dynamics are possible: – Uplink only; – Downlink only; – Conversational (alternating uplink and downlink). This is the recommended standard testing mode. Other testing modes are considered to be used only for special purposes; – "Duplex" (uplink and downlink flow simultaneously).
	Codec settings.
	Algorithm and scale used for speech quality evaluation.
Call clear-down	Guard time.
Pause	Pause duration. For current UEs, a pause time of at least 15 seconds (guard time) is recommended. However, this duration may be adjusted to local conditions or special testing goals, but this must be reported. If the pause duration is too short, side effects may occur, resulting in all kinds of transient effects and distortions in measurement data. It should be made certain that all the QoS parameters to be measured are not affected by the pause time.

The last transaction within a measurement sequence does not require a pause.

8.4.1.3 Additional transaction result definitions

For call set-up assessment beyond QoS data acquisition, typically a state model driven by suitable trigger-events information combined with information from the call control engine is being used. This state model may also be used to determine timing information for each phase.

Service usability, i.e., presence of a usable two-way connection, shall be verified by a procedure based on content test transmissions within a given time window. If within this time window no connection can be verified, the set-up attempt shall be considered as failed and the call attempt be terminated.

A call is active only as long as both sides consider it to be active. A call is therefore considered to be dropped if either side detects a dropped call.

Above definitions lead to the following decision tree for the outcome of a call (Figure 8-1 includes the end-of-call cases).

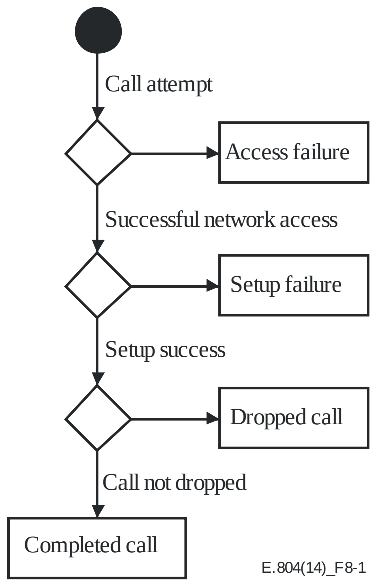


Figure 8-1 – Telephony measurement outcome

8.4.1.4 Content quality

For content quality assessment, data is generated at the receiving end. For downlink content, data storage is therefore straightforward; quality-assessment data is simply included with other data items making up the result file. For uplink content, at some point in time results from the involved B-parties have to be combined with the data from the originating A-party.

For assessing content quality of complete transmitted speech samples, at least the following methods are possible:

- Real time assessment (streaming mode), where the quality assessment algorithm continuously outputs the defined quality measures.
- "Offline" assessment, where content is first recorded in some way and later being processed.

Data processing must make sure that only such content quality data is used which lies inside the "connection active" time window and is in line with one of the defined content quality parameters.

8.4.1.5 Verification of usable two-way connection

Only calls with a valid two-way end-to-end information connection shall be considered for content quality assessment (valid calls).

A non-valid call shall be treated like a dropped call, with a modifier indicating this particular cause.

8.4.2 Speech telephony

8.4.2.1 Transaction definition and transaction types for speech telephony

See clause 8.4.1.

8.4.2.2 Parameter overview for speech telephony

See clause 8.4.1.

8.4.2.3 Additional transaction results for speech telephony

See clause 8.4.1.

8.4.2.4 Content quality for speech telephony

See clause 8.4.1.

8.4.2.5 Verification of usable two-way connection for speech telephony

This shall be verified by a procedure based on audio test transmissions within a given time window. If within this time window no audio connection can be verified, the set-up attempt shall be considered to be failed and the call attempt be terminated.

NOTE – To make sure an audio connection is valid, it is assumed that an appropriate kind of data analysis on audio flow is performed (see [ITU-T P.56]).

8.4.3 Video telephony

8.4.3.1 Transaction definition and transaction types for video telephony

The basic transaction for video telephony testing is equivalent to a single call to a counterpart extension.

Due to existing usage and hardware, typical video calls will be between UEs, so no further call types are distinguished.

Unlike other services, it is currently assumed that video-call testing will require a high degree of abstraction in the sense that testing-system architectures may differ quite thoroughly from those found in the "real connections". Presently, there are no known terminal-based protocol stacks delivering data necessary for testing or QoS assessment. Therefore, it is assumed that PC-based implementations of video protocol stacks have to be used, where the UE serves only as the modem part of the connection.

It is further assumed that typical tests will use data which serve as load and carry diagnostic data at the same time.

8.4.3.2 Parameter overview for video telephony

Table 8-6 provides parameter overview for video telephony measurements.

NOTE – Content flow will typically be governed by the video protocol and is assumed to involve simultaneous data flow in both directions.

Table 8-6 – Parameter overview for video telephony measurements

Phase	Parameters
Service access	Connection type (CSD bearer type).
	Video-telephony protocol used.
	Call counterpart technology and architecture. This includes the type of test being made (e.g., "transparent" test of modem connection).
	Timeout value.
Service usage	Call duration.
Call clear-down	Guard time.
Pause	Pause duration.

The last transaction within a measurement sequence does not require a pause.

8.4.3.3 Additional transaction result definitions for video telephony

It is assumed that the A and B party video-telephony protocol stacks are compatible, and given a usable end-to-end connection of proper type, a video-telephony connection can be established. It is assumed that if the network cannot provide a connection with the required bearer properties, this will either result in failure to provide a usable end-to-end connection or the video-telephony protocol stack will detect this by indicating connection failure.

It is conceivable that the network provides the wrong bearer type. Due to limited "decision power" determining the usability of an end-to-end connection, this could go undetected at first, but leads to quality deficiencies due to inappropriate bearer properties.

8.4.3.4 Content quality for video telephony

Currently no stable specification for video quality exists. It is assumed that the audio part of the connection will be tested essentially the same way as for speech telephony.

8.4.3.5 Verification of usable two-way connection for video telephony

It is assumed that the video telephony protocol being used implicitly checks for useful connection.

8.4.3.6 Call set-up considerations for MTSI calls

When MTSI calls are set up, any network address translation (NAT) devices present in the call path need to be opened. This can be done automatically by the network for managed NATs, or handled by the MTSI clients for unmanaged NATs. The call set-up, as seen by the calling party, is assumed to be done when the first valid media packet is received from the called party. This should be interpreted as the first packet containing the negotiated media payload type.

The RTP standard has a number of fixed media payload types, but MTSI uses dynamic payload type assignments, which means that the media payload types are allocated from the range 96 to 127. The relation between the payload type and the actual codec and codec format used is then defined in the SDP during the call set-up negotiation.

Thus the client needs to use the information in the SDP and the outcome of the negotiation, to identify which payload type is finally agreed, and use the first reception of that payload type as the end of the call set-up phase.

8.4.4 Group call

8.4.4.1 Transaction definition and transaction types for group calls

The basic transaction for group call testing is an extension to an individual telephony speech call with two parties involved (A-party and B-party). One Test UE (A-party) will call n counterparts (B_i-parties) having $1 \leq i \leq n$. The n counterparts are typically Test UEs.

Type is group call (GC) and it will not be distinguished between mobile originating (MO) and mobile termination (MT).

In principle, two options for a typical measurement scenario are available. Either all group members may be connected to the same common measurement system at one location (not limiting that the measurement system might be in mobility) or each group member is connected to a separate measurement system and these might be distributed at various locations (again not limiting that all or some of the measurement systems might be in mobility).

8.4.4.2 Parameter overview for group calls

Table 8-7 provides a parameter overview for group call measurements.

Table 8-7 – Parameter overview for group call measurements

Phase	Parameters
Service access	Call counterparts (group to call).
	Call type is group call (GC).
	Timeout for failed-call condition.
Service usage	Call duration.
	Content flow direction: Typically, only one party can talk at one point in time. A pattern may be specified, which describes, at what point in time a certain party has to talk to the others.
Call clear-down	Guard time.
Pause	Pause duration. Depending on the used radio access technology, the duration of required pauses may differ significantly. However, this duration may be adjusted to local conditions or special testing goals, but this must be reported. If the pause duration is too short, side effects may occur, resulting in all kinds of transient effects and distortions in measurement data. It should be made certain that all the QoS parameters to be measured are not affected by the pause time.

The last transaction within a test sequence does not require a pause.

8.4.4.3 Additional transaction result definitions for group calls

It is assumed that all members of the target group are known by the measurement system(s). All group members are controlled and called exclusively by the measurement system(s).

For quality measurements, the following rules apply: As a first measurement option, all parties may talk the same amount of time. The sequence of the talking party shall be A-Party, B_1-Party, B_2-Party,…., B_n-Party. A group call always ends after the B_n-Party has finished its last talking period. In this case, N × (N+1) quality values are measured per group call. A quality value per group call should apply statistical means (e.g., a simple approach would be the arithmetic average) with respect to aggregation and is for further study.

As a second measurement option, only the initiating A-Party will generate a talk burst and the group call is ended after the speech phase of the A-Party. In this case, N quality values are measured per group call.

In both cases, the roles of the participating group members might change, e.g., a former B-Party can cyclically take the role of the initiating A-party whereas the former A-party gets the role of a B-Party.

If a B_i-Party is disconnected during the group call, the quality values of this B_i-Party will be excluded from the overall quality value.

8.4.4.4 Content quality for group calls

See clause 8.4.1.

8.4.4.5 Verification of usable connection for group calls

The usability of the connection will be tested implicitly by the content quality algorithm.

8.5 Store-and-forward services measurement

8.5.1 General aspects

The basic transaction for store-and-forward services testing is equivalent to transmission of a single unit of content (which may, however, contain several parts such as audio, video and text) between two parties.

Typically, content is, in the end-to-end perspective, being sent either from some fixed-network source to a UE (mobile termination, MT) or from one UE to another (mobile originated to mobile, MOM). However, MOM testing with a two-UE set-up may cause additional uncertainty due to coverage and signalling effects in the destination UE.

Basically, there are two recommended methods of testing:

a) using a destination UE in a fixed location with virtually 100 per cent transfer probability. Part of the measurement procedure should be cyclical reference checks to assure reception quality respectively to create baseline data; or

b) using a fixed-network destination such as a large account server which is accessed by suitable means from a fixed-network location.

In all cases, testing methodology requires that measurement data are being collected in at least two locations and need to be integrated before the final QoS evaluation can be made.

For practical reasons, measurements are typically limited to a certain time window. Technically it is possible that content being sent in previous test session arrives at the system after the session has been finalized. Technical provisions are needed to ensure proper handling of such content, either by ignoring it or by correctly assigning it in post-processing.

Content used by the measurement system shall carry appropriate signature to identify it as belonging to the measurement, to identify the test session within it was sent, and to enable unambiguous correlation between sent and received content.

8.5.1.1 Transaction phase and parameter overview

Table 8-8 provides transaction phase and parameter overview for store-and-forward services.

Table 8-8 – Transaction phase and parameter overview for store-and-forward services

Phase	Parameters
Sending	Destination equipment type.
Notification	Timeout for notification.
Retrieval	
Pause	Pause duration.

8.5.1.2 Additional transaction result definitions

The nature of store-and-forward services implies that loss rates need precise definition. From the user's point of view, there is expectation that content is delivered within the maximum expected delivery time (MEDT). Even if technically an outstanding content item may be delivered later, it should be considered lost anyway. This leads, in turn, to the problem of over-aged content items which may disturb subsequent measurements. Suitable means of eliminating such effects should be taken, e.g., if the service allows for it, setting a reasonable lifetime for the content item, and using unique sequence numbering in a way to prevent aforementioned effects.

Post-processing must ensure that, if the measurement system is shut down before a given maximum waiting time has expired, all pending units of content, i.e., those with an age less than MEDT are completely removed from QoS evaluation since there is no way to predict if these elements will arrive within MEDT or not.

The influence of over-aged content should be reduced to a minimum by using a selective retrieval. Only the content delivered within the MEDT should be downloaded when possible.

8.5.1.3 Content quality for store-and-forward services

Definitions of these service types are given in the respective service-related clauses.

8.5.2 SMS measurements

8.5.2.1 General aspects of SMS measurements

There are two modes for SMS transmission:

- text mode;
- PDU mode.

Basically, these modes should be equivalent including options available. In practice, mode support is mobile-dependent.

For each SMS, a "lifetime" can be set after which a SMS is deleted in the short message service centre (SMSC). While in PDU mode, this parameter is part of the parameter structure, in text mode it is set in the mobile device, which may not be supported by all mobile device types. This leads to the recommendation that PDU mode should be used.

8.5.2.2 Transaction definitions and transaction types for SMS

The basic transaction for SMS testing is equivalent to transmission of a single SMS.

Typically, user SMS are being sent either from some fixed-network source to a mobile (SMS mobile termination, SMS-MT), or from one mobile to another (SMS mobile originating, SMS-MO, with regard to mobile-based testing). However, uplink SMS testing with a two-mobile set-up may cause additional uncertainty due to coverage and signalling effects at the destination UE. Basically, there are two recommended methods of testing:

- using a destination mobile in a fixed location with virtually 100 per cent transfer probability. Part of the measurement procedure should be cyclical reference checks to assure reception quality respectively to create baseline data; or
- using a fixed-network destination such as a large account SMS server which is accessed by suitable means from a fixed-network location.

In all cases, testing methodology requires that measurement data be collected in at least two locations and be integrated before the final QoS evaluation can be made.

To ensure comparability and statistical validity of transactions, the following outer conditions need to be constant throughout a test case:

- SMS mode.
- Call counterpart.
- SMSC being used.
- Call timing including behaviour in case of delivery failure to the network.

For reasons of comparability in density, it is required that all SMS sending attempts which form part of the statistics must follow a constant time pattern. Additional SMS sending attempts, e.g., in the case of failure of delivery to the network must not affect the pattern of the statistically relevant call attempts.

SMS being sent from either side should contain information enabling validity and integrity checking during evaluation. The recommended minimum set includes:

- signature assuring that only test system SMS are being considered;
- sequence number for SMS delivery timing.

If required, a test SMS may contain additional data (padding) to obtain SMS with a length assumed to be typical for a particular network.

If required, more powerful means of ensuring that only SMS generated by the test system can be added. For example, the SMS may contain a code which is created from checksum content and a seed value delivered by the receiving side. This ensures that even if a static signature is duplicated by operational errors or malfunction of other test systems, the system will be robust against it.

Furthermore, for the general design of SMS testing, communication between the mobile testing system and its stationary counterpart is required to ensure certain starting conditions:

- Prior to actual testing, the SMS storage of the UE should be cleared to exclude SMS delivery failure due to memory shortage.
- The stationary side should start sending SMS only after a "go" from the mobile side, ensuring that no transient effects occur.

8.5.2.3 Testing mode for SMS-MT

For this type of testing, SMS are being generated at a constant rate by the stationary side.

All SMS received which have not been generated by the stationary side shall be ignored for quality assessment.

For reception behaviour, two modes exist; it is the mobile-dependent modes which are supported.

a) On SMS reception, the mobile device generates an indication message (CMTI message) on its data connection to the personal computer (PC). The actual SMS can then be requested by a command.

b) The mobile device outputs the SMS directly on its data connection to the PC.

From the control and trigger point accuracy considerations, mode a) is preferred.

8.5.2.4 Testing mode for SMS-MO

For this type of testing, SMS are being generated at a constant rate by the mobile side, plus eventual extra SMS in case of failure of delivery to the network.

8.5.2.5 Transaction phase and parameter overview for SMS

Table 8-9 provides transaction phase and parameter overview for SMS.

Table 8-9 – Transaction phase and parameter overview for SMS

Phase	Parameters
Sending	Type and destination equipment type (depends on type, e.g., to mobile device, to large account SMS server, etc.)
	SMSC used
	Content size
Notification	Timeout for notification
	Notification mode for the incoming SMS
Retrieval	
Pause	Pause duration. For current GSM equipment, a pause time of at least 15 s (guard time) is recommended

8.5.2.6 Possible transaction results for SMS

See clause 8.5.1.

8.5.2.7 Content quality for SMS

It is assumed that the SMS service protect its payload sufficiently so it can be assumed that no distortions in content will occur.

8.5.3 MMS

8.5.3.1 General aspects of MMS measurements

With respect to general aspects of store-and-forward systems in MMS testing special care has to be taken for proper handling of the incoming notification SMS not belonging to the active session. They shall be treated as follows:

- Incoming SMS not being a valid push SMS shall be ignored.
- Incoming valid push SMS identified as being sent by the testing system but not belonging to the current test session may be ignored (see general remarks).
- For all other push SMS, it is optional to download the MMS and to check if it belongs to the measurement.

NOTE – If the network under test modifies the push SMS content in a way that it cannot be detected if the push SMS belongs to the measurement, ignoring the push SMS has the risk that a successful MMS from the user's perspective is reported as unsuccessful by the measurement system.

8.5.3.2 Transaction definitions and transaction types for MMS

The basic transaction for MMS testing is equivalent to transmission of a single MMS, which may however contain several parts such as picture or video, audio and text.

Typically, user MMS are being sent either from a mobile device to another mobile device (MOM or MTM depending on measurement set-up), or from a mobile device to a fixed-network location (MOF). In the case of MMS the destination is typically one or more e-mail addresses.

In case a destination mobile device is classified by the network as not capable of MMS reception, a "legacy SMS" is transmitted instead. This SMS will contain information on how MMS content can be downloaded via Internet. The format of such legacy SMS is not standardized.

In mobile-to-mobile scenarios there is always the risk of ambiguity with respect to end-to-end quality assessment. In the case of store-and-forward measurements, this is less critical due to the definition of QoS parameters which allows distinguishing the uplink and the downlink components. Nevertheless, QoS parameters expression end-to-end performance will be affected. Therefore, two methods of testing are recommended:

a) using a B-party mobile in a fixed location with very good network coverage. Part of the measurement procedure should be appropriate cyclical reference checks to obtain baseline data; or

b) using an e-mail destination.

To ensure comparability and statistical validity of transactions, the following outer conditions need to be constant throughout a test case:

- MMS content;
- call counterpart;
- MMS proxy being used;
- send timing including behaviour in case of delivery failure to the network.

For reasons of comparability in density, it is required that all MMS sending attempts which form part of the statistics must follow a constant time pattern. Additional MMS sending attempts, e.g., in the case of failure of delivery to the network must not affect the pattern of the statistically relevant call attempts.

MMS being sent from either side should contain information enabling validity and integrity checking during the evaluation. The recommended minimum set includes:

- signature assuring that only test system MMS are being considered;
- sequence number for MMS delivery timing.

If required, a test MMS may contain additional data (padding) to obtain content with structure and size assumed to be typical for QoS measurement purposes.

Further for general design of MMS testing, communication between the mobile testing system and its stationary counterpart is required to ensure that prior to actual testing, the MMS and SMS storage of the UE are cleared to exclude failures due to memory shortage or other limitation.

8.5.3.3 Transaction phase and parameter overview for MMS

Unlike SMS, posting and retrieving MMS is handled via a network service (WAP) rather than embedded in low-level functionality of the network. Also, uploading or downloading MMS is a process which takes considerable time. Therefore, the simple phase scheme – and, consequently, the possible-result definitions, have been extended by inserting subphases.

Table 8-10 shows transaction phase and parameter overview for MMS.

Table 8-10 – Transaction phase and parameter overview for MMS

Phase	Subphase	Parameters
Sending	Network access	Equipment types and capabilities
	MMS gateway access	MMS gateway address (proxy used)
	Upload	Content composition Content size User agent string (identification of the receiving UE)
Notification		Timeout for notification
Retrieval	Network access	Equipment type and capabilities
	MMS gateway access	MMS gateway address (proxy used)
	Download	
Pause		Time between MMS postings

8.5.3.4 Additional transaction result definitions for MMS

The following definitions are based on the assumption that, from technical feedback during the service usage, the user notices (and cares about) different ways a MMS transaction can fail. The following definitions are to be seen in conjunction with other similar result definitions:

- Service access failed: The MMS transaction could not be started due to inability to access the network or the service.
- Dropped while posting: Content could not be uploaded successfully.
- Lost: The content package was successfully uploaded, but the recipient did not get notification within the MEDT.
- Dropped while retrieving: The recipient received a notification, but was unable to retrieve the MMS.

NOTE – Unlike for the upload, this includes network or service access failure.

- Completed: Content has been successfully delivered to the B-party (which includes completeness of the package).

8.5.3.5 Content quality for MMS

In MMS transmission, transcoding may be used by the MMS infrastructure to adapt content from the format of the sending UE to a format supported by the receiving UE. Due to the fact that transcoding is integral part of MMS which is an explicit change of content, no direct comparison of content sent and received is possible.

It is currently assumed that correct transcoding is validated by means other than QoS measurements. Therefore, it is not subject of this Recommendation.

8.5.4 E-mail

8.5.4.1 Transaction definitions for e-mail

E-mails are sent from one e-mail client to another one via one or more e-mail servers. Both clients can independently use a wireless or a wired connection to their respective server.

Figure 8-2 depicts transaction definitions for e-mail.

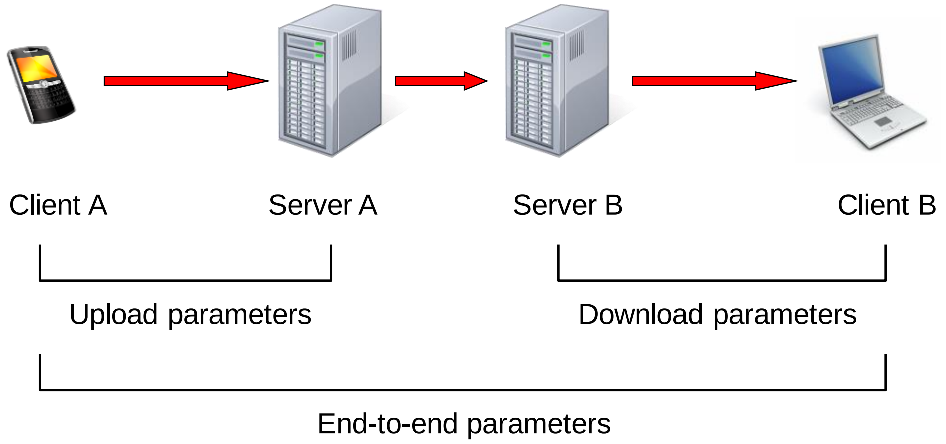


Figure 8-2 – Transaction definitions for e-mail

8.5.4.1.1 Protocols

The following protocols apply to e-mail transactions:

- For upload:
  - Simple mail transfer protocol (SMTP) [IETF RFC 2821].
- For download:
  - Internet message access protocol, version 4 (IMAP4), see [b-IETF RFC 3501];
  - Post office protocol, version 3 (POP3) [IETF RFC 1939].

NOTE – The POP3 protocol does not follow the store-and-forward approach because it is not capable to push (i.e., forward) e-mails to the receiver side. When using the POP3 protocol, the client has to poll the e-mail server for new e-mails.

8.5.4.1.2 Reference content

An e-mail consists of a header and a body. The body may contain text (e.g., plain text or HTML) or attachments (e.g., archives, music, documents, etc.).

For the download tests, it must be assured that the required reference content is present on the server. For the upload tests, it must be assured that successful upload is not prevented due to storage size limitations.

In addition, the measurements should not be influenced by additional (unwanted) or old e-mails. Appropriate measures should be taken (e.g., junk filter or clean-up of e-mail accounts).

Each e-mail uploaded shall have a unique identifier inside the e-mail header for unambiguous identification. Each e-mail downloaded shall have a unique identifier inside the e-mail header and the e-mail body for unambiguous identification. This unique identifier can be used for the aggregation of test results using the same set-up (e.g., e-mails of the same size).

Each e-mail uploaded shall have a unique test probe identifier inside the e-mail header to identify the uploading system. This is needed when different systems are uploading e-mails to a commonly used receiving account.

Refer to clause 10 for more details on the reference content to be used for e-mail testing.

8.5.4.1.3 Content integrity

After each e-mail transfer, it should be checked if the transferred reference content is received completely and is identical to the original data (including attachments).

For e-mail upload, additional procedures are necessary to guarantee the integrity of the transferred content.

The measured values of all QoS parameters, except for the access related QoS parameters, depend on the e-mail content used. Therefore, it is only allowed to aggregate test results which used the same e-mail content set-up, except for the access-related QoS parameters.

8.5.4.1.4 Push functionality

Both the client and the server shall support push functionality ("idle" feature) for tests using the IMAP4 protocol [IETF RFC 2177].

8.5.4.1.5 Header only download

The e-mail client used shall support the feature of downloading only the header of an e-mail contained within the inbox.

8.5.4.1.6 Timeouts

Every QoS parameter has its own configurable timeout setting which depends on the reference content and the technology used.

Refer to clause 10 for timeout settings.

8.5.4.1.7 General requirements and limitations

The used e-mail client should be comparable to popular e-mail clients used by users with respect to the behaviour and the performance. The performance of the used server should be comparable to commercially used systems.

The following information shall be logged:

- Type of e-mail client used (including version number, build, maximum number of parallel sockets).
- Authentication mode used.
- In case that a commercial client is used for testing not all of the pauses defined in clause 8.5.4.2 may be user configurable.

8.5.4.2 Transaction scenarios for e-mail

The following transaction scenarios are possible to test the service from an end user perspective:

1) Upload scenario.
2) Download scenario.
3) End-to-end scenario.

The first scenario only considers uploading an e-mail, while the second scenario only examines the download path. The third scenario tests the complete service chain, from the sending an e-mail by the A-party to the reception of the e-mail by the B-party.

8.5.4.2.1 Upload scenario

The upload test cycle is defined as follows:

1) Connect to the mobile network (set up IP connectivity);
2) Configurable pause (default 15 s);
3) Login client A to server A;
4) Configurable pause (default 15 s);
5) Attempt to upload up to, e.g., ten e-mails to server A, with a configurable pause (default 15 s) after each upload;
6) Disconnect from the mobile network;
7) Configurable pause (default 15 s) before starting the next test sequence.

Figure 8-3 shows message flow for e-mail upload scenario.

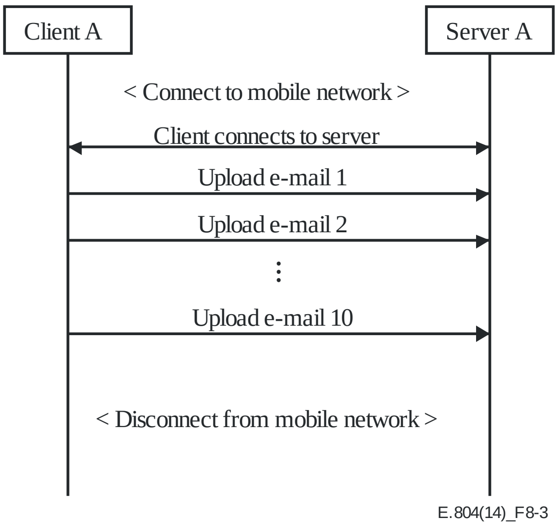


Figure 8-3 – Message flow for e-mail upload scenario

8.5.4.2.2 Download scenario

The download test cycle is defined as follows:

1) Connect to the mobile network (set up IP connectivity);
2) Configurable pause (default 15 s);
3) Login client A to server A;
4) Configurable pause (default 15 s);
5) Attempt to download all, e.g., ten e-mail header from server A, with a configurable pause (default 15 s) after each download;

6) Configurable pause (default 15 s);

7) Attempt to download up to, e.g., ten e-mails from server A, with a configurable pause (default 15 s) after each download;

8) Disconnect from the mobile network;

9) Configurable pause (default 15 s) before starting the next test sequence.

Figure 8-4 shows message flow for e-mail download scenario.

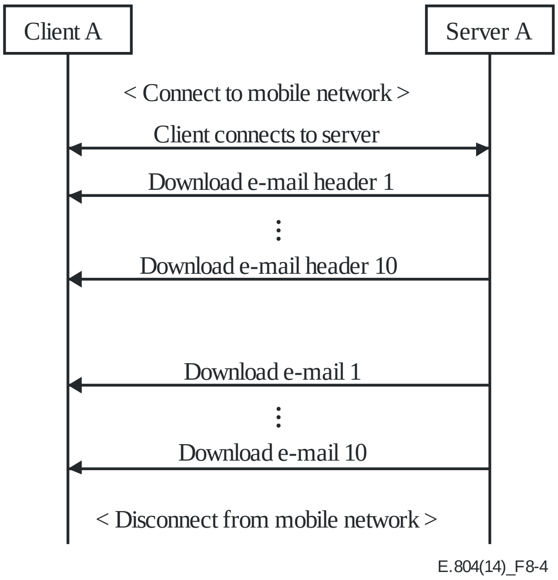


Figure 8-4 – Message flow for e-mail download scenario

8.5.4.2.3 End-to-end scenario with IMAP4

For the end-to-end scenario it is necessary to point out the dependency of the measurement of the download parameters from the e-mail upload: Only in case of a successful upload by the A-party will the B-party be able to download the incoming e-mail (either by polling the inbox or after having received a notification from the e-mail server).

The end-to-end test cycle is defined as follows:

Initialization phase:

1) The A-party and the B-party connect to the mobile network (set up IP connectivity);

2) Configurable pause (default 15 s);

3) Login client B to server B logs; client B then waits for incoming notifications.

Upload phase:

1) Client A connects to server A;

2) Configurable pause (default 15 s);

3) Upload up to ten e-mails to the server, with a configurable (default 15 s) pause after each upload;

4) Disconnect from the mobile network;

5) Configurable pause (default 15 s) before starting the next test sequence.

Notification-Phase

This phase runs in parallel to the upload phase.

1) Client B is notified of incoming e-mails by server B and initiates the header download.

2) Server B notifies client B of a new incoming e-mail.

Download phase:

This phase runs in parallel to the notification phase.

1) Attempt to download all notified e-mails.

Disconnect phase:

1) The A-party and the B-party disconnect from the mobile network;

2) Configurable pause (default 15 s) before starting the next test sequence.

8.5.5 SDS

Definitions for short data service (SDS) are for further study.

8.6 Data measurements

In the following clauses, data measurements for Internet-related services are described. While the process of obtaining a connection is different between circuit-switched and packet-switched access, there are many similarities for the actual data transfer phase.

8.6.1 Common aspects

8.6.1.1 Transaction definition and transaction types for data measurements

A transaction consists of access to a server to obtain content which is a closed unit from the user's perspective, e.g., a single downloaded file, or a web-site viewing access which may consist of several single objects which form the desired web page to be viewed.

8.6.1.2 Server types

The following categories of servers are distinguished; it is assumed that data service access is made to a server or entity within the general Internet domain:

- Third-party content in the public Internet. This type of counterpart is termed as A-servers.
- Accounts on Internet servers which are under control of the testing system or under control of testing personnel (e.g., web domains assigned for testing purposes). This type of counterpart shall be termed B-servers.
- Special servers not reachable via public Internet (e.g., in the GGSN domain), or equipped with additional instrumentation, e.g., for IP-level tracing or non-Internet capabilities (e.g., UDP testing). This type of counterpart shall be termed C-servers.

Access to a particular type of server will be termed using the same letter, e.g., A-access for access to an A-server.

It is assumed that A-servers are generally outside the sphere of control of testing systems, in particular, no baseline information for load or other conditions is obtainable. In case of the B-servers and the C-servers, such control may exist, but will typically be limited in some way due to IP-security reasons, in particular if IP access is intra-network.

Typically, when testing Internet services, availability may depend on influences other than those under test, and performance will be affected by third-party traffic. Meaningful tests therefore shall contain appropriate measures to exclude such effects from QoS assessment, or make sure that all networks under test are affected the same way. It is assumed that such tests are performed by fixed network units. Suggested methods are:

- Cyclical availability checks on target servers or domains.
- Cyclical access-time tests.

It is assumed that for services where login is required, accounts used by the test system are valid, give positive login, and are good for full access to all activities forming the test.

NOTE – This covers read/write privileges and directory-access rights (e.g., for FTP).

User's point of view typically includes assumptions of a time the user is ready to wait before an action is considered to be failed. At the same time, IP service access typically goes with timeout windows at several levels, e.g., for inactivity over a certain period of time. Test design must combine these two aspects to a reasonable, technically feasible set of parameters.

8.6.1.3 Test data content

When using web content (web sites or single pages) for testing, it must be taken into account that such content will typically change frequently (e.g., content of popular web portals) and therefore performance tests may give varying results over time. Test design must ensure that such effects are excluded from QoS assessment. Preferably, standardized and constant web content shall be used.

The degree of control a testing system has further depends on the type of service.

With e-mail and FTP (assuming appropriate access privileges), data content can be determined exactly (e.g., by uploading files to be downloaded as part of subsequent tests.

8.6.1.4 Transaction phase and parameter overview

8.6.1.4.1 General

To ensure comparability and statistical validity of transactions, the following outer conditions need to be constant throughout a test case:

- Access timing including behaviour in case of failure to obtain IP access. For reasons of comparability in density, it is required that all access attempts which form part of the statistics must follow a constant time pattern. Additional attempts, e.g., in the case of network or service unavailability must not affect the pattern of the statistically relevant access attempts.

For services using buffering, such as video streaming, there may be the situation that while actual data transfer is already running, the visual appearance is that of still waiting. This period of time shall be considered part of service usage.

Typically, in IP services periods of inactivity can occur with the session still intact. On the other hand, it must be taken into account that a useful PDP context is indicated by the system, which is in fact not useful. Therefore, testing shall include cyclical "lifecheck" measures. A possible method is Internet control message protocol (ICMP) pings to the DNS, not if such pings are blocked by the network, DNS accesses with dummy URL. Due to possible URL/IP address storage, it must be assured that actual DNS access takes place.

Independent of the access type, the following general information elements need to be logged in order to ensure reproducibility and comparability of tests:

- Operating system (type and version).
- MTU size.
- Logical location of server (e.g., public Internet or GGSN) with respect to the effects possibly created by the other traffic.
- Maximum server throughput inbound and outbound, per session response per connection.

8.6.1.4.2 Packet-switched access

Table 8-11 shows transaction phase and parameter overview for packet-switched access.

Table 8-11 – Transaction phase and parameter overview for packet-switched access

Phase	Subphase	Parameters
Internet access	Network access	Equipment types and capabilities Type of access used (CSD/PSD. UE initialization) APN and other initial settings
	Session access	Access and authentication parameters for the basic Internet access
	DNS access	DNS (in case of non-automatic DNS assignment during session access procedure)
	Domain access	Target URL, or server IP address Account being used (where appropriate)
Data transfer		Content composition and size
Cleardown		Pause between access attempts

8.6.1.4.3 Circuit-switched access

Table 8-12 shows transaction phase and parameter overview for circuit-switched access.

Table 8-12 – Transaction phase and parameter overview for circuit-switched access

Phase	Subphase	Parameters
Internet access	Connection set-up	Equipment types and capabilities Parameters for dial-up connection
	Session access	Access and authentication parameters for the basic Internet access
	DNS access	DNS (in case of non-automatic DNS assignment during session access procedure)
	Domain access	Target URL, or server IP address Account being used (where appropriate)
Data transfer		Content composition and size
Cleardown		Pause between access attempts

8.6.1.5 Possible transaction results

A data transaction is commonly termed "session" in contrast to "call" regardless of the type of connection (CS or PS). However, the general result term "dropped" shall be used, leading to "dropped session" for non-completed service usage.

In case of dropped sessions, the testing system shall indicate which of the possible principal causes occurred for the purpose of deciding of the network under test is to be blamed or not:

- Loss of radio connection.
- Loss of basic IP connection. If DNS access is still possible after a session loss, it is assumed that the basic IP connection is still intact.
- Loss of Internet access: If access to another domain or server is still possible, it shall be assumed that basic Internet services are still available.
- Loss of connection to the server.

For the single phases of Internet access (as part of general service access), the following additional definitions are given:

- **Session access**: Access and authentication procedure for basic Internet access. Successful access is assumed when a temporary IP address good for data transactions has been assigned to the testing system.
- **DNS access**: Obtaining an IP address from a URL (web site name, server name). Successful DNS access is assumed when an IP address has been obtained from a given, valid URL.
- **Domain access**: This is the actual Internet access as seen from the user's perspective. For testing purposes, a successful domain access can be assumed when communication with the target server has been proved. The respective action will depend on the service under test.

8.6.1.6 Content quality

Content quality relates to the quality as perceived by the user.

8.6.1.7 Content integrity

Content integrity is correctness and completeness.

Technical precautions should be taken to make sure that the content received is the one expected.

8.6.2 FTP

8.6.2.1 Transaction definition and transaction types for FTP

The basic transaction for FTP testing consists of Internet access to a FTP server followed by either downloading or uploading a single file of given size.

It is understood that basic access procedures such as server login shall not be considered as part of a transaction.

For download tests, it must be assured that the file to be downloaded is actually available on the server. For upload tests, it must be assured that no storage-size limitations prevent successful upload.

8.6.2.2 Transaction phase and parameter overview for FTP

Same as clause 8.6.1.

In addition, the following parameters shall be logged:

- type of FTP client used;
- protocol used (TCP/IP or UDP);
- type used (active or passive).

8.6.2.3 Possible transaction results for FTP

Same as clause 8.6.1.

8.6.2.4 Content quality for FTP

Same as clause 8.6.1.

8.6.2.5 Content integrity for FTP

Same as clause 8.6.1. It is recommended to use file size and checksum comparison.

8.6.3 HTTP

8.6.3.1 Transaction definition and transaction types for HTTP

The basic transaction for HTTP testing consists of Internet access followed by downloading a web site for a given URL response of given structure.

8.6.3.2 Transaction phase and parameter overview for HTTP

Same as clause 8.6.1.

In addition, the following parameters shall be logged:

- Type of browser used (including version number/build). For test-system browsers, maximum number of parallel socket connections.
- Reference web site used (e.g., structure, size of the documents, etc.).

8.6.3.3 Possible transaction results for HTTP

Same as clause 8.6.1.

8.6.3.4 Content quality for HTTP

Same as clause 8.6.1.

8.6.3.5 Content integrity for HTTP

It is assumed that a typical HTTP downloaded page consists of a main page and a number of sub-elements contained in this page (web elements). Based on the definitions in clause 8.6.1, recommended method for content integrity checking is to verify that the expected main page has been loaded, and to verify the expected number of web elements and their respective type, size and content. (see also Appendix VII).

QoS parameters shall only be compared directly (e.g., in benchmarks) if they are obtained without accelerators (performance enhancement proxies) or with the same accelerator behaviour for all networks involved. In practice, since the exact working of such accelerators is not known, this leads to the requirement that download times shall only be compared directly if such accelerators are not in the chain. Otherwise, when reporting such values the fact that different accelerator operation was in effect and that values are not to be compared shall be explicitly emphasized.

8.6.4 E-mail

Refer to clause 8.5.4 as the procedures described there are usable for direct services as well if notification is disabled on the e-mail server.

8.6.5 WAP

8.6.5.1 Transaction definition and transaction types for WAP

The basic transaction for WAP testing consists of WAP access and download of a WAP page.

To guarantee comparability and statistical validity of transactions, the following parameters have to be logged:

- Protocol (WAP1.x or WAP2.0).
- Access-point (gateway).
- Type of mobile used/emulated (user-agent string, accept-header).
- Mobile browser (version number, build).
- Depending on the test goal, one of the following clauses should be used:
  - A given reference WAP page (static WAP content)

    Picture count and text size has to be stable.
  - A static url, e.g., main page of the WAP-portal (variable WAP content)

    Picture count and text size have to fulfil minimum requirements depending on the url.

  NOTE 1 – It is required that the WAP page has a size greater than one packet.
- It is required that all WAP session attempts follow a constant time pattern.

NOTE 2 – The size of the mobile cache can be neglected, because all tests should be performed with an empty mobile cache.

The mobile specific effects (rendering) are not represented in case of non-application level testing.

8.6.5.2 Transaction phase and parameter overview for WAP

Same as clause 8.6.1.

In addition, the following parameters shall be logged:

- Protocol (WAP1.x or WAP2.0).
- Access-point (gateway).
- mobile used (user-agent string, accept-header, browser type).
- Structure of the WAP page (e.g., size of content and/or number of pictures).
- URL of the WAP page.

When measuring the parameter using drive tests or field tests, it is recommended that the following general method, shown in Figure 8-5, be used.

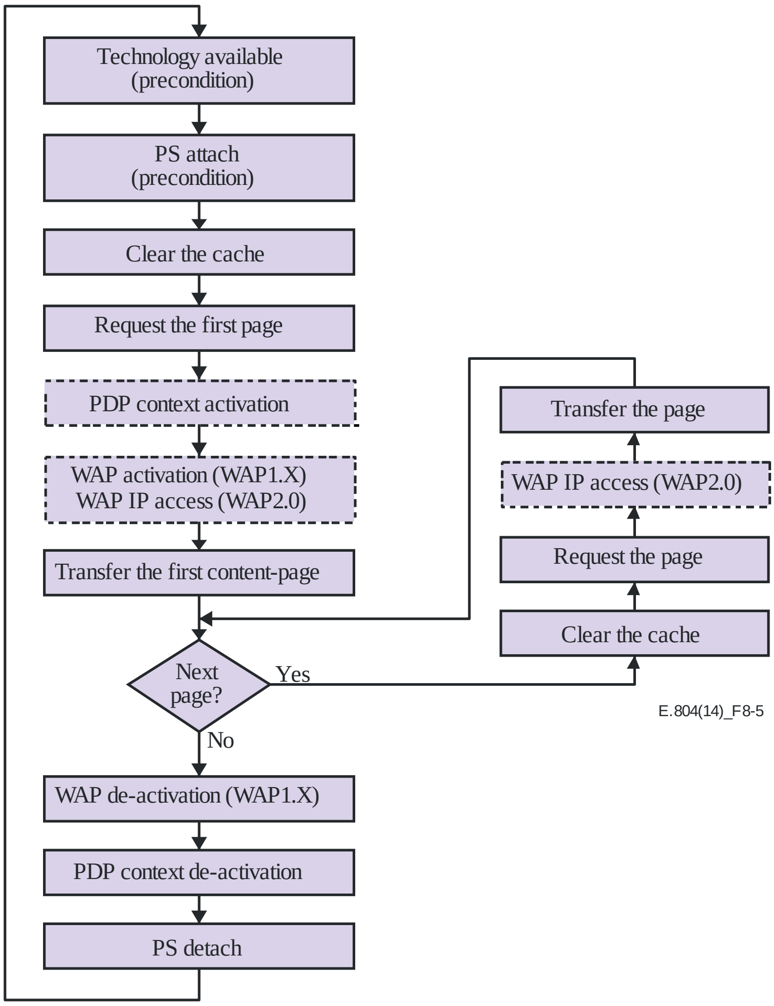


Figure 8-5 – General method for transaction phase and parameter overview for WAP

8.6.5.3 Possible transaction results for WAP

Same as clause 8.6.1.

8.6.5.4 Content quality for WAP

Same as clause 8.6.1.

8.6.5.5 Content integrity for WAP

The following is to verify that the received page is not an "error message" page and that the requested page is downloaded.

It is assumed that a typical WAP downloaded page consists of a main page and a number of sub-elements contained in this page, which may have sub-elements themselves. Based on the definitions in clause 8.6.1, recommended method for content integrity checking is to verify that all elements of the expected main page have been completely loaded.

- In case of testing a given reference WAP page (static WAP content), the whole (known) content should be verified.

In case of testing public WAP pages with variable WAP content, it is necessary to define and check the page regarding minimum criteria (e.g., five pictures and text size greater than 150 Byte). For identifying the content of the requested page it is necessary to check static page content (logo, keyword).

8.6.6 Streaming video

8.6.6.1 Transaction definition and transaction types for streaming video

The basic transaction for streaming video testing consists of Internet access to a streaming server followed by replay access to a given content from this server.

8.6.6.2 Transaction phase and parameter overview for streaming video

Same as clause 8.6.1.

In addition, the following parameters shall be logged:

- streaming server and client versions used;
- Operating system (OS) and relevant configuration of streaming server;
- streaming protocol used.

8.6.6.3 Possible transaction results for streaming video

Same as clause 8.6.1.

8.6.6.4 Content quality for streaming video

To be decided.

8.6.6.5 Content integrity for streaming video

To be decided.

8.6.7 Media download

8.6.7.1 Transaction definition and types for media download

There is a need for content providers and operators to control the usage of downloaded media objects. Download is the means by which a media object is delivered to the UE. Digital rights management (DRM) is the means to control the usage of the media object once it has been downloaded.

DRM enables content providers to define rules (rights) for how the media object should be used. It is possible to associate different rights with a single media object. Different rights may have different prices. A content provider can grant a user the rights to preview media objects for free and charge the user only for the full usage rights. Since the value lies in the rights and not in the media object itself, DRM makes it possible to sell the rights to use the media object, rather than selling the media object itself. The rights can be delivered to the consuming UE by downloading them together with the content or by sending the rights object separately from content. The former case (combined delivery) is simpler whereas the latter case (separate delivery) provides more security by making it more difficult to steal the content.

8.6.7.1.1 OMA network elements

Figure 8-6 shows OMA network elements.

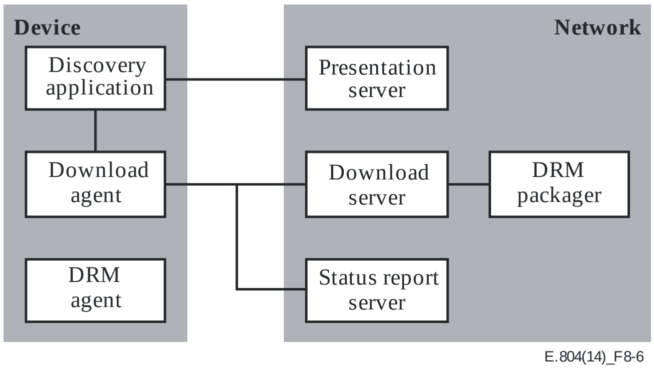


Figure 8-6 – OMA network elements

Download descriptor can be located on either the presentation server or the download server.

Discovery application: The user discovers media objects on the web by using a WAP browser or applications specifically created for one type of media.

Download agent: Launched after the discovery application downloaded the download descriptor and handles the remaining part of the download.

DRM agent: Handles DRM media objects.

Presentation server: May be a web or WAP "portal".

Download server: Is responsible for the download transaction, moving the actual media object from the server to the download agent.

Status report server: Receives the posted status reports.

DRM packager: Is wrapping media objects into DRM containers and is also responsible for generating the rights.

8.6.7.1.2 OMA download use cases

8.6.7.1.2.1 Combined OMA download use case

Figure 8-7 shows combined OMA download use case.

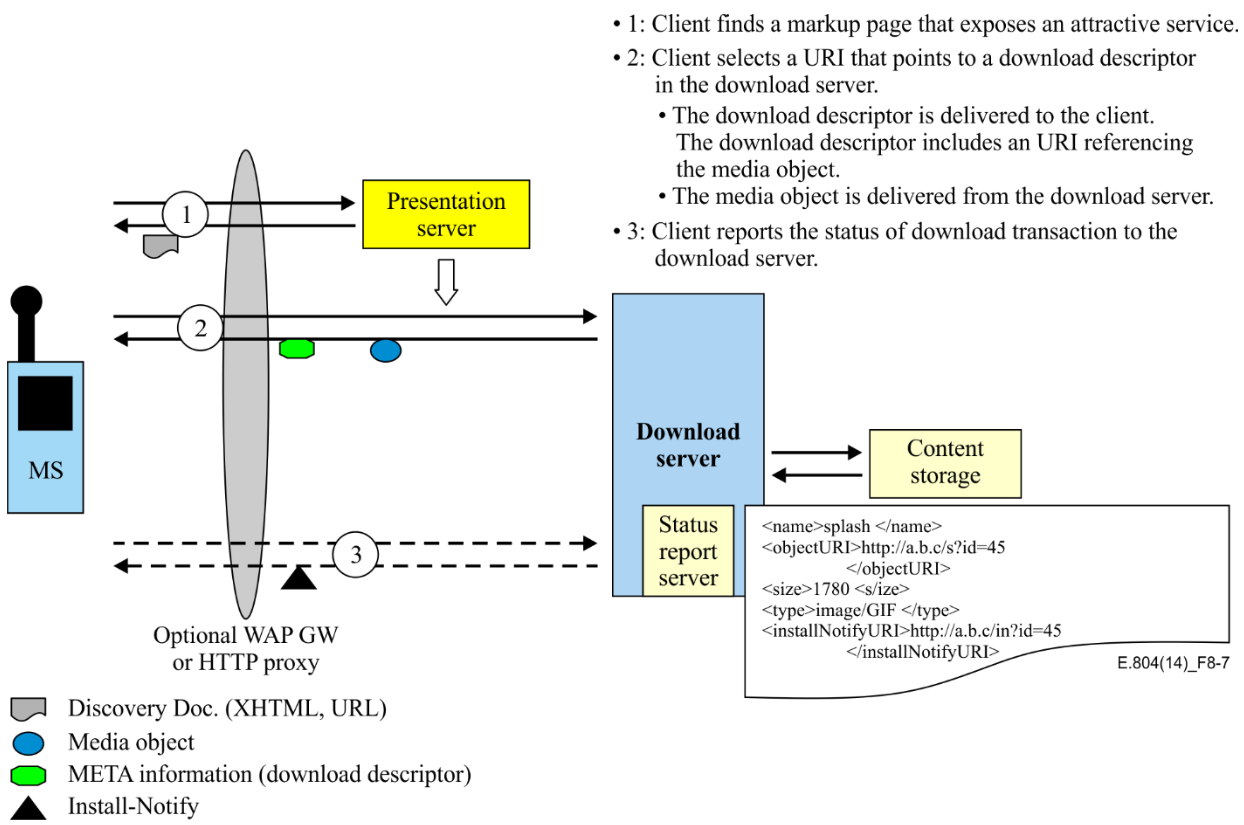


Figure 8-7 – Combined OMA download use case

The OMA download use cases extend the basic HTTP download use case by using a download descriptor. In the combined OMA download use case, a media object and a download descriptor is downloaded from an HTTP server by using one GET request and response. Because the media object and the download descriptor are both delivered together, the user is unable to confirm the download before the delivery of the media object. Optionally, a status report is posted to a URL specified in the download descriptor.

When this use case starts, the media object and the corresponding download descriptor is packaged into one multipart entity available on an HTTP URL, the URL must also be available on the device to the user (e.g., as a link in a WAP page). When this case ends, if the main scenario is completed, the media object is available on the device.

Main scenario:

1) User initiates a GET request to the URL – by selecting a link in a web page, for example.
2) Client sends GET request to the server and waits for a response.
3) Server serves up the requested resource, the multipart with the media object and the download descriptor, and sends a GET response back to client.
4) Client accepts GET response, with the HTTP headers and the data, from the server.
5) The information in the download descriptor is presented to the user.
6) If the download descriptor indicates that a status report shall be posted, a status report is posted to the specified URL.
7) The media object is made available to the user (e.g., saved on the file system).

8.6.7.1.2.2 Separate OMA download use case

Figure 8-8 shows separate OMA download use case.

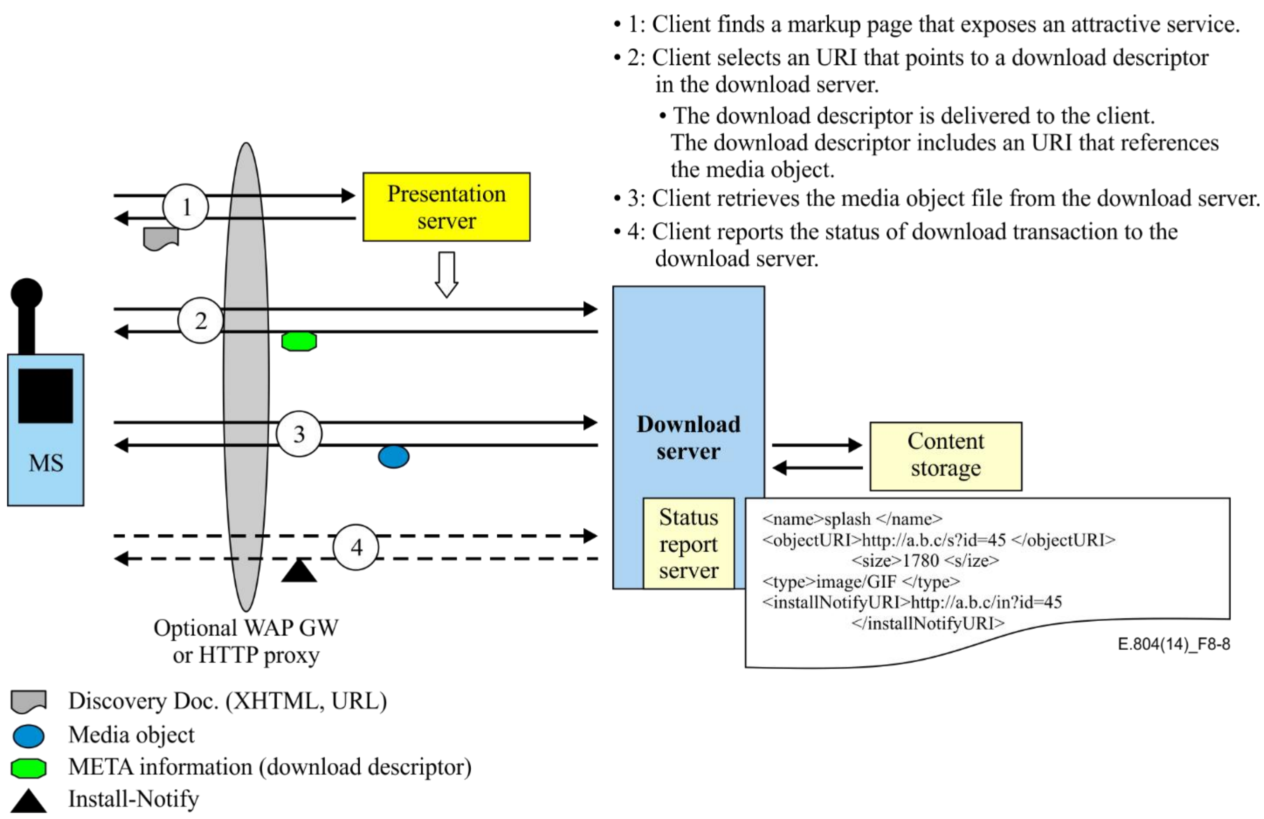


Figure 8-8 – Separate OMA download use case

In the separate OMA download use case, a media object and a download descriptor is downloaded from an HTTP server by using two GET request and response. The user is able to confirm the download based on pre-download capability checks in the device, and via a device specific download user interface. Optionally, a status report is posted to a URL specified in the download descriptor.

When this use case starts, the media object and download descriptor are available on two separate HTTP URLs, the download descriptor URL must also be available on the device to the user (e.g., as a link in a WAP page). The media object URL is available on the download descriptor. When this case ends, if the main scenario is completed, the media object is available on the device.

Main scenario:

1) User initiates a GET request to the URL – by selecting a link in a web page, for example.
2) Client sends GET request to the server and waits for a response.
3) Server serves up the requested resource, the download descriptor, and sends GET response back to client.
4) Client accepts GET response, with the HTTP headers and the data, from the server.
5) The information in the download descriptor is analysed by the UE (capability checks) and user is given a chance to confirm the download.
6) HTTP download is used to deliver the media object.
7) If the download descriptor indicates that a status report shall be posted, a status report is posted to the specified URL.
8) The media object is made available to the user (e.g., saved on the file system).

8.6.7.1.3 DRM use cases

8.6.7.1.3.1 Combined delivery DRM use case

Figure 8-9 shows combined delivery DRM use case.

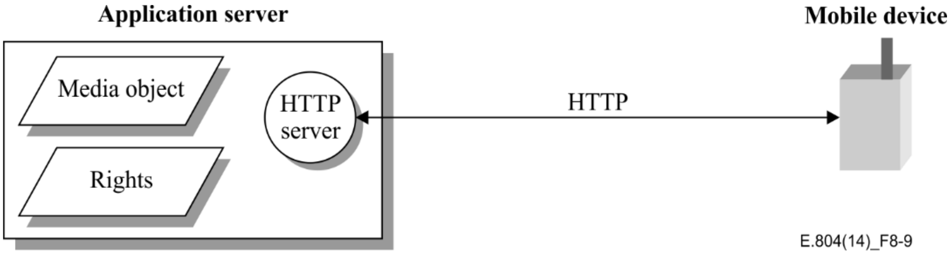


Figure 8-9 – Combined delivery DRM use case

A protected media object, together with the corresponding rights object, is downloaded from an HTTP server. When this case ends, if the main scenario is completed, the media object is available on the UE. It can be used only according to the granted usage rights.

Main scenario: The content provider, using a DRM packager, packages the media object and the rights object into one DRM message. The DRM message is made available to the UE (e.g., by publishing it on a web page). HTTP download or OMA download is used to download the DRM message (media and rights object). The user is using the media object and the DRM agent ensures that it is used according to the rights.

8.6.7.1.3.2 Separate delivery DRM use case

Figure 8-10 shows separate delivery DRM use case.

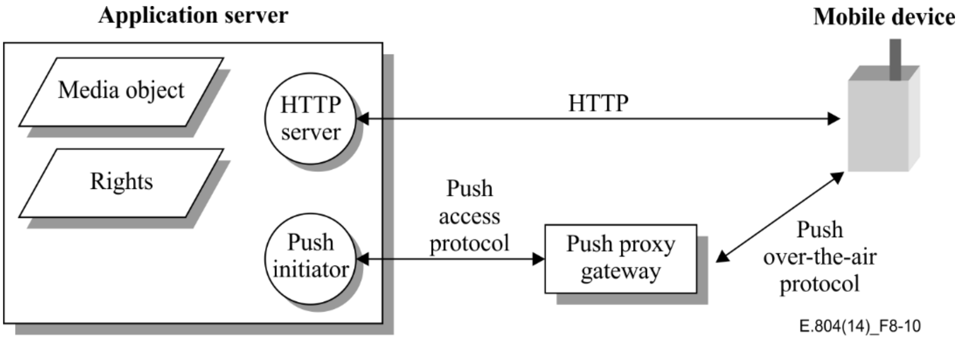


Figure 8-10 – Separate delivery DRM use case

A protected media object is downloaded from an HTTP server. Later, the corresponding rights object is delivered to the UE via WAP Push message (e.g., over SMS). When this case ends, if the main scenario is completed, the media object is available on the UE. It can be used only according to the granted usage rights.

Main scenario: The content provider, using a DRM packager, packages the encrypted media object into a DRM content format. The key to decrypt the media object is put into the rights object. The DRM content format URL is made available to the UE (e.g., by publishing it on a web page). The rights object is not available to the UE. The HTTP download or OMA download is used to deliver the DRM content format. The user waits for the rights object to be delivered via a WAP Push message from the content provider. (If OMA download was used, the status report could be used as a "trigger" in the network to push the rights object to the UE, after a successful download of the encrypted media object). The media object is decrypted using the key from the rights. The user is using the media object and the DRM agent ensures that it is used according to the rights.

8.6.7.2 Transaction phase and parameter overview for media download

8.6.7.2.1 Overview of basic parameters for media download

Charts in Figure 8-11 (taken from [IETF RFC 3481]) show on which basic parameters the media download service sequence is based.

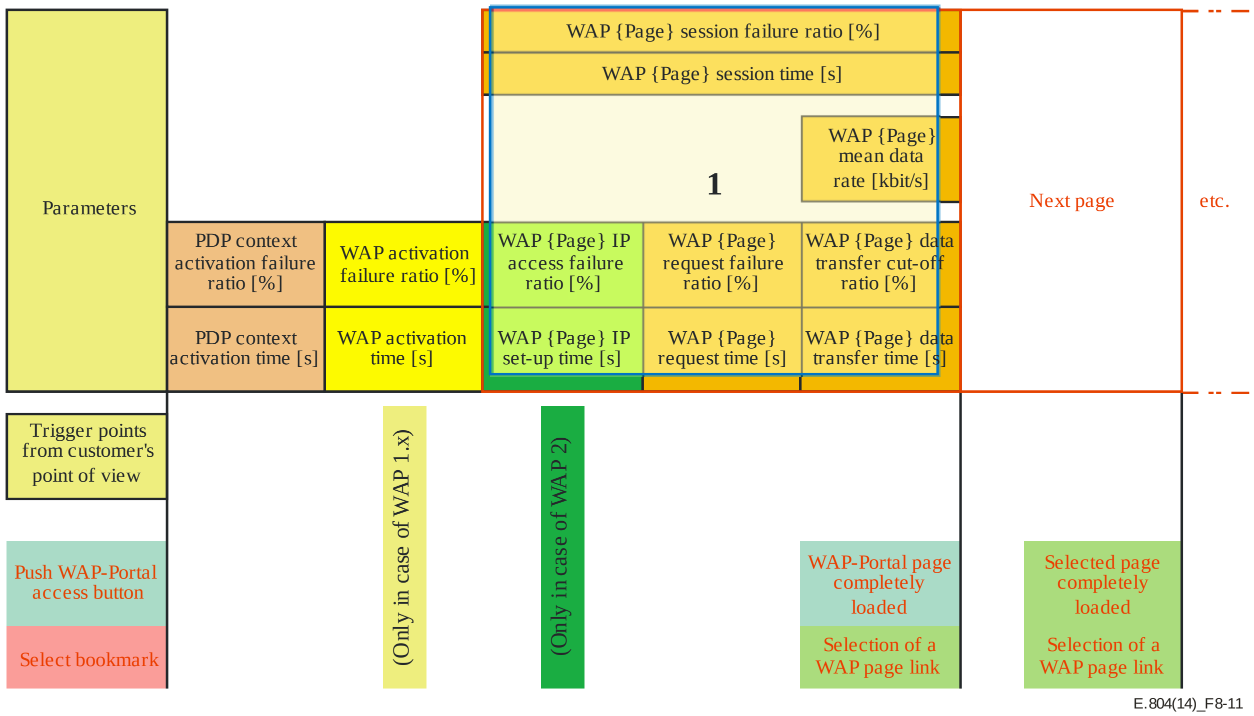
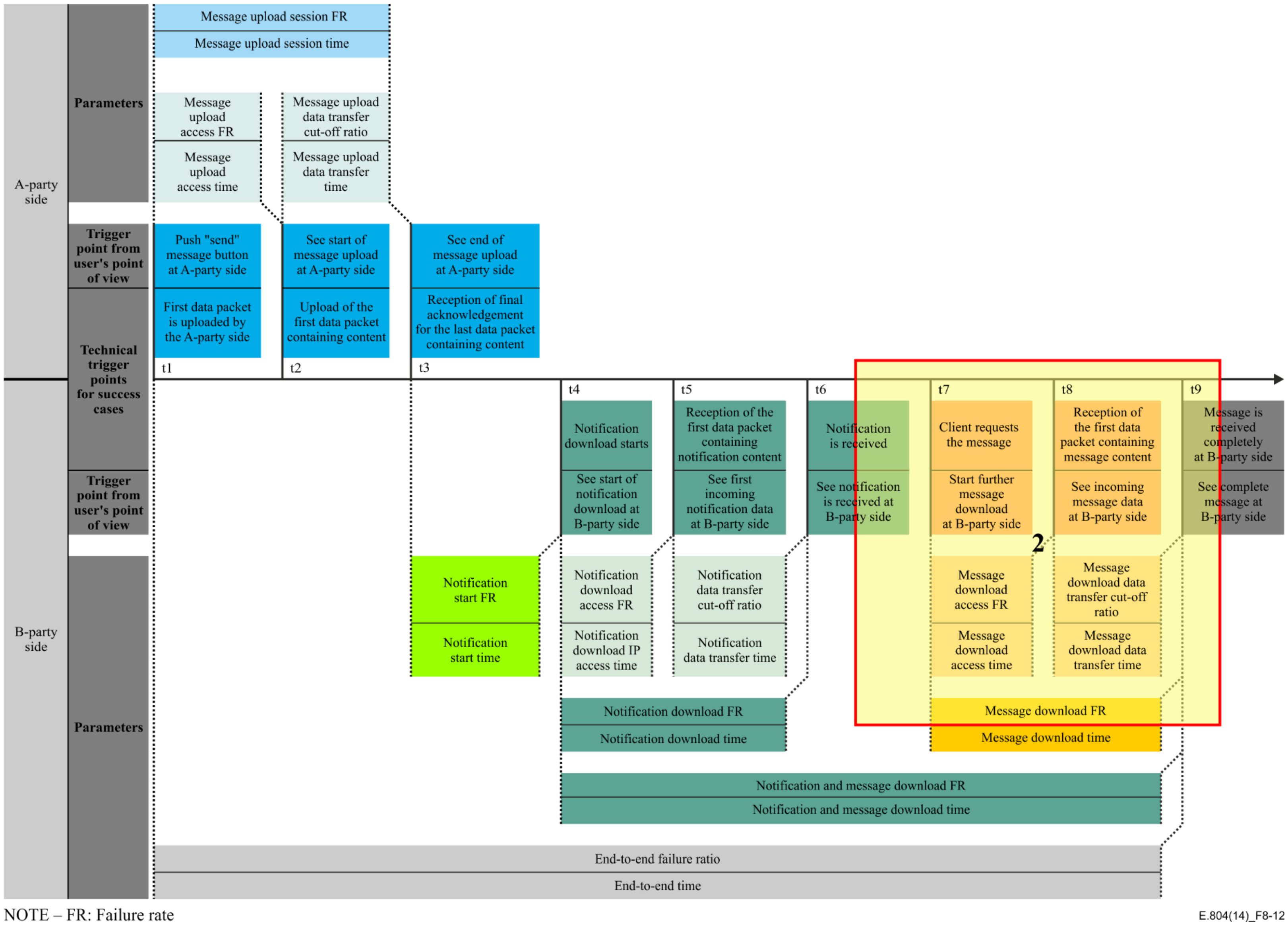


Figure 8-11 – WAP or equivalent parameters for download part

For the download part, the marked WAP or equivalent HTTP parameters should be used.

Figure 8-12 provides store-and-forward parameters for DRM separate delivery part.



NOTE – FR: Failure rate

Figure 8-12 – Store-and-forward parameters for DRM separate delivery part

For the DRM separate delivery part, the marked store-and-forward parameters should be used.

8.6.7.2.2 Media download subphases

The following text boxes are providing subphase specific information about the whole media download service sequence. The information includes the basic parameters used and the trigger points they are based on.

A: Purchase link

Trigger point from user's point of view	Technical description/protocol part
Start: Select download purchase link.	Start: • Same as WAP/HTTP
Stop: Payment receipt page is successfully loaded within the specified time limit.	Stop: • Same as WAP/HTTP (content: payment receipt page)

B: Download descriptor

Trigger point from user's point of view	Technical description/protocol part
Start: Appearance of the payment receipt page.	Start: • Same as WAP/HTTP (Stop from A)
Stop: Media file data page is successfully loaded.	Stop: • Same as WAP/HTTP (content: Download descriptor)

C: Media object

Trigger point from user's point of view	Technical description/protocol part
Start: Push download button.	Start: • Same as WAP/HTTP (Stop from B)
Stop: Media file download is completed.	Stop: • Same as WAP/HTTP (content: Media object)

D: Media and DRM object

Trigger point from user's point of view	Technical description/protocol part
Start: Push download button.	Start: • Same as WAP/HTTP (Stop from B)
Stop: Media file download is completed.	Stop: • Same as WAP/HTTP (content: Media and DRM object)

E: Media object and download descriptor

Trigger point from user's point of view	Technical description/protocol part
Start: Appearance of the payment receipt page	Start: • Same as WAP/HTTP (Stop from A)
Stop: Media file download is completed	Stop: • Same as WAP/HTTP (content: Media object and Download Descriptor)

F: Media and DRM object and download descriptor

Trigger point from user's point of view	Technical description/protocol part
Start: Appearance of the payment receipt page	Start: • Same as WAP/HTTP (Stop from A)
Stop: Media file download is completed.	Stop: • Same as WAP/HTTP (content: Media and DRM object and download descriptor)

G: Install notify

Trigger point from user's point of view	Technical description/protocol part
Start: Push save or install button if content is not saved automatically.	Start: • Same as WAP/HTTP (Stop from C, D, E or F)
Stop: Downloaded file is successfully saved or installed within the specified time limit.	Stop: • Sending of the Install Notify message

H: DRM object

Trigger point from user's point of view	Technical description/protocol part
Start: Downloaded file is successfully saved or installed within the specified time limit.	Start: • Sending of the Install Notify (Stop from G)
Stop: DRM Message is received	Stop: • Same as store-and-forward

8.6.7.2.3 Combined OMA download and combined delivery DRM use case

The overview chart shown in Figure 8-13 illustrates the minimum combination of subphases for a media download service sequence in case of combined OMA download and combined delivery DRM use case.

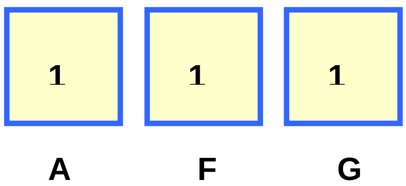


Figure 8-13 – Combined OMA download and combined delivery DRM use case

8.6.7.2.4 Separate OMA download and combined delivery DRM use case

The overview chart shown in Figure 8-14 illustrates the minimum combination of subphases for a media download service sequence in case of separate OMA download and combined delivery DRM use case.

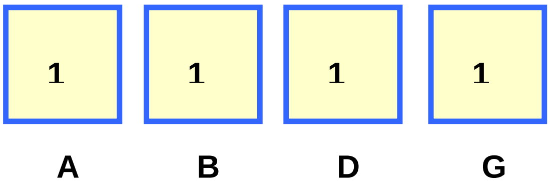


Figure 8-14 – Separate OMA download and combined delivery DRM use case

8.6.7.2.5 Combined OMA download and separate delivery DRM use case

The overview chart shown in Figure 8-15 illustrates the minimum combination of subphases for a media download service sequence in case of combined OMA download and separate delivery DRM use case.

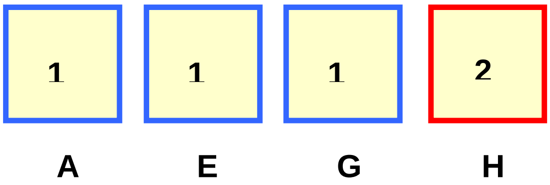


Figure 8-15 – Combined OMA download and separate delivery DRM use case

8.6.7.2.6 Separate OMA download and separate delivery DRM use case

The overview chart shown in Figure 8-16 illustrates the minimum combination of subphases for a media download service sequence in case of separate OMA download and separate delivery DRM use case.

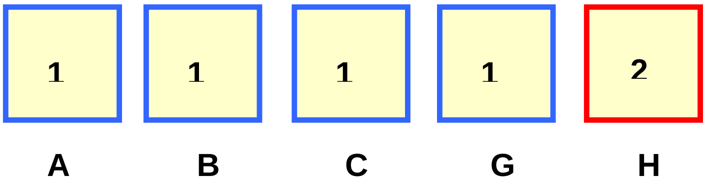


Figure 8-16 – Separate OMA download and separate delivery DRM use case

8.6.7.2.7 Additional information about parameters for media download

Same as clause 8.6.1 (WAP/HTTP/store-and-forward).

Media download session time as the sum over all the subphase times (whole sequence/session was successful).

Media download session failure ratio over all the subphases.

In addition, the following information should be logged:

- Use case information.
- Media object information: Artist, title, format and size.

8.6.7.2.8 Recommended testing method for media download

Test objects for media download should not be reported to chart ranking agencies and billed with real costs. They should be endlessly purchasable without impact on service sequence.

When measuring the parameters using drive tests or field tests, it is recommended that for example the general WAP method be used. In case of separate delivery DRM use case, the SMS memory should be erased before sequence starts and be checked at the end of the service sequence.

NOTE – Frequently, usage of public content can take influence on official chart rankings. Purchase of an already purchased song can change the service sequence.

8.6.7.3 Possible transaction results for media download

Same as clause 8.6.1 (WAP/HTTP/store-and-forward).

Media download session time as the sum over all the subphase times (when whole sequence/session was successful).

Media download session failure ratio over all the subphases.

8.6.7.4 Content quality for media download

Same as clause 8.6.1 (WAP/HTTP/store-and-forward).

NOTE – On the application layer the content quality for media download could be evaluated by playback of downloaded media.

8.6.7.5 Content integrity for media download

Same as clause 8.6.1 (WAP/HTTP/store-and-forward).

NOTE – On the application layer the content integrity for media download could be evaluated by playback of downloaded media.

9 Requirements for quality of service measurement equipment

9.1 Overview

9.1.1 General aspects

All tests are based on emulation of a typical user using services provided in a public mobile network (PMN). All of the services to be tested (see clause 7) can be emulated by the mobile QoS test equipment (MQT) which can be installed in a vehicle, can be carried around by a pedestrian or is installed for semi-stationary use (e.g., office environment).

Test scenarios need to distinguish the following principal user cases.

1) User-to-user services (typically telephony).
2) Store-and-forward services (e.g. SMS).
3) Information services (e.g., accessing the Internet or FTP download).
4) Push services.

Figure 9-1 provides an overview of a QoS test scenario.

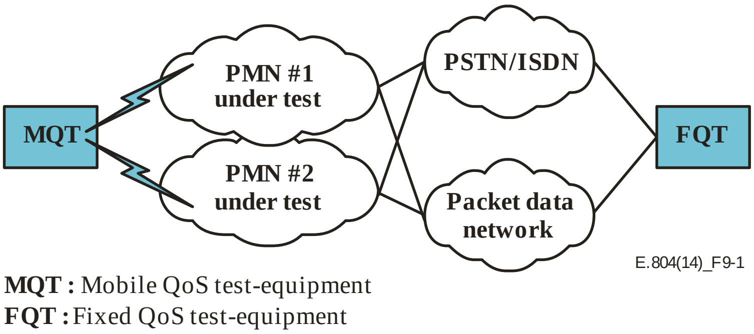


Figure 9-1 – QoS test scenario overview

Some of the services require test-equipment connected to a non-mobile network emulating the counterpart of the typical mobile user or the host offering the service. This part will be called fixed QoS test-equipment (FQT). The FQT may be connected via a public network (e.g., PSTN, ISDN or PDN) or via a network internal connection point (e.g., at MSC). The FQT for type 3) and 4) services could be composed as a (virtual) Internet service provider (ISP).

Below, requirements will be described on a per scenario basis. Those requirements not belonging to a specific scenario, e.g., antenna requirements will be grouped together.

Depending on how far the MQT can be automated or not, there is distinction between:

- MQT-LC: local control and operation; or
- MQT-RC: remote control and operation.

Although the same type of classification (-LC or -RC) can be made for FQT, most of the FQT are remote controlled.

9.1.2 Considerations on trigger points

Without loss of generality, it can be assumed that any feasible test equipment will contain some kind of communication terminal (UE) which may be a special type (e.g., a trace phone) or a standard UE. Also, it can be assumed that each of such devices will provide information from different communication layers, from application layer (close to the user interface) down to lower layers, e.g., operating-system events, TCP/IP layer, or layer 3 signalling information, which is used as trigger points for QOS parameters processing.

When considering the event chain, action is typically triggered by some emulated user action which finally causes some action on the air interface. This process of event propagation is deterministic, allowing some kind of mapping between layers, in the limits of available information, but will inevitably be associated with some communication and processing delay in each stage.

Therefore, choice of the layer from which to get trigger point information determines the view expressed in a QOS parameters. Generally, choosing lower-level events such as layer 3 gives a more network-centric view, while events on higher levels tend to produce views more user-related. From this, the following guidelines result:

- Within the same QoS parameters, the source layer for events used as trigger points should be the same.
- In benchmarking, all networks under test should be tested using the same type of UE, and QOS parameters for all networks under test should use trigger points from the same layer. When changing the source layer for a given trigger point, changes in QOS parameters should be expected, and respective calibration measurements should be taken to assess influence on QOS parameters both quantitatively and qualitatively.

9.2 General requirements

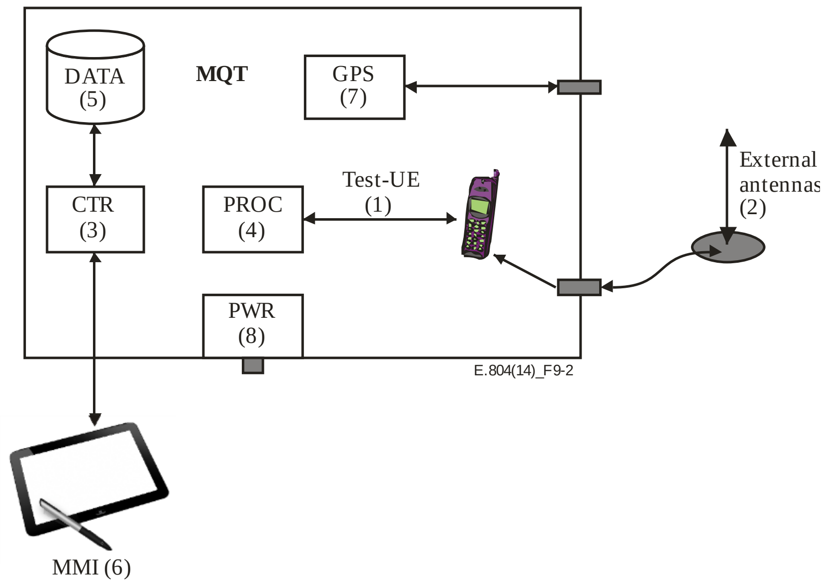
9.2.1 General requirement for data logging

The measurement system must provide means to collect and store reliably all relevant measurement data. Additionally, all configuration parameters have to be stored to be able to reproduce the test.

The system has to provide means to detect and sort out invalid measurement cycles to avoid misrepresenting statistics. The evaluation of the measured values is typically done during post processing. Measurement cycles which are removed from the measured data have to be reported.

9.2.2 Overview

The typical components of the mobile QoS test equipment (MQT) are as illustrated in Figure 9-2.



(1)	Test-UE:	User equipment emulating the typical user device (1…n).
(2)	Antenna:	Test-UE antenna or external antenna.
(3)	Controller:	Controls all the active parts in MQT.
(4)	Processor:	Controlling the test-UE and pre-processing measurement data. Optionally the tasks of the processor can be moved into controller.
(5)	Storage:	Storage of measurement data and programs.
(6)	MMI:	Man machine interface for control and configuration of a MQT-LC or for diagnosticsin case of a MQT-RC.
(7)	PS:	Positioning system (GPS or indoor navigation).
(8)	PWR:	Power supply.

Figure 9-2 – Typical components of the mobile QoS test equipment (MQT)

9.2.3 Required information for logging

9.2.3.1 Information on measurement set-up

Measurement set-up needs to be reproduced if necessary. This requires that the configuration of the measurement equipment, with which the measurement has been done, needs to be recorded.

NOTE – The measurement results not only depend on the configuration of the measurement equipment, also on other circumstances like day of the week and time of day which influence the measurement results considerably.

9.2.3.1.1 General information

The following list is considered to be the minimum requirements.

Information automatically collected:

- Versions of measurement equipment:
  - Hardware version.
  - Software version of measurement application.
  - Operating system version (operation system and service pack).
- Date, time of day (UTC time + time zone).

Manually entered information:

- User.
- Comment.
- All other information which cannot be collected automatically, on the required test case control parameters, to re-run the test case under the same conditions.

9.2.3.1.2 Information on user equipment in use

For the set-up of the user equipment (UE) in use, the following list represents the minimum required parameters:

Information automatically collected:

- Type of user equipment.
- Firmware version.
- Unique UE ID (e.g., IMEI, serial number, MAC, etc.).
- IMSI (configuration of SIM card can have a significant influence on the measurement result).
- Software version of driver for operating system, if used.
- All settings of the control software.

Manually entered information:

- Antenna:
  - Type.
  - Extra attenuation.
  - Total cable loss (e.g., Cables, RF combiners, etc.).

9.2.3.1.3 Information on store-and-forward set-ups

The following information has to be logged:

- Number of service centre.
- Access parameters.
- Transmitted message, video and/or audio.
- Timeout values.

9.2.3.1.4 Information on data test set-ups

For data tests, the following list represents the minimum required parameters.

Information automatically collected:

- Any stack parameter configuration or difference to the standard of the used operating system, e.g., information about TCP stack parameter changes.
- Servers.
- All settings of the control software concerning the data test set-up, e.g., FTP settings.

9.2.3.2 Measurement data

Each measured item has to be stored with the corresponding timestamp.

In addition to the trigger points specified in clause 7, the measurement equipment shall collect the following list of data:

- Network ID (MCC, MNC, CI, LAC), respective data items with a rate sufficient to track the user equipments behaviour.

9.2.3.3 Status information

The system has to record information about the status and progress of the current measurement.

9.2.3.4 Trigger points

The system has to record all necessary trigger points. See clause 7.

9.2.3.5 QoS parameters

If possible, QoS parameters shall be calculated during the measurement and be shown on the man machine interface (MMI).

For test cases on distributed systems, the calculation process for QoS parameters has to be done in a post process.

9.2.4 Test-UE

Basic requirements on the Test-UE:

- Compliant to the corresponding specifications (e.g., 3GPP, TETRA).
- For usage of the AT interface for trigger point measurements, the UE has to be conformant to the corresponding specifications, e.g., [ETSI TS 127 007] and [ETSI TS 127 005] for 3GPP and [ETSI EN 300 392-5] for TETRA.
- Remote controllable to initiate the QoS tests with required parameter settings.
- Delivering the necessary data which is required for the QoS test.
- For benchmark tests only, UE with the same capabilities can be used. (e.g., max. number of timeslot (TS) allowed or best type of speech codec).
- Depending on the test case, additional requirements may be relevant.

9.2.5 Antennas

Depending on the test case, the Test-UE's own antenna or an external antenna has to be used.

Where applicable, the antennas have to be arranged in a well-defined fixed way with a minimum distance to each other reducing RF influence on an acceptable level in an equal radio environment. The coupling loss between 2 UEs should be minimum 40.5 dB (as specified in clause 2 of [ETSI TS 100 910]).

NOTE 1 – Certain types of system integration will not allow satisfying these requirements, due to the limitation of space (e.g., backpack system). However, the antennas should be mounted on pre-determined points of the system which guarantee a minimum RF influence.

External antennas are typically used for measurements in vehicles (car, train, ship). Two RF scenarios can be defined for vehicles:

1) User with car kit and external antenna:
   - Car mounted antenna with no extra attenuation has to be used.
2) User without car kit, using the UE only (In-car use):
   - Internal antenna of UE can be used.

   NOTE 2 – Simulation without body loss.

   or

– External antenna connected with an overall attenuation of approximately 13 dB (cable loss plus extra attenuation) should be used.

NOTE 3 – More information about antenna attenuation is to be found in [ETSI TR 102 581].

9.2.6 Controller/processor/storage

The performance of the unit should be high enough and have no measurable impact on the correctness of the data collection. If the unit runs out of any resources, it shall inform the user on the MMI.

9.2.7 Man machine interface (MMI)

9.2.7.1 Local controlled systems

The MMI has to allow full operation of the system by the operator. The main functionalities can be monitored and the operator is alerted in case of main failures. Some basic failure diagnostic is possible.

9.2.7.2 Remote controlled systems

An MMI can be connected to perform basic tests and some failure diagnostics. Unattended systems should provide means to generate alarms upon operational faults in the system.

9.2.8 Time sources

The clocks on the measurement systems have to be synchronized periodically. The required accuracy of the timestamps of the measurement item is:

- Relative: 20 ms.
- Absolute: 250 ms.

9.2.9 Environmental conditions

The measurement system has to meet at least the minimal environmental conditions requirements defined in clauses 9.3.4 and 9.4.5.

9.3 Fixed QoS test equipment (FQT)

9.3.1 General

Depending on the test scenarios defined in clause 9.1 of this Recommendation, there will be different requirements for the FQT and MQT.

9.3.2 Controller

The performance of the unit should be high enough and have no measurable impact on the correctness of the data collection.

9.3.3 Time-sources

See clause 9.2.8.

9.3.4 Environmental conditions

The FQT has to meet the following environmental conditions:

- Temperature: 5 °C to 40 °C.
- Humidity: maximum 90 per cent.

9.3.5 FQT for telephony measurements

9.3.5.1 Common aspects

For the user-to-user services different FQTs can be used. Also, the connection point to the network can vary between an ISDN or PSTN line and a direct link at the MSC/RNC.

However, the type of server and the connection used have to be stored in the final measurement result.

The topology of the distributed system (MQT and FQT) results in the necessity of merging the measurement results made on the FQT and on the MQT. The result will be merged on the time base.

A unique identifier has to be included in the measurement files to enable identifying what files belong to the same measurements. This identifier has to be generated automatically by the measurement system.

9.3.5.2 Telephony voice

Calculating telephony QoS parameters including speech quality requires a counterpart on the fixed network. This is typically a PC which is connected to the PSTN. An application on the PC answers the incoming calls from the MQT or generates calls to the MQT.

A FQT application for telephony voice service handles the CS connections of the voice calls, like an answering machine. It controls 1 to n mobile station numbers (MSNs). Each MSN has its own profile for incoming or outgoing calls. The FQT application has to have the following capabilities:

- Auto answering of incoming calls.
- Speech quality assessment of the incoming voice calls (uplink).
- Providing speech samples for the downlink.
- Generating MT calls.

9.3.5.3 Telephony video

For telephony video different types of connection points are available:

- ISDN line/direct connection at MSC/RNC.
- UMTS user equipment with video telephony application.
- UMTS user equipment used as modem with a video telephony stack on the PC.

A FQT application for telephony video handles the CS connections of the video calls. It controls 1 to n connection points. Each connection point has its own profile for the incoming calls. The FQT application has to have the following capabilities:

- Auto answering of incoming video calls.
- Providing speech/video samples for the downlink.

9.3.6 FQT for store-and-forward services

Store-and-forward services typically transports information between two user equipments. Therefore, the typical difference between MQT and FQT does not apply for store-and-forward services. The quality of store-and-forward services depends on one side on the RF coverage and on the other side, much more than other services, on the network internal infrastructure like the SMSC.

The measurement system shall be able to measure all specified QoS parameters using user equipments (UEs) only.

9.3.6.1 Common aspects

A general problem is that the A-party does not get a confirmation if the message was received successfully by the B-party. Also, the network internal infrastructure can change the order of messages. The B-party of the measurement system has to be able to handle this effect.

9.3.7 FQT for data measurements

The FQT is the measurement server as the physical machine plus the service application.

It shall serve the service requests from the MQT in order to achieve the QoS parameter defined in clause 7. The server has to support the user profiles defined in clause 10.

It has to be ensured that the highest possible throughput on the measurement server is higher than the expected throughput for the measurements.

9.4 Mobile QoS test equipment (MQT)

9.4.1 General

Depending on the test scenarios defined in clause 9.1 of this Recommendation, there will be different requirements for the MQT.

9.4.2 Controller

The performance of the unit should be high enough and have no measurable impact on the correctness of the data collection.

9.4.3 Geographical positioning

Geographical data (position, speed, and heading) shall be collected during a mobile measurement. The geographical position can be retrieved by the following alternatives:

- Geographical information shall be taken from general positioning system (GPS) whenever possible.
- If no GPS signal is available other tools have to be used, e.g., navigation on a geo-referenced bitmap.

9.4.3.1 Format of geographical coordinates

- Outdoor: World geodetic system 1984 (WGS-84).
- Indoor: Fixed reference points on a geographical referenced map and WGS-84 positions calculated by the software.

9.4.3.2 Accuracy

The accuracy of the geographical positioning has to meet the following requirements:

- Outdoor: < 15 m.
- Indoor: < 10 m.

9.4.4 Time-sources

See clause 9.2.8.

9.4.5 Environmental conditions

The FQT has to meet at least the following environmental conditions:

- Temperature: 5 °C to 40 °C.
- Humidity: maximum 90 per cent.

Information about the necessary power supply has to be available.

9.4.6 MQT for telephony measurements

9.4.6.1 Common aspects

For the user-to-user services different MQTs can be used. The topology of the distributed system (MQT and FQT) results in the necessity of merging the measurement results made on the MQT and on the FQT. The result will be merged on the time base.

Due to the fact, that these systems will operate in moving vehicles or are carried around, they have to be ruggedly constructed. However, all local laws concerning industrial and/or road safety regulations have to be satisfied.

The manufacturer shall provide a system manual, which shows the set-up of the system. For easy identification, elements of the system have to be labelled. Labels on all cables and connectors shall simplify the operation of the system.

9.4.6.2 Telephony voice

The MQT for voice has to provide software, which provides the means to generate the necessary calls or to answer automatically incoming calls from the FQT.

The received speech sample is measured via the analogue output of the UE. As a consequence, the hardware set-up of the system has to include the necessary electrical adaptation between certain UEs and the controller (soundcard).

The call generator of the MQT shall establish voice call as specified in clause 10.

The hardware set-up of such a system is essential for the correctness of the measured QoS parameters. The system includes a UE specific part, which is responsible for the electrical adaptation of the audio output of the handset to the input of the soundcard of the controller.

9.4.6.3 Telephony video:

The MQT for video has to provide a call generator, which generates automatically calls as specified in clause 10 or to answer automatically incoming calls from the FQT. The system has to be capable of providing the following measurement methods.

- Video telephony using the UE application; or
- Video telephony using a video telephony stack on the PC.

9.4.7 MQT for store-and-forward services

See clause 9.3.6.

9.4.8 MQT for data measurements

9.4.8.1 Common aspects

Data measurements require a client on the MQT side for the different applications. These clients can be either part of the MQT application or an external application remotely controlled by the MQT. However, the MQT has to log the type and the configuration of the client used for the measurement.

9.4.8.2 MQT for FTP

The used FTP client has to support the following points:

- Active/passive mode.
- Common firewall support.
- Downloading from subdirectories.
- Uploading to subdirectories.

9.4.8.3 MQT for e-mail

The e-mail client has to support POP3 and SMTP or IMAP for sending and receiving e-mails.

9.4.8.4 MQT for HTTP

The used HTTP client (browser) has to support all common HTTP versions. The client has to download an implemented static version of one of the ETSI reference webpages[1]. For details see [ETSI TR 102 505]. The used HTTP version and all other settings have to be logged.

[1] Available at: http://portal.etsi.org/TBSiteMap/STQ/HTLMReferenceWebPage.aspx

9.4.8.5 MQT for WAP

The WAP client shall support all common WAP versions. The used WAP version has to be logged.

9.4.8.6 MQT for streaming services

The MQT has to support the required streaming clients.

9.5 Mobile based measurement equipment

The functionalities of the MQT as described in the clause 7 can also be realised on a single mobile phone. In such mobile based measurement equipment different types of QoS tests can run and test results can be logged for post processing. Mobile based measurement equipment may be controlled by an application, remotely or may be operated by a person.

This type of equipment is for further study.

10 Definition of typical measurement profiles

10.1 Measurement profiles

Measurement profiles are required to enable benchmarking of different networks both within and outside the national boundaries. It is necessary to have these profiles so that when a specific set of tests is carried out then users are comparing "like for like" performance.

It is recognized that many factors will affect comparability:

- number of sessions;
- session durations;
- time between sessions;
- demanded QoS settings for data services;
- protocol settings (like TCP/IP settings for data services or adaptive multi-rate (AMR) settings for speech services);
- usage profile during the session;
- fixed network test equipment like test servers for data sessions;
- user profile stored in the HLR or the GPRS register (GR);
- geographic location;
- type of location (e.g., indoor, hotspot, city, suburban, rural, train, etc.);
- speed when mobile;
- type of vehicle;
- type of antenna;
- handset type;
- handset hardware and firmware version;
- service being tested and limitations of service;
- network configuration;
- mobile users' population density.

For the points mentioned above where there is no recommendation or requirement in this Recommendation, the settings experienced by a regular user of the service under test in the network under test shall be used as a guideline.

As far as possible, all particular values, e.g., timeout values are named preserving the name of the respective QoS parameters as defined in clause 7.

10.1.1 Classification of measurement environments

For interpretation and comparability of test results, it is important to know in which measurement environment the tests were performed. The environment classifications described in Tables 10-1 and 10-2 shall be used. Since the type of the measurement locations may be interpreted differently, the particular understanding of the location type determining a category shall be described in the results report.

Table 10-1 – Stationary tests

Category	Location type	Additional information
S1O:	Airports, railway stations, shopping centres, malls, business districts and exhibition areas	Outdoor measurement
S1I:	Airports, railway stations, shopping centres, malls, business districts and exhibition areas	Indoor measurements

Table 10-2 – Drive tests and walk tests

Category	Location Type	Additional information
D1:	Train measurements	
D2:	Urban areas (medium cities)	
D3:	Highways	
D4:	Rural areas (country roads)	
D5:	Large cities	
W1:	Walk tests (indoor measurements)	
W2:	Walk tests (outdoor measurements)	
NOTE – Drive tests may be performed in the car using external antenna with an appropriate attenuation.		

10.1.2 Service profiles

This clause describes recommended service profiles used for testing.

10.1.2.1 Telephony

The service profiles defined for telephony might be applicable for different scenarios, e.g., mobile-to-mobile or mobile-to-fixed, and the respective results should not be compared directly, if so.

To achieve comparable statistics when performing a benchmark, there should be no fixed pause between calls. Instead, a fixed call window is defined in which the call has to be performed. If the call fails or drops, the next call attempt shall only be made when the next call window arrives.

A minimum pause interval between two call attempts should be applied to prevent network related problems between connection release and the next establishment (e.g., signalling in the packet switched data (PSD) or mobility management).

10.1.2.1.1 Speech telephony

The following call durations (CDs) shall be used:

- CD1: 10 seconds for call set-up testing;
- CD2: 120 seconds for typical tests, default call duration;
- CD3: 300 seconds for stability tests;

- CD4: The call duration for, e.g., TETRA is for further study.

Call window: Call duration + 30 seconds, (for the set-up and release phases) + 30 seconds (for the minimum pause interval), for the default call duration this results in 180 seconds.

Timeout values:

- Telephony {service non-accessibility | set-up time} timeout: 20 seconds.

10.1.2.1.2 Video telephony

Video telephony should be tested in mobile-to-mobile scenarios. The following call durations shall be used:

- CD1: 10 seconds for call set-up testing;
- CD2: 120 seconds for typical tests (default call duration);
- CD3: 300 seconds for stability tests.

Call window: Call duration + 30 seconds (for the set-up and release phases) + 30 seconds (for minimum pause interval), for the default call duration this results in 180 seconds.

Timeout values:

- VT service {Non-accessibility | access time} timeout: 20 seconds;
- VT audio or video set-up {failure ratio | time} timeout: 30 seconds.

10.1.2.1.3 Group call

Group calls should be tested for the mobile-to-mobile(s) scenarios. The following call durations shall be used:

- CD1: 20 seconds for typical tests (default call duration);
- CD2: 60 seconds for stability tests.

Call window: Call duration + 30 seconds (for the set-up and release phases), + 30 seconds (for minimum pause interval), for the default call duration this results in 80 seconds.

Timeout values:

- Group call service non-accessibility timeout: 5 seconds.

10.1.2.2 Messaging services

For all messaging services it is important that the recipient of a message is not interrupted by the next message while retrieving the previous one. For this reason, it is important that the interval between sending two messages is larger than the 95th percentile of the end-to-end duration, unless measures are taken to avoid this kind of interference.

It should be noted that mobility of either the sender of a message or the receiver of a message or both can have an impact on the results. Therefore, it is recommended that measurements are not only performed stationary, but also with mobility of one or both participants. In all cases, the used scenario has to be stated.

10.1.2.2.1 {SMS | SDS}

{SMS | SDS} should be tested in mobile-to-mobile scenarios and without concatenation. Thus, the user data should be chosen in a way that it will fit into a single message.

The interval between two consecutive SMS shall be 70 seconds.

The transmission window of measurements shall be 175 seconds.

Timeout-values:

- {SMS | SDS} service non-accessibility timeout: 65 seconds;

- {SMS | SDS} completion failure ratio timeout: 175 seconds;
- {SMS | SDS} receive confirmation failure ratio timeout: for further study;
- {SMS | SDS} consumed confirmation failure ratio timeout: for further study.

10.1.2.2.2 Concatenated {SMS | SDS}

For further study.

10.1.2.2.3 MMS

MMS should be tested end-to-end. That means, a MMS sent by the A-Party should be received by the B-Party using also a mobile phone. The advantage of this testing is that the MO direction at the A-Party and the MT direction at the B-Party can be measured. Both directions together are the end-to-end parameters described in clause 7.

The following MMS sizes shall be used:

- MMS1: 2 kbyte;
- MMS2: 28 kbyte;
- MMS3: 90 kbyte.

If the MMS is not delivered at the destination after the MMS end-to-end failure ratio timeout, the MMS delivery is considered failed. MMS delivered after this time is not taken into account for end-to-end delay, but into end-to-end failure ratio.

Timeouts for MMS over GPRS:

The timeouts for MMS send, retrieval and end-to-end failure are dependent on the MMS size. For GPRS all MMS uploads with less than 5 kbits and all MMS downloads with less than 10 kbits are considered to be cut-off.

MMS send failure ratio (MO) timeout: (195 + size [kbyte] $\times$ 8 $\times$ 2/10) [seconds].

MMS retrieval failure ratio (MT) timeout: (195 + size [kbyte] $\times$ 8 $\times$ 1/10) [seconds].

The fixed part of 195 seconds incorporate the time for PDP context activation and WAP activation and shall be used as a whole, i.e., the single timeouts for PDP context and WAP activation shall not be considered.

MMS end-to-end delivery failure ratio timeout: (590 + size [kbyte] $\times$ 8 $\times$ 2/10 + size [kbyte] $\times$ 8 $\times$ 1/10) [seconds].

The fixed part of the 590 seconds incorporates the time for PDP context activations, WAP activations and notification and a security margin. It shall be regarded as a whole, i.e., the single timeouts shall not be considered.

MMS notification failure ratio timeout: 120 seconds.

Timeouts for MMS over UMTS:

- The timeouts for MMS send, retrieval and end-to-end failure are dependent on the MMS size.
- The respective required minimum upload and download data rate is for further study.
- MMS send failure ratio (MO) timeout: for further study.
- MMS retrieval failure ratio (MT) timeout: for further study.
- MMS end-to-end delivery failure ratio timeout: for further study.
- MMS notification failure ratio timeout: 120 seconds.

10.1.2.3 Data services

10.1.2.3.1 Circuit switched

Circuit switched data services shall be tested for 100 per cent of mobile originated calls (MOCs). Call duration shall be either 300 seconds or is defined by the usage profile used during the data session. The pause interval between call attempts shall be 30 seconds. The usage profile used during the data session is defined in clause 10.1.3.

10.1.2.3.2 Packet switched

Packet switched data services shall be tested for 100 per cent of MOC sessions. Session duration shall be either 300 seconds or is defined by the usage profile used during the data session. The pause interval between session set-up attempts shall be 30 seconds. The usage profile used during the data session is defined in clause 10.1.3.

NOTE – In order to ensure comparable results in benchmark testing (on changing access technologies), the number of measurements per time on the compared channels should be equal (by using test windows or regular intermediate results) or the individual measurements should be appropriately weighted in the aggregation.

10.1.2.3.2.1 Service-independent timeout values

- Attach timeout: 75 seconds.

It might occur that the user equipment sends more than one attach request towards the SGSN, since retries are necessary. A maximum of four retries are possible (timer T3310 expires after 15 seconds for each attempt, see [ETSI TS 124 008].

- PDP context activation timeout for GSM and 3G networks: 150 seconds.

It might occur that the user equipment sends more than one PDP context activation request towards the SGSN, since retries are necessary. A maximum of four retries are possible (timer T3380 expires after 30 seconds for each attempt, see [ETSI TS 124 008]).

- PDP context activation timeout for TETRA networks: 120 seconds.

The PDP_ACTIVATE_WAIT timer expires after 30 seconds for each attempt, see clause 28.5.1.1 of [ETSI EN 300 392-2]. The number of possible retries RETRY_ACTIVATION is fixed to 3, see clause 28.5.2 of [ETSI EN 300 392-2]. Therefore, the timeout interval for the PDP context activation procedure is 120 seconds, i.e., if the PDP context activation procedure was not completed after 120 seconds, it is considered as failure.

10.1.2.3.2.2 Service-dependent timeout values

Timeout values for an FTP (UL and DL) service are:

- Service accessibility timeout: 150 seconds + IP-service access timeout.
- Set-up time timeout: 150 seconds + IP-service access timeout.
- IP-service access timeout: 30 seconds.

Data transfer cut-off timeout:

- Over GPRS:

  UL: File size [kbyte] × 8 × 2/19;

  DL: File size [kbyte] × 8 × 1/10.
- Over UMTS:

  UL and DL: File size [kbyte] × 8 × 1/50.
- Dual mode: The average between the timeout over GPRS and UMTS shall be considered.

Timeout values for an HTTP service are:

- Service accessibility timeout: 150 seconds + IP-service access timeout.
- Set-up time timeout: 150 seconds + IP-service access timeout.
- IP-service access timeout: 30 seconds.
- Data transfer cut-off timeout:
  - Over GPRS:

    UL: File size [kbyte] × 8 × 2/10;

    DL: File size [kbyte] × 8 × 1/10.
  - Over UMTS:

    UL and DL: File size [kbyte] × 8 × 1/50.
  - Dual mode: The average between the timeout over GPRS and UMTS shall be considered.

Timeout values for an e-mail (IMAP, POP3 and SMTP) service are:

- Service accessibility timeout: 150 seconds + IP-service access timeout.
- Set-up time timeout: 150 seconds + IP-service access timeout.
- IP-service access timeout: 60 seconds.

Data transfer cut-off timeout:

- Over GPRS:

  UL: File size [kbyte] × 8 × 2/10;

  DL: File size [kbyte] × 8 × 1/10.
- Over UMTS:

  UL and DL: File size [kbyte] × 8 × 1/50.
- Dual mode: The average between the timeout over GPRS and UMTS shall be considered.

Timeout values for a streaming service are:

- Streaming service access timeout: 30 seconds.
- Stream reproduction start timeout (initial buffering): 60 seconds.
- Rebuffering timeout (single): 30 seconds.
- Rebuffering timeout (total): 75 per cent of session time.

NOTE 1 – It might occur that a streaming client goes from rebuffering back to playback within the rebuffering timeout (single), but goes back to one or more rebuffering periods afterwards. The rebuffering timeout (total) defines a limit in terms of a maximum of allocated time for all rebuffering periods.

- Maximum allowed rebuffering frequency: 20 rebuf/min.

NOTE 2 – The streaming client might go into recurrent rebufferings. If the number of rebuffering occurrences within a minute exceeds this limit the session is aborted.

- Teardown timeout: 30 seconds.

10.1.3 Usage profiles for data sessions

For data session measurements, the client application, e.g., web browser, FTP client or mail client, as well as the server application, e.g., web server, FTP server or mail server, should behave in a way similar to the majority of client applications used by the user and server applications used by the data service providers.

Also, the operating system on both sides, namely the client and the server side, should be chosen with respect to the operating system commonly used by the user and the data service provider, respectively.

In case a network operator whose network is to be measured provides the user with some client application, it should be ensured that any change introduced by such application to the client operating system should have been applied prior to the measurement, as well. This is especially true for changes which would have an impact on the measurement results, for example changes to the operating system's TCP stack. Such client applications are for example provided in order to allow the user a single point of access to network related configurations and to data service clients, e.g., web browser, mail client, etc. Furthermore, such client application might optimize operating system parameters, including tuning of, e.g., TCP settings, with respect to the connection type and technology to be used.

NOTE 1 – In some cases it is desirable not to install such client application itself since this might have some unwanted impact on the measurement. For example, if such application would generate unwanted network traffic in order to check for updates or if the application would continuously try to connect to the mobile device preventing some measurement application form controlling the device.

NOTE 2 – The use of different operating systems as well as the use of operating systems with different TCP parameter settings in general might have a large impact on the results obtained. With respect to the operating systems, this is due to the different implementations of the TCP/IP stack. This needs to be considered in the case of benchmarking exercises where the client and/or server operating systems and/or the changes applied on the client sides of the compared networks are not the same. However, the settings of the TCP stack of both the client and the server operating system should be recorded in order to allow for better interpretation and in-depth evaluation of the measurement data.

NOTE 3 – Proxy servers installed in the networks IP core network may act as the TCP peer instead of the application server the tests are performed against. In benchmarking scenarios, the existence of different proxy servers might have an additional impact on the results, which should be considered when comparing them.

NOTE 4 – A reference for optimized TCP parameters over cellular networks, like the second-generation (2.5G) and the third-generation (3G) wireless networks, is [IETF RFC 3481].

NOTE 5 – Some of the client applications referred to above might also change the way a data service is accessed from the client side, for example by introducing some client to the user's operating system which changes the transport protocol between the user's operating system and some proxy server. In such case, the trigger points as defined in clause 7 might not be measurable anymore and should therefore be mapped to the application layer. Especially, in case of benchmarking exercises where different operating systems are used on the client side, such mapping might have an additional influence on the measurements. The definition of such trigger point mappings for the different operations systems is not in the scope of this Recommendation.

For all tests, a dedicated test server should be used as a well-defined reference. Under no circumstances should a commercial server be used, since the content on such a server may change over time. This makes future reproduction of the results impossible.

In order to avoid issues with DNS host name resolution like including effects of DNS caching strategies of the used operating system into the measurement, the test server should either be identified by an IP address and not by its fully qualified domain name (FQDN) or it shall be ensured that the local resolver will contact a remote DNS name server in case a host name resolution is requested by an application. Furthermore, the DNS name server should be able to perform the resolution within its local zone, in case DNS lookup time is to be included into any quality of service parameter to get calculated. The latter is needed in order to exclude effects of DNS caching strategies of the DNS name server(s) involved into the measurement.

The measurement of data services should take place against a reference server only used for testing purposes. The reference server should be connected to the public Internet with a link capable of carrying the accumulated traffic of all mobiles testing against that server (e.g., if a benchmark with 4 networks is performed, the server should be able to deliver at least 4 times the maximum nominal speed of a single wireless link). There should be no bias concerning the IP connectivity to this server from a specific operator (e.g., bandwidth or hop-count).

The capabilities of the test UE shall be stated in the results report.

10.1.3.1 Web browsing using HTTP

For the measurement of web browsing, the reference server should contain a static version of one of the ETSI reference webpages[2]. For details see [ETSI TR 102 505].

The browser used for testing should behave in a way similar to the browser used by most of the users. It should be able to support the same HTTP capabilities and headers and open the same number of parallel download threads to download the content as the reference browser.

After one test cycle (one download of the reference page), the complete data representation of the reference page content shall be cleared from the local cache of the browser. Furthermore, it should be made sure, that all TCP connections between the server and the client are closed (i.e., no HTTP kept alive). There should be a pause of at least 6 seconds between the cycles.

NOTE – Any data related to performance enhancement proxy (PEP) settings, like JavaScript scripts or Cookies need to be prevented from getting cleared from the browser's cache.

For the test, only HTTP download should be used. HTTP upload shall not be used.

Testing of content integrity is not mandatory for this test, but highly recommended.

10.1.3.2 E-mail access

E-mail access should take place against a reference mail server.

For the measurement of e-mail, services reference content should be used.

A reference e-mail shall have a body containing only the string "Test" and attachment(s) chosen from the following reference content building blocks with respect to expected data rates of the network under test.

Available building blocks are (files to be used as attachments): 100 kbyte, 200 kbyte, 500 kbyte, 1 Mbyte, 2 Mbyte, 5 Mbyte and 10 Mbyte.

NOTE – See also the table in Appendix VIII for typical upload or download times versus file size and used data rate.

It is recommended to use the binary files provided by ETSI[3]; they contain random data in order to exclude optimizer or accelerator effects.

A cycle should consist of mail upload using SMTP and mail download using IMAP or POP3. Both upload and download should represent typical user behaviour.

After a test cycle, all TCP connections to the server should be disconnected.

Testing of content integrity is mandatory for this test.

10.1.3.3 File transfer using FTP

FTP testing should take place against a reference FTP server. The server should support the standard FTP commands and both active and passive mode transfer of data. There should be no bandwidth limitation on application level.

In case of multisession scenarios, the reader shall be aware of the resulting effects with respect to the QoS parameters measured for each single session. With that, any use of multisession scenarios shall be stated in the results report.

In case of downloading a chunked single file via multiple data connections simultaneously, one shall be aware of the resulting effects with respect to the QoS parameters measured. With that, any use of

[2] Available at: http://portal.etsi.org/TBSiteMap/STQ/HTLMReferenceWebPage.aspx

[3] http://docbox.etsi.org/STQ/Open/TS%20102%20250-5%20Binary%20files

a simultaneous download of a chunked single file via multiple data connections as well as the number of chunks used to transfer the file during this session shall be stated in the results report.

10.1.3.4 File sharing using UDP

To be decided.

10.1.3.5 Synthetic tests

10.1.3.5.1 UDP

For further study.

10.1.3.5.2 ICMP

For further study.

10.1.3.5.3 TCP

For further study.

11 Post processing and statistical methods

11.1 Important measurement data types in mobile communications

Appropriate data analysis methods should depend on the type of the given data as well as on the scope of the analysis. Therefore, before analysis methods are described, different data types are introduced and differences between them are pointed out.

Four general categories of measurement results are expected when QoS measurements are done in mobile communications.

11.1.1 Data with binary values

Single measurements related to the topics:

- service accessibility, service availability;
- service retainability, service continuity;
- error ratios, error probabilities;

in general show a binary outcome, i.e., only two outcomes are possible. This means the result of a single trial leads to a result which is either valued positive or negative related to the considered objective. The result may be recorded as decision-results Yes/No or True/False or with numerical values 0 = successful and 1 = unsuccessful (i.e., errors occur) or vice versa. Aggregation of trials of both types allows to calculate related ratios which means the number of positive/negative results is divided by the number of all trials. Usually, the units of nominator and denominator are the same, namely number of trials.

Example: If established voice calls are considered to test the service retainability of a voice telephony system, every successfully completed call leads to the positive result "Call completed", every unsuccessfully ended call is noticed as "Dropped call" which represents the negative outcome. After 10 000 established calls, the ratio of dropped calls related to all established calls can be calculated. The result is the call drop probability.

11.1.2 Data out of time-interval measurements

Measurements related to the time domain occur in the areas:

- duration of a session or call;
- service access delay;

- round trip time and end-to-end delay of a service;
- blocking times, downtimes of a system.

The outcome of such measurements is the time span between two time stamps marking the starting and the end point of the time periods of interest. Results are related to the unit "second" or multiples or parts of it. Depending on the measurement tools and the precision needed, arbitrarily small measurement units may be realized.

Example: Someone can define the end-to-end delivery time for the MMS service by a measurement which starts when the user at the A party pushes the "send" button and which stops when the completely received MMS is signalled to the user at the B party.

11.1.3 Measurement of data throughput

Measurements related to data throughput result in values which describe the ratio of transmitted data volume related to the required portion of time. The outcome of a single measurement is the quotient of both measures. Used units are "bit" or multiples thereof for the data amount and "second" or multiples or parts thereof for the portion of time.

Example: If a data amount of 1 Mbit is transmitted within a period of 60 seconds, this would result in a mean data rate of approximately 16.66 kbit/s.

11.1.4 Data concerning quality measures

Examples are given by the quality of data transfer which may be measured by its speed or evaluations of speech quality measured on a scale, respectively.

Measurements related to audio-visual quality can be done objectively by algorithms or subjectively by human listeners. The outcome of audio-visual quality evaluation is related to a scaled value which is called mean opinion score (MOS) for subjective testing. Thereby, two types of quality measurement are distinguished; subjective and objective measurements. If quantitative measures are identified which are highly correlated to the quality of interest, this will simplify the analysis. However, if this is not possible, some kind of evaluation on a standardized scale by qualified experts is needed. The result may therefore be given either as the measurement result or as a mark on a pre-defined scale.

Example: Within a subjective test, people are asked to rate the overall quality of video samples which are presented to them. The allowed scale to rate the quality is defined in the range from 1 (very poor quality) to 5 (brilliant quality).

Table 11-1 summarizes the different kinds of QoS related measurements, typical outcomes and some examples.

Table 11-1 – QoS related measurements, typical outcomes and examples

Category	Relevant measurement types	Examples
Binary values	Service accessibility, service availability Service retainability, service continuity Error ratios, error probabilities	Service accessibility telephony, service non-availability SMS Call completion rate, call drop rate Call set-up error rate
Duration values	Duration of a session or call Service access delay Round trip time, end-to-end delay Blocking times, system downtimes	Mean call duration Service access delay WAP ICMP ping roundtrip time Blocking time telephony, SGSN downtime
Throughput values	Throughput	Mean data rate GPRS Peak data rate UMTS
Content quality values	Audio-visual quality	MOS scores out of subjective testing

11.2 Distributions and moments

11.2.1 Introduction

The objective of data analyses is to draw conclusions about the state of a process based on a given data set, which may or may not be a sample of the population of interest. If distributions are assumed, these specify the shape of the data mass up to the parameters associated with each family of distributions specifying properties like the mean of the data mass. Location or dispersion shifts of the process will in general result in different parameter estimates specifying the distribution. Therefore, the information available from the data is compressed into one or few sufficient statistics specifying the underlying distribution.

Many statistical applications and computations rely in some sense on distributional assumptions, which are not always explicitly stated. Results of statistical measures are often only sensible if underlying assumptions are met and therefore only interpretable if users know about these assumptions.

This clause is organized as follows. First, distributions, moments and quantiles are introduced in theory in clauses 11.2.2 to 11.2.4. This part of the Recommendation is based on the idea of random variables having certain distributions. Random variables do not take single values but describe the underlying probability model of a random process. They are commonly denoted by:

$$X \sim \text{distribution (parameters)}$$

From the distributional assumptions, moments and quantiles of random variables are derived in theory.

Data is often viewed as being the realizations of random variables. Therefore, data analysis mainly consists of fitting an appropriate distribution to the data and drawing conclusions based on this assumption. Clause 11.2.5 briefly summarizes the estimation of moments and quantiles.

Subsequently, a number of important distributions are introduced in clause 11.2.6, each of which is visualized graphically to give an idea of meaningful applications. Within this clause, testing distributions are also introduced as they are needed in clause 11.2.7 for the derivation of statistical tests.

11.2.2 Continuous and discrete distributions

The main difference between the data types described above can be explained in terms of continuous and discrete distributions. Data with binary values follow a discrete distribution, since the probability mass is distributed only over a fixed number of possible values. The same holds for quality measurements with evaluation results on a scale with a limited number of possible values (i.e., marks 1 to 6 or similar).

On the contrary, time-interval measurements as well as quality measurements based on appropriate quantitative variables may take an infinitely large number of possible values. In theory, since the number of possible outcomes equals infinity, the probability that a single value is exactly realized is zero. Probabilities greater than zero, are only realized for intervals with positive width. In practice, each measurement tool will only allow a limited precision resulting in discrete measurements with a large number of possible outcomes. Nevertheless, data from measurement systems with reasonable precision are treated as being continuous.

Formal definitions for continuous and discrete distributions are based on probability density functions as is described in the following.

11.2.3 Definition of density function and distribution function

11.2.3.1 Probability density function (PDF)

Probability density functions (PDFs) specify the probability mass either for single outcomes (discrete distributions) or for intervals (continuous distributions).

A PDF is defined as a function f: IR → [0, ∞] with properties:

i) $f(x) \geq 0$ for all $x \in S$.

ii) $\int_S f(x)dx = 1$ for continuous distributions or $\sum_S f(x) = 1$ for discrete distributions.

In other words, the values of the PDF are always non-negative, meaning that negative probabilities are neither assigned to values nor intervals, also the summation or integration over the PDF always results in 1 (= 100 per cent), meaning that any data value will always be realized.

Example 1: A PDF for binary data may be given by $f(x) = \begin{cases} 0{,}1: & x=1 \\ 0{,}9: & x=0 \end{cases}$, which implies that the probability for a faulty trial (x=1) is 10 per cent, while tests are completed successfully with probability 90 per cent.

Example 2: For time-interval measurements PDFs may take any kind of shape, as an example a normal distribution with mean 10 (seconds) is assumed here. The PDF for this distribution is given by $f(x) = \frac{1}{\sqrt{2\pi}} \exp\left\{-\frac{1}{2}(x-10)^2\right\}$. Other examples for continuous distributions will follow later on.

Example 3: If for instance categories for speech quality are defined as 1 = very poor up to 5 = brilliant, a PDF for the resulting data may be given by $f(x) = \begin{cases} 0{,}1 & : & x \in \{1,2,3\} \\ 0{,}4 & : & x = 4 \\ 0{,}3 & : & x = 5 \end{cases}$.

Figure 11-1 summarizes all three assumed example PDFs for different data types.

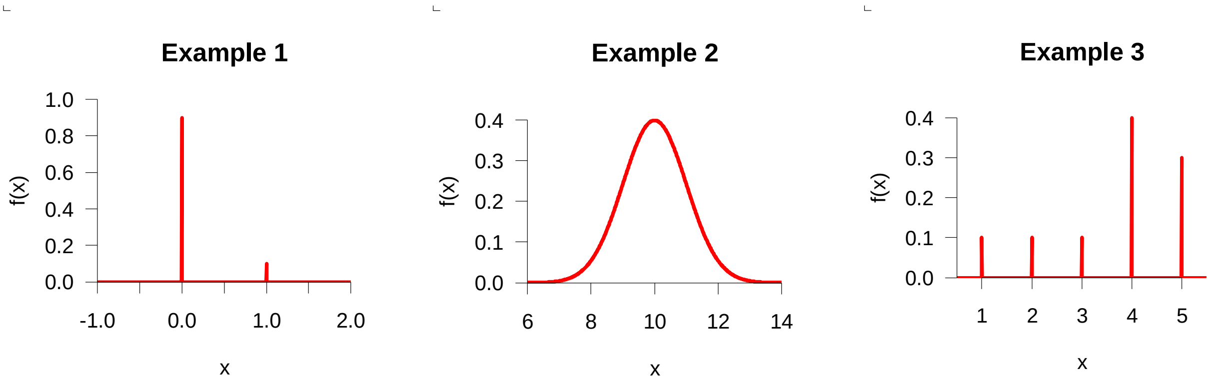


Figure 11-1 – Probability density functions (PDFs) of examples 1 to 3

11.2.3.2 Cumulative distribution function (CDF)

A cumulative distribution (or density) function (CDF) is computed from the corresponding PDF as described before by summing (discrete) or integrating (continuous) over the density mass up to the current value.

A function $F: IR \rightarrow [0,1]$ with $F(x) = \sum_{\tilde{x} \leq x} f(\tilde{x})$ for discrete and $F(x) = \int_{-\infty}^{x} f(\tilde{x})d\tilde{x}$ for continuous distributions is called CDF. This implies $F(x) \rightarrow 1$ for $x \rightarrow \infty$ and $F(x) \rightarrow 0$ for $x \rightarrow -\infty$.

In other words, the value of the CDF corresponds to the proportion of the distribution left of the value of interest. For the three examples from above, the CDFs are given in Figure 11-2.

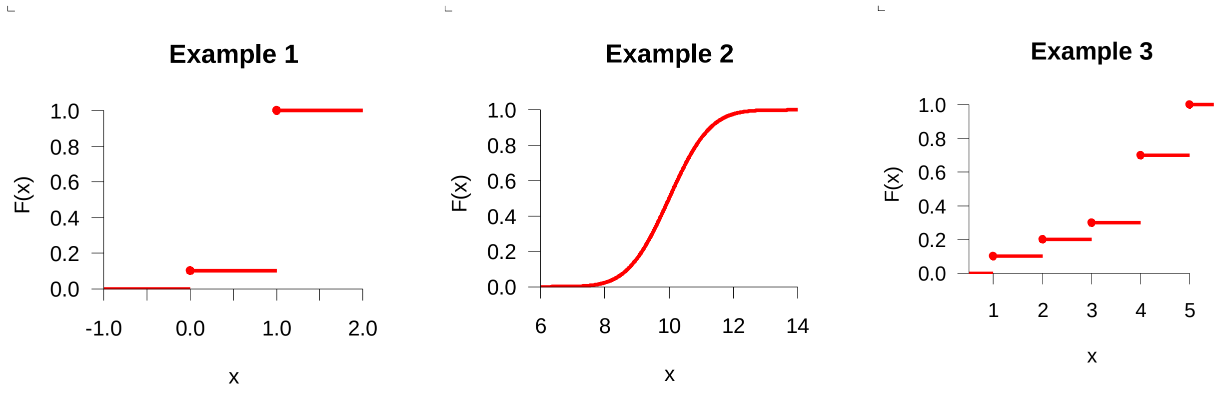


Figure 11-2 – Cumulative distribution functions (CDFs) of examples 1 to 3

11.2.4 Moments and quantiles

Moments are main characteristics of distributions. The most important moments are:

- the **expected value** (first moment), specifying the location of the distribution;
- the **variance** (second central moment), specifying the dispersion around the expected value of the distribution; and
- the **skewness** (third central moment), specifying whether a distribution is symmetric or skewed.

These moments are defined as follows.

a) The expected value (first moment, mean) of a random variable *x* with CDF *f(x)* is defined as $E(x)=\int x\cdot f(x)\,dx$ for continuous distributions or $E(x)=\sum x\cdot f(x)$ for discrete distributions, respectively.

b) The variance (second central moment) of a random variable *x* with CDF *f(x)* is defined as $Var(x)=\int (x-E(x))^2\cdot f(x)\,dx$ for continuous distributions or $Var(x)=\sum (x-E(x))^2\cdot f(x)$ for discrete distributions, respectively. The square root of the variance called standard deviation, denoted as $\sigma(x)$, is often more informative since it is defined on the original data scale.

c) The skewness (third central moment) of a random variable *x* with CDF *f(x)* is defined as $\int (x-E(x))^3\cdot f(x)\,dx$ for continuous distributions or $\sum (x-E(x))^3\cdot f(x)$ for discrete distributions, respectively. A value of zero indicates a symmetric distribution.

Example 1: For the CDF from example 1 the moments are given by $E(x)$ = 0.1×1+0.9×0 = 0.1, $Var(x)$ = 0.1×0.9^2 +0.9×0.1^2 = 0.09 resulting in a standard deviation $\sigma(x)$ = 0.3. The skewness can be computed as 0.1×0.9^3+0×9× (–0.1)^3 = 0.072 indicating that the distribution is not symmetric.

Example 2: The moments of the above normal distribution can be computed by partial integration and the fact that the PDF integrates to 1, or by utilizing the properties of normal distributions stating that the mean and standard deviation are the parameters μ and σ of the PDF $f(x\,|\,\mu,\sigma)=\frac{1}{\sigma\sqrt{2\pi}}\exp\left\{-\frac{1}{2\sigma^2}(x-10)^2\right\}$ and that normal distributions are always symmetric. This results in $E(x)$ = 10, $Var(x)$ = σ^2 = 1, which also equals the standard deviation and skewness = 0 for the above example.

Example 3: For the CDF of example 3, moments are computed by $E(x) = 0.1\times1+0.1\times2+0.1\times3+0.4\times4+0.3\times5 = 3.7$, $Var(x) = 1.61$ and negative skewness of –1.824.

The moments are computable for all three example PDFs. Nevertheless, they are not always meaningful. In particular in the third example, the possible outcomes are "very poor" to "brilliant", which may be ordered and named 1 to 5 as has been done before, but the expected value of 3.7 does not have a strict meaning. The same applies for higher moments, since the values of the variable of interest are not quantitative, but ordered qualitative.

In case of non-symmetric distributed data, moments may not be appropriate for describing the distribution of interest. An alternative measure of location is given by the **median**, which can be viewed as the point cutting the distribution into halves, namely 50 per cent of the distribution mass are smaller and 50 per cent are larger than the median.

More generally, quantiles are defined for each possible percentage. The α-**quantile** cuts the distribution in a part of α×100% of the distribution smaller than this value and (1-α)×100% larger than this value. The median as a special case is also called 50%-quantile.

A **formal definition of quantiles** is given in the following:

- "The α-quantile q_α with $\alpha \in [0, 1]$ is defined as the smallest number q_α satisfying $F(q_\alpha) \le \alpha$ (for α = 0, the minimum value with positive probability or –∞ is defined, respectively)".

Quantiles, shown in Figure 11-3, are easiest to illustrate and to compare with the examples of CDFs given above. For each CDF, the 5%, 50% and 75%-quantiles are added to the corresponding plot.

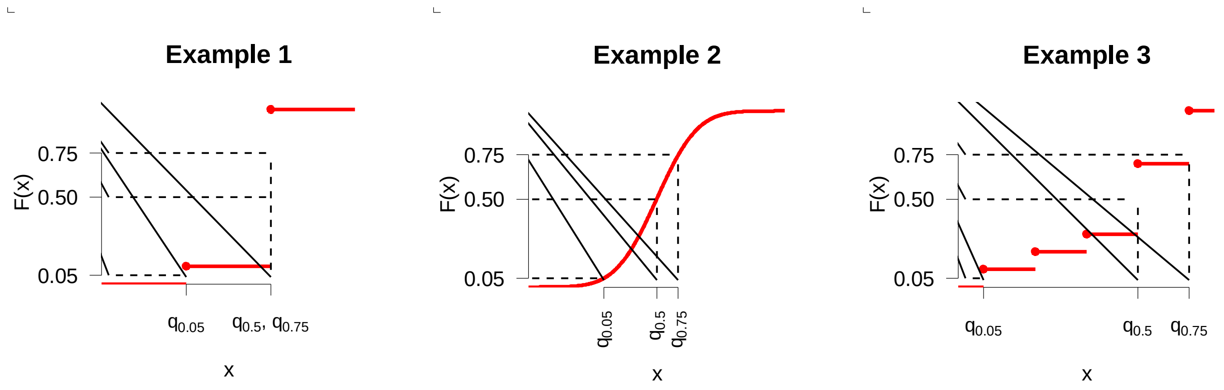


Figure 11-3 – Illustration of theoretical quantiles for examples 1 to 3

11.2.5 Estimation of moments and quantiles

If only samples from the population of interest are available, theoretical moments may not be computed, but have to be estimated empirically.

A sample-based estimator of the expectation of the underlying distribution is given by the **empirical mean** $\bar{x} = \frac{1}{n}\sum_{i=1}^{n} x_i$, where $x_i, i = 1, \ldots, n$ are the sample values. The **variance** of a distribution is commonly estimated by $s^2 = \frac{1}{n-1}\sum_{i=1}^{n}(x_i - \bar{x})^2$ with resulting **empirical standard deviation** $s = \sqrt{\frac{1}{n-1}\sum_{i=1}^{n}(x_i - \bar{x})^2}$.

For estimating quantiles, the above definition of theoretical quantiles is commonly replaced by a linear interpolating function. This function on one hand ensures that all quantiles are realized within the range of the empirical distribution (0%-quantile equals the minimum of the data, 100%-quantile equals the maximum of the data). The interpolation on the other hand allows a "better guess" of the real quantile if only few data are given and the underlying distribution is continuous. The commonly used computation formula is given by:

$$q_\alpha = (1-f)x_{(i)} + f \cdot x_{(i+1)}$$

where:

$$i = \lfloor 1+(n-1)\cdot\alpha \rfloor,\ f = 1+(n-1)\cdot\alpha - i \ \text{ and}$$

$$x_{(n+1)} := x_{(n)}.$$

Here $x_{(i)}$ denotes the i-th ordered data value and $\lfloor z \rfloor$ denotes the largest integer less or equal to z, i.e., $\lfloor 3{,}2 \rfloor = 3, \lfloor 4{,}9 \rfloor = 4$. Therefore, with the computation of i, the quantile is localized depending on the value of α between $x_{(i)}$ and $x_{(i+1)}$. The interpolation between these two values is done according to the deviation f between i and $(1 + (n\text{-}1)\times\alpha)$.

Examples of empirical CDFs and empirical quantiles for data simulated from the example distributions 1 to 3 are given in Figure 11-4. The solid black line represents the empirical quantiles derived by the above formula (from 0 to 100 per cent).

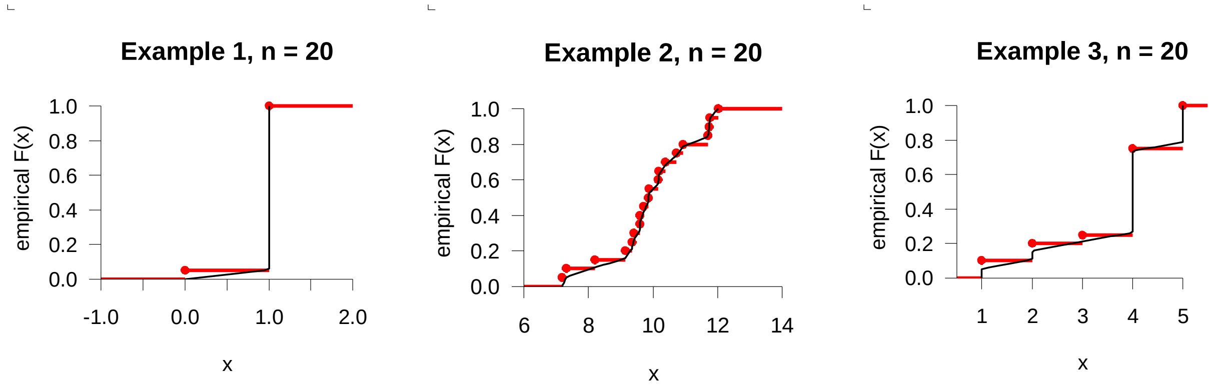


Figure 11-4a

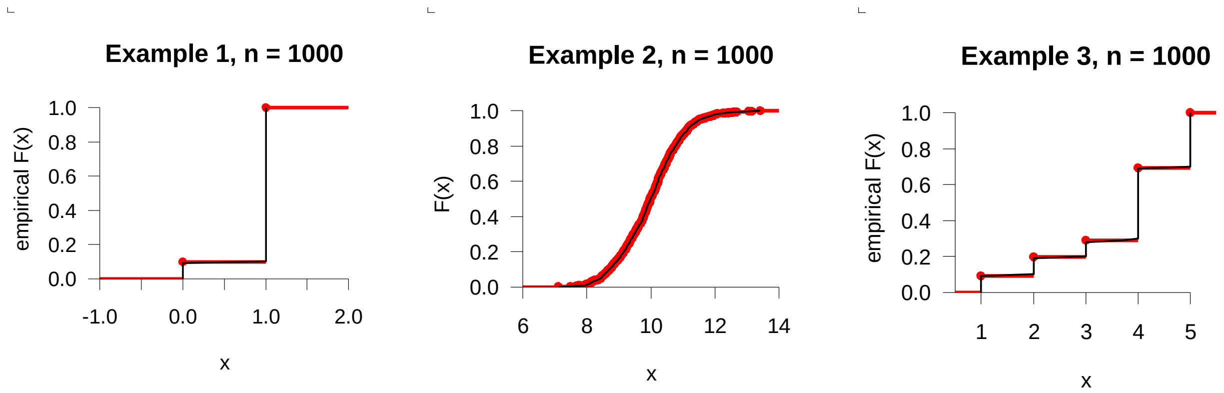


Figure 11-4b

Figure 11-4 – Illustration of empirical CDFs and quantiles for examples 1 to 3

Note that the above estimation procedure should be applied with great care for data sets with only few data values where the underlying distribution is presumably discrete, since the estimated quantiles also take values differing from those contained in the given data set. This can also be seen from Figure 11-4a in the plots for samples with sample size n = 20.

11.2.6 Important distributions

In this clause some of the important distributions related to practical usage in telecommunications are described. Either the mentioned distributions are directly related to measurement results or they are necessary to evaluate these results in a second step. Further relevant distributions may be appended later.

In general, distributions are specified by certain parameters which describe their main characteristics. Commonly, the characteristics are expressed in terms of their moments, i.e., mean value and standard deviation or variance, respectively. Wherever possible, the relevant characteristics are given as well as examples of possible use-cases. In general, continuous and discrete distributions are distinguished further on.

11.2.6.1 Continuous distributions

A large number of different continuous distributions are available to describe measurement results in a statistical manner. An overview is for instance given by [b-Law] or [b-Hartung]. For practical purposes in the field of quality of service (QoS) probing, the distributions described below are probably the most relevant ones.

11.2.6.1.1 Normal distribution

The normal distribution, also called Gaussian distribution (or bell-shaped distribution) is used for many natural processes whenever a symmetric continuous distribution seems appropriate. (An example was given before and density functions of further normal distributions are given in Figure 11-5.)

	Normal distribution
Notation	$X \sim N(\mu, \sigma^2)$
Parameters	μ, σ
PDF	$f(x) = \frac{1}{\sigma\sqrt{2\pi}} \exp\left\{-\frac{1}{2\sigma^2}(x-\mu)^2\right\}$
CDF	$F(x) = \int_{-\infty}^{x} \frac{1}{\sigma\sqrt{2\pi}} \exp\left\{-\frac{1}{2\sigma^2}(t-\mu)^2\right\} dt$
Expected value	$E(X) = \mu$
Variance	$Var(X) = \sigma^2$
Remarks	Standard normal distribution with $\mu = 0$ and $\sigma = 1$, see clause 11.2.6.1.1.1

The normal distribution is uniquely specified by its mean and standard deviation. For normally distributed data, about 68 per cent of the data are realized within the interval [μ – σ, μ + σ], 95 per cent are realized within [μ – 2σ, μ + 2σ] and 99.7 per cent are realized within [μ – 3σ, μ + 3σ]. The last interval is also called 6σ-interval which gave the name to the popular "Six-sigma"-courses.

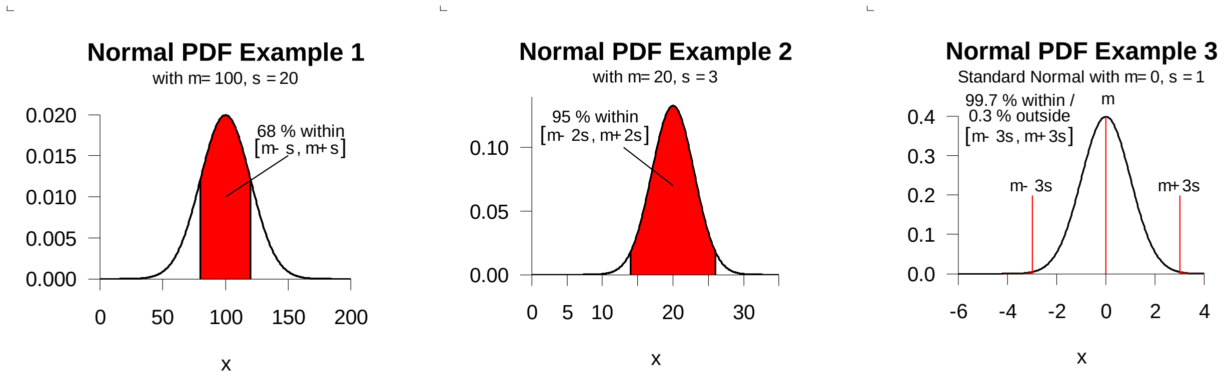


Figure 11-5 – Density functions of three different normal distributions

Normally (or nearly normally) distributed data is found quite often in practice, in particular in nature, for example human or animal body heights.

11.2.6.1.1.1 Standard normal distribution

All normal distributions or normally distributed data can be standardized by subtracting the mean and afterwards dividing by the standard deviation of the distribution or data resulting in a standard normal distribution with mean $\mu = 0$ and standard deviation $\sigma = 1$. The inverse computation leads back to the original distribution or data. Therefore, all normal distributions may be reduced to the standard normal, if the parameters μ and σ are known or estimated. Because of this and the fact that many statistical tests are based on the normal distribution, statistical textbooks often provide the quantiles of the standard normal distribution. In particular, the α-quantile of the standard normal distribution is denoted as u_α .

In example 3 of Figure 11-5, the density of the standard normal distribution is given.

	Standard normal distribution
Notation	$X \sim N(0,1)$
Parameters	none
PDF	$f(x) = \frac{1}{\sqrt{2\pi}} \exp\left\{-\frac{1}{2} x^2\right\}$
CDF	$F(x) = \int_{-\infty}^{x} \frac{1}{\sqrt{2\pi}} \exp\left\{-\frac{1}{2} t^2\right\} dt$
Expected value	$E(X) = 0$
Variance	$Var(X) = 1$
Remarks	

11.2.6.1.1.2 Central limit theorem

Another reason for the frequent use of normal distributions (in particular for testing purposes) is given by the central limit theorem, one of the most important theorems in statistical theory. It states that the mean of n equally distributed random variables with mean μ and variance σ^2 approaches a normal distribution with mean μ and variance σ^2/n as n becomes larger. This holds for arbitrary distributions and commonly the typical shape of the normal distribution is sufficiently reached for $n \geq 4$. For further details about the central limit theorem see [b-Law] or [b-Mood] (see bibliography).

A number of tools was developed for checking whether data (or means) are normal, namely test procedures like the well-known Kolmogorov-Smirnov goodness-of-fit test (see clause 11.2.6.6.1.2) among others or graphical tools like histograms or QQ-plots. The mentioned graphical tools will be introduced in clause 11.3.

11.2.6.1.1.3 Transformation to normality

As has been seen, the normal distribution is very powerful and can be applied in many situations. Nevertheless, it is not always appropriate, in particular in technical applications, where many parameters of interest have non-symmetric distributions. However, in these situations it may be possible to transform the data to normality. This idea leads for instance to the Log-Normal distribution, which is often assumed for technical parameters.

11.2.6.1.2 Log-Normal distribution

The distribution of a random variable is said to be log-normal, if the logged random variable is normally distributed, which is denoted by *log(x) ~ N* (μ, σ^2).

	Log-normal distribution
Notation	$X \sim LN(\mu,\sigma^2)$ or $\log(X) \sim N(\mu,\sigma^2)$
Parameters	μ,σ
PDF	$f_X(x) = \begin{cases} \frac{1}{\sigma x\sqrt{2\pi}} \exp\left\{-\frac{(\ln(x)-\mu)^2}{2\sigma^2}\right\} & \text{if } x>0 \\ 0 \text{ else} \end{cases}$
CDF	$F(x) = \int_{-\infty}^{x} \frac{1}{\sigma\sqrt{2\pi}} \exp\left\{-\frac{1}{2\sigma^2}(t-\mu)^2\right\} dt$
Expected value	$E(x) = \exp(\mu + \frac{1}{2}\sigma^2)$
Variance	$Var(x) = \exp(2\mu+\sigma^2)(\exp(\sigma^2)-1)$
Remarks	

Log-normal distributions are skewed and have heavier upper tails compared to the normal distribution implying a higher variability in the upper quantiles. Density examples for different values of μ and σ are given in Figure 11-6 and illustrate that the log-normal distribution can take a variety of different shapes.

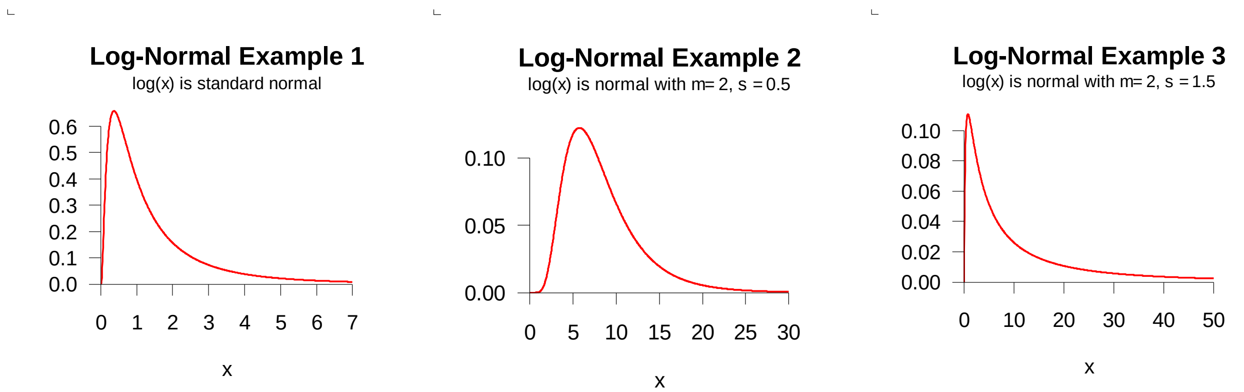


Figure 11-6 – Density functions of log-normal distributions

11.2.6.1.2.1 Use-case: transformations

A given data set can be checked whether it is distributed according to a log-normal distribution by computing the log of the data values and using one of the graphical tools mentioned before for verifying the normal distribution for the logged data. Empirical mean and standard deviation of the transformed data can then be used for estimating the parameters of the distribution, respectively.

Similarly, other transformation-based distributions can be derived from the normal distribution, for instance for the square-root transformation $\sqrt{x} \sim IN(\mu, \sigma^2)$ or the reciprocal transformation $1/x \sim IN(\mu, \sigma^2)$. A general concept based on power-transformations of x was proposed in [b-Box].

11.2.6.1.3 Exponential distribution

For modelling arrival processes, often the negative exponential distribution is used. The relevant parameter for this distribution is λ which symbolizes the life cycle of a process. Concerning arrival processes, λ is named the inter-arrival rate of succeeding events.

	Exponential distribution
Notation	$X \sim \mathrm{Exp}(\lambda)$
Parameters	$\lambda > 0$
PDF	$f(x) = \lambda \exp(-\lambda x)$ if $x \geq 0$
CDF	$F(x) = 1 - \exp(-\lambda x)$ if $x \geq 0$
Expected value	$E\{X\} = \frac{1}{\lambda}$
Variance	$Var\{X\} = \frac{1}{\lambda^2}$
Remarks	Life-cycle description, survival function: Survival probability $P(X > x) = \exp(-\lambda x)$

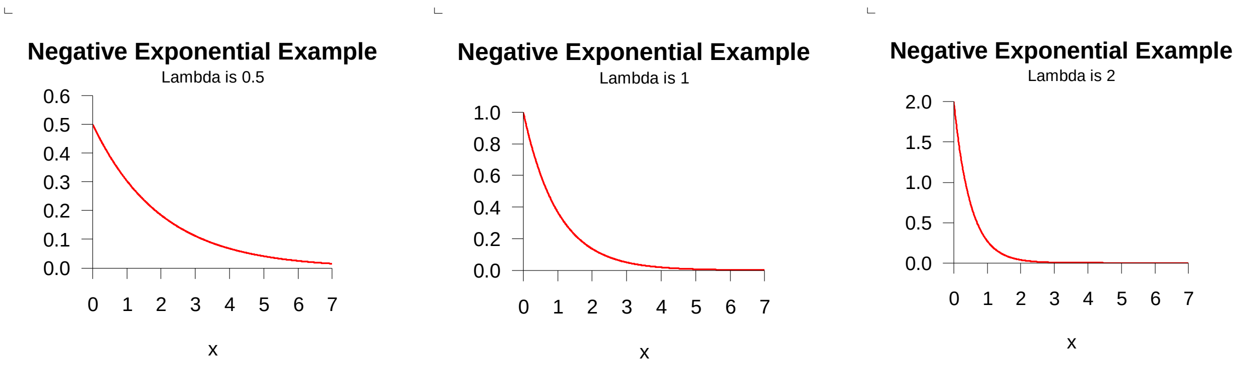


Figure 11-7 – Density functions of negative exponential distributions

11.2.6.1.4 Weibull distribution

The Weibull distribution is a heavy-tailed distribution which means the distribution is skewed with a non-negligible part of the probability mass in the tail. This distribution can be used to describe processes which have a rare frequency, but which are not negligible due to their weight.

	Weibull distribution
Notation	$X \sim Weibull(\alpha, \beta)$

	Weibull distribution
Parameters	α with $\alpha \geq 0$, β with $\beta > 0$
PDF	$f_x(x) = \alpha\beta x^{\beta-1} \exp(-\alpha x^{\beta})$ if $x > 0$
CDF	$F_x(x) = 1 - \exp\left(-\alpha x^{\beta}\right)$ if $x \geq 0$
Expected value	$E\{X\} = \alpha^{-\frac{1}{\beta}} \Gamma\left(\frac{1}{\beta} + 1\right)$ with Γ Gamma function
Variance	$Var\{X\} = \alpha^{-\frac{2}{\beta}} \left[\Gamma\left(\frac{2}{\beta} + 1\right) - \left(\Gamma\left(\frac{1}{\beta} + 1\right)\right)^2\right]$ with Γ Gamma function
Remarks	Fatigue of material $Weibull(2, \beta)$ is a Rayleigh distribution with parameter β. Rayleigh is used for description of fading effects.

The Gamma function is defined as the integral function $\Gamma(x) = \int_0^{\infty} \exp(-t)\, t^{x-1}\, dt$. One important relationship for the Gamma function is given by $\Gamma(x+1) = x \cdot \Gamma(x)$. For integer values n this relation transforms into $\Gamma(n) = (n-1)!$

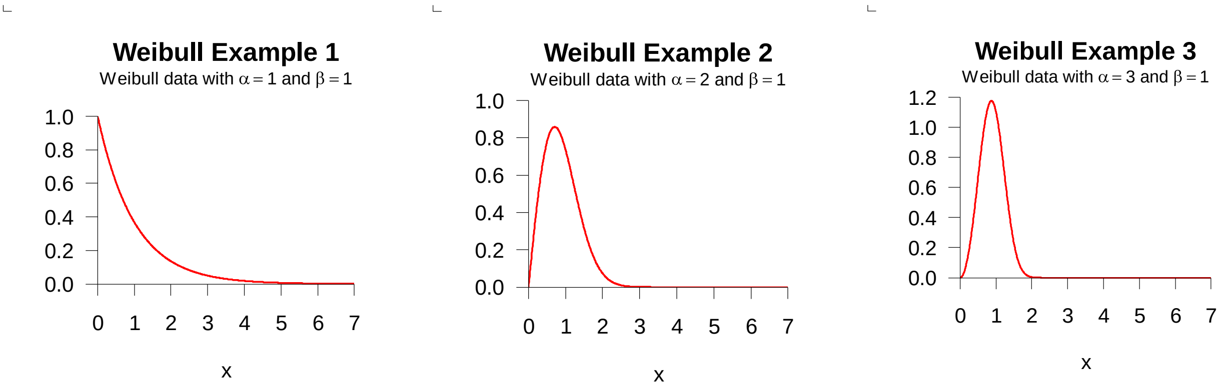


Figure 11-8 – Density functions of Weibull distributions

11.2.6.1.5 Pareto distribution

The Pareto function also models a heavy tailed distribution. One common use-case of this distribution is the modelling of packet-oriented data traffic. For example, the size of HTTP requests and replies as well as FTP downloads can be described as a Pareto function.

Figure 11-9 shows density functions of Pareto distributions.

	Pareto distribution
Notation	$X \sim Pareto(c, \alpha)$
Parameters	c scale and location parameter α shape parameter
PDF	$f(x) = \alpha \cdot x^{-(\alpha+1)} \cdot c^{\alpha}$ for $x > c$
CDF	$F(x) = 1 - \left(\frac{c}{x}\right)$
Expected value	$E\{X\} = \frac{c}{\alpha - 1}$ for $\alpha > 1$
Variance	$Var\{X\} = \frac{c\alpha}{(\alpha-1)^2 \cdot (\alpha - 2)}$ for $\alpha > 2$
Remarks	

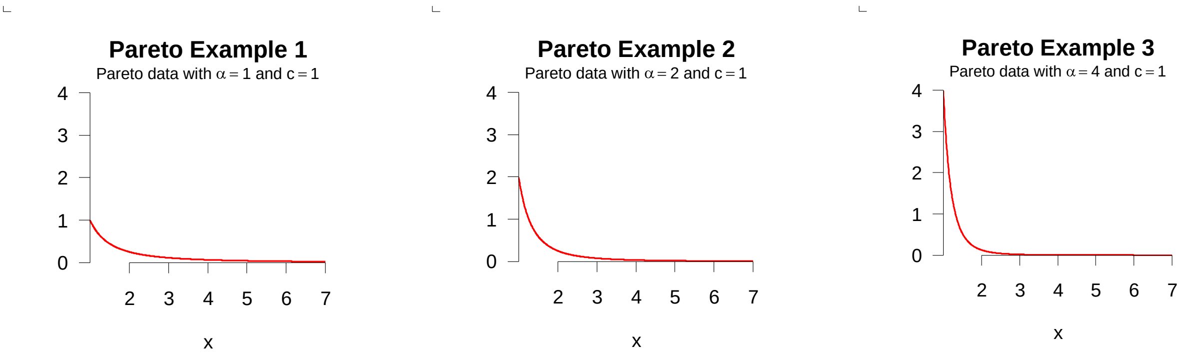


Figure 11-9 – Density functions of Pareto distributions

11.2.6.1.6 Extreme distribution (Fisher-Tippett distribution)

For modelling extremely seldom events with a high and negligible influence, the extreme distribution may be appropriate.

Example 1: In service probing, this distribution for example relates to the amount of data which is transferred via FTP data connections. Whereas most of the users generate traffic in the range of some ten or hundred megabytes, occasionally there may be users which like to transfer, for example, 10 gigabytes in one session. When modelling the overall FTP data traffic, these users cannot be neglected due to their immense data volume, but their occurrence probability is very low.

Example 2: Concerning insurance cases, single incidents which require a very high financial effort arise when for example an explosion eliminates a complete factory building. Again, due to the high financial impact, these cases have to be taken into account even though that they occur rarely.

Figure 11-10 shows density functions of extreme distributions.

	Extreme distribution
Notation	$X \sim Extreme(\alpha, \beta)$
Parameters	α shape parameter β scale parameter
PDF	$f(x) = \frac{1}{\beta} \cdot \exp\left(-\frac{x-\alpha}{\beta}\right) \cdot \exp\left[-\exp\left(-\frac{x-\alpha}{\beta}\right)\right]$
CDF	$F(x) = \exp\left[-\exp\left(-\frac{x-\alpha}{\beta}\right)\right]$
Expected value	$E\{x\} = \alpha + \beta\gamma$ with $\gamma \approx 0{,}57721566$ constant of Euler-Mascheroni
Variance	$Var\{x\} = \frac{\Pi^2 \beta^2}{6}$ for $\alpha > 2$
Remarks	

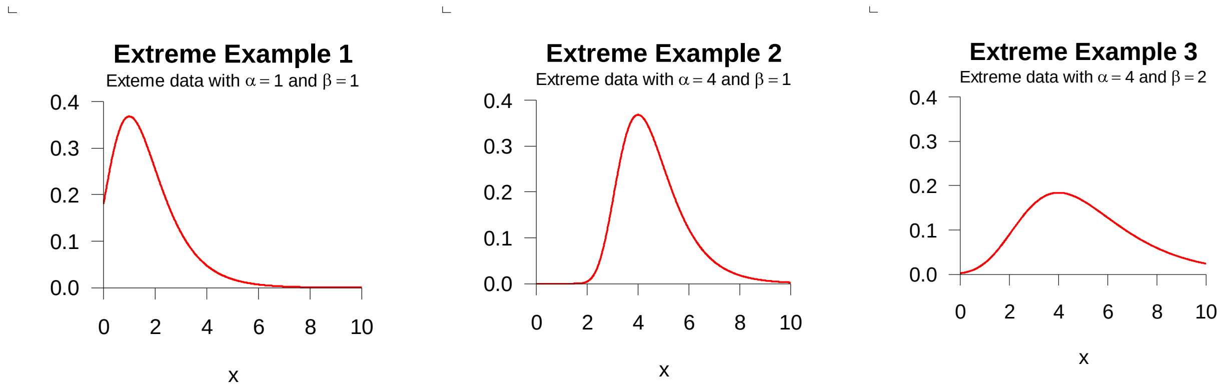


Figure 11-10 – Density functions of extreme distributions

11.2.6.2 Testing distributions

Statistical tests are commonly applied to reject an assumption in favour of an alternative assumption. Therefore, most tests are based on some kind of measure of deviation. This may be the deviation of data from a model assumption or from an assumed mean value, a target value and so on. For computational ease, single deviations are often assumed to be normally distributed.

Based on these concepts, three important testing distributions are introduced in the following, namely the Chi-square-, F- and Student t-distributions.

11.2.6.2.1 Chi-square distribution with *n* degrees of freedom

If the result of a service probing is assumed to be the result of a number of independent standard normal processes, this distribution provides a basis for testing against this assumption. For evaluation purposes concerning the χ^2 distribution, see clause 11.2.6.4.

A χ^2 distribution represents a combination of n independent random variables $Z_1, ..., Z_n$ where each random variable is standard normal, i.e., $Z_i \sim N(0,1)$. The combination is done according to:

$$\sum_{i=1}^{n} Z_i^2 \sim \chi_n^2$$

The result of this combination is called a "(central) χ^2 distribution with n degrees of freedom".

Figure 11-11 shows density functions of Chi-square distributions.

	(Central) Chi-square distribution
Notation	$X \sim \chi_n^2$ Random variable $X = \sum_{i=1}^{n} Z_i^2$
Parameters	n : degrees of freedom $Z_1, ..., Z_n$: independent standard normal random variables: $Z_i \sim N(0,1)$
PDF	$f(x) = \frac{1}{2^{\frac{n}{2}} \cdot \Gamma\left(\frac{n}{2}\right)} \cdot x^{\frac{n}{2}-1} \cdot \exp\left(-\frac{x}{2}\right)$ for $x > 0$
CDF	$F(x) = \int_{-\infty}^{x} f(\xi)\ d\xi$ No closed solution available
Expected value	$E\{X\} = n$
Variance	$Var\{X\} = 2n$
Remarks	Combination of n statistically independent $N(0,1)$ random variables (standard normal) Approximation: $F(x) = P(X \le x) \cong \Phi\left(\frac{x-n}{\sqrt{2n}}\right)$

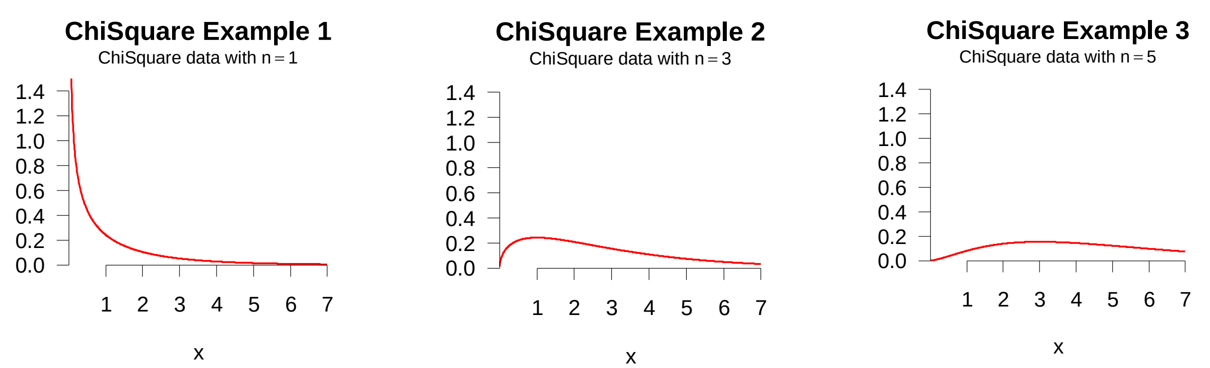


Figure 11-11 – Density functions of Chi-square distributions

11.2.6.2.1.1 Further relations

The referenced gamma function is defined as the integral function:

$$\Gamma(x) = \int_0^{\infty} \exp(-t) \cdot t^{x-1} dt$$

Additional useful relations according to this function are:

$$\Gamma(x+1) = x \cdot \Gamma(x)$$

and

$$\Gamma(n) = (n-1)! \text{ if } x = n \text{ is an integer value}$$

11.2.6.2.1.2 Relation to empirical variance

- If the mean value μ is known, the empirical variance of n normally distributed random variables reads $s_\mu^2 = \frac{1}{n} \cdot \sum_{i=1}^{n} (x_i - \mu)^2$. With this piece of information, a chi-square distribution is given for the following expression: $n \cdot \frac{s_\mu^2}{\sigma^2} \sim \chi_n^2$.
- Without knowledge of μ, the empirical variance $s^2 = \frac{1}{n-1} \cdot \sum_{i=1}^{n} (x_i - \overline{x})^2$ estimates the variance of the process. The appropriate relation in this case reads $(n-1) \cdot \frac{s^2}{\sigma^2} \sim \chi_{n-1}^2$.

11.2.6.2.2 Student t-distribution

If a standard normal and a statistically independent chi-square distribution with n degrees of freedom are combined, according to $X = \frac{U}{\sqrt{Z/n}}$, where $Z \sim \chi^2$ (chi-square distributed) and U ~ $N(0,1)$ (standard normal distributed), the constructed random variable X is said to be t-distributed with n degrees of freedom. Alternatively, the denomination "Student t-distribution" can be used.

Figure 11-12 shows density functions of student t-distributions.

	Student t-distribution
Notation	$X \sim t_n$ Random variable $X = \frac{U}{\sqrt{Z/n}}$ with $U \sim N(0,1)$, $Z \sim \chi_n^2$, independent.
Parameters	n : degrees of freedom
PDF	$f(x) = \frac{\Gamma\left(\frac{n+1}{2}\right)}{\Gamma\left(\frac{n}{2}\right) \cdot \sqrt{\Pi \cdot n}} \cdot \left(1 + \frac{x^2}{n}\right)^{-\frac{n+1}{2}}$
CDF	$F(x) = \int_{-\infty}^{x} f(\xi)\ d\xi$ No closed solution available
Expected value	$n \geq 2:\ E\{Z\} = 0$

Variance	$n \geq 3: \; Var\{Z\} = \frac{n}{n-2}$
Remarks	The PDF is a symmetric function with symmetry axis $x = 0$. Additional relation for α-quantiles $t_{n;\alpha}$: $t_{n;\alpha} = -t_{n;1-\alpha}$

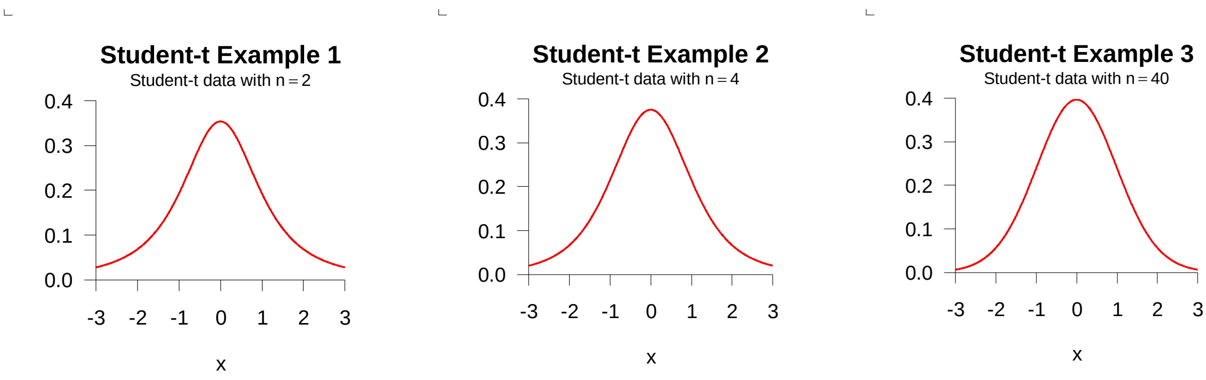


Figure 11-12 – Density functions of student t-distributions

11.2.6.2.2.1 Relation to normal distribution

It may not be obvious, but *t*-distributions with large number of degrees of freedom may be approximated by a standard normal distribution.

- The standardization of normal variables was covered before: If $X \sim N(\mu, \sigma^2)$, then $(X-\mu)/\sigma \sim N(0, 1)$.
- Consider the case of data assumed to be normal with unknown variance. As stated before, the empirical variance is then related to a chi-square distribution. The empirical mean and variance of *n* normally distributed ($N(\mu, \sigma^2)$) random variables $X_1, X_2, ..., X_n$ are given by:

$$\bar{X} = \frac{1}{n} \cdot \sum_{i=1}^{n} X_i$$

$$S^2 = \frac{1}{n-1} \cdot \sum_{i=1}^{n} \left(X_i - \bar{X} \right)^2$$

With these relations, the relation between the t-distribution and the *n* normal distributed random variables is given by:

$$\sqrt{n} \cdot \frac{\bar{X} - \mu}{\sqrt{S^2}} \sim t_{n-1}.$$

11.2.6.2.3 F distribution

The F distribution is a combination of m standard normal distributed random variables Y_i and n standard normal distributed random variables V_i which are combined as described below. Again, m and n are called "degrees of freedom" of this distribution.

This distribution is often used for computation and evaluation purposes, for example in relation with confidence intervals for the binomial distribution (Pearson-Clopper formula). In general, it compares two types of deviations, for instance if two different models are fitted.

Figure 11-13 shows density functions of F distributions.

	F distribution
Notation	$X \sim F_{m,n}$ Random variable $X = \dfrac{\frac{1}{m} \cdot \sum_{i=1}^{m} Y_i^2}{\frac{1}{n} \cdot \sum_{i=1}^{n} V_i^2}$
Parameters	m,n : degrees of freedom $Y_1,\ldots,Y_n$: independent random variables according to $N(0,1)$ $V_1,\ldots,V_n$: independent random variables according to $N(0,1)$
PDF	$f(x) = \dfrac{\left(\frac{m}{n}\right)^{\frac{m}{2}} \cdot x^{\frac{m}{2}-1}}{B\left(\frac{m}{2},\frac{n}{2}\right)} \cdot \left(1+\frac{m}{n} \cdot x\right)^{-\frac{m+n}{2}}$ for $x > 0$ with $B(p,q) = \dfrac{\Gamma(p) \cdot \Gamma(q)}{\Gamma(p+q)}$ Eularian beta function
CDF	$F(x) = \int_{-\infty}^{x} f(\xi)\ d\xi$ No closed solution available
Expected value	$n > 2:\ E\{Z\} = \dfrac{n}{n-2}$
Variance	$n > 4:\ Var\{Z\} = \dfrac{2n^2(m+m-2)}{m(n-2)^2(n-4)}$
Remarks	A $F_{m,n}$ related distribution can be interpreted as the quotient of a χ_m^2 distribution and a χ_n^2 distribution multiplied with $\frac{n}{m}$.

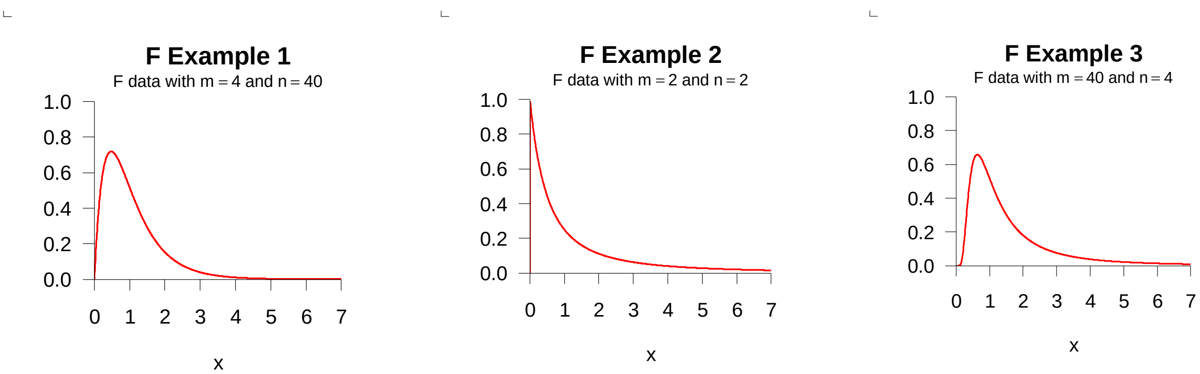


Figure 11-13 – Density functions of F distributions

11.2.6.2.3.1 Quantiles

For quantile computation purposes, the following relations may be useful:

$$F_{n_1,n_2;1-\gamma} = \frac{1}{F_{n_2,n_1;\gamma}}$$

In general, quantile values of this distribution are tabulated.

11.2.6.2.3.2 Approximation of quantiles

If the desired quantile value cannot be found in tables, the following approximation may be helpful:

If the γ -quantile is wanted with γ in the range of $0.5 < \gamma < 1$, the relation

$$F_{n_1,n_2;\gamma} \cong \exp(u \cdot a - b)$$

applies where $u = u_\gamma$ is the γ -quantile of the standard normal distribution $N(0,1)$.

The symbols a and b are derived from the following equations:

$$a = \sqrt{2d + cd^2}$$

$$b = 2 \cdot \left(\frac{1}{n_1 - 1} - \frac{1}{n_2 - 1} \right) \cdot \left(c + \frac{5}{6} - \frac{d}{3} \right)$$

$$c = \frac{(u_\gamma)^2 - 3}{6}$$

$$d = \frac{1}{n_1 - 1} + \frac{1}{n_2 - 1}$$

11.2.6.2.3.3 Relations to other distributions

When the F distribution comes to usage, the following relations may ease the handling of this distribution:

- Relation to t distribution for $n_1 = 1$: $F_{1,n_2;\gamma} = \left(t_{n_2;\frac{1+\gamma}{2}} \right)^2$.
- Relation to χ^2 distribution for $n_2 \to \infty$: $F_{n_1,\infty;\gamma} = \frac{1}{n_1} \cdot \chi_{n_1;\gamma}{}^2$.
- If $n_1 \to \infty$ and $n_2 \to \infty$, the distribution simplifies to: $F_{\infty,\infty;\gamma} = 1$

11.2.6.3 Discrete distributions

Discrete distributions describe situations where the outcome of measurements is restricted to integer values. For example, the results of service access tests show either that service access is possible (mostly represented by a logical "1" value) or that it is not possible (mostly represented by a logical "0" value). Depending on the circumstances under which such "drawing a ball out of a box" tests are executed, different statistical distributions apply as shown in clauses 11.2.6.3.1 to 11.2.6.3.4.

11.2.6.3.1 Bernoulli distribution

The starting point of different discrete distributions is given by the Bernoulli distribution. It simply describes the probability p of a positive outcome of a single test where only two states are allowed, generally a positive one and a negative one. As soon as more than one single test is executed, further discrete distribution may be applied as shown in the following clauses.

Figure 11-14 shows Density functions for Bernoulli distributions.

	Bernoulli distribution
Notation	$X \sim Bernoulli(p)$
Parameters	$p \in (0,1)$
PDF	$p(x) = \begin{cases} 1-p & \text{if } x = 0 \\ p & \text{if } x = 1 \\ 0 & \text{otherwise} \end{cases}$
CDF	$F(x) = \begin{cases} 0 & \text{if } x < 0 \\ 1-p & \text{if } 0 \le x < 1 \\ 1 & \text{if } 1 \le x \end{cases}$
Expected value	$E\{X\} = p$
Variance	$Var\{X\} = p \cdot (1-p)$
Remarks	

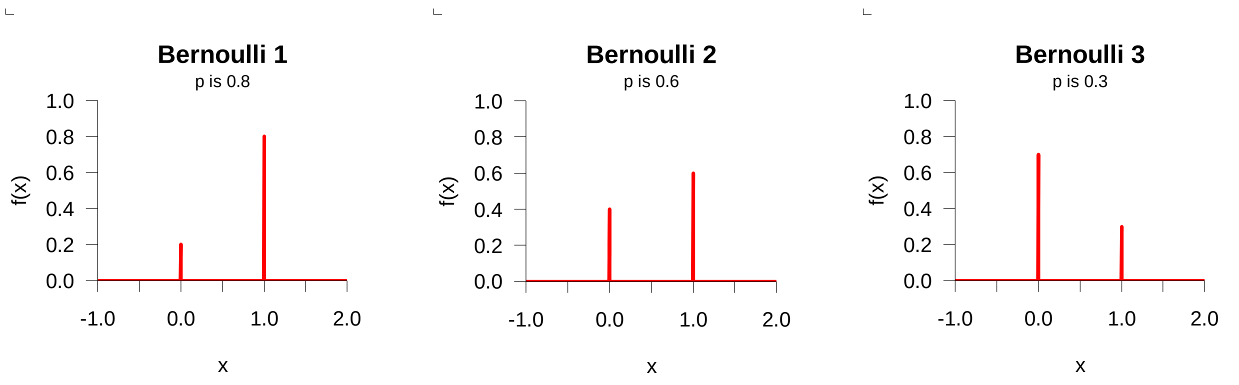


Figure 11-14 – Density functions for Bernoulli distributions

11.2.6.3.2 Binomial distribution

Whenever the outcome of a test is either true or false, the binomial distribution can be applied. Where a "black or white" interpretation of results is appropriate, this distribution is able to describe the measurement process. Due to this "yes or no" character, the binomial distribution can be interpreted as the result of different Bernoulli tries. Relevant examples related to service probing are service access issues (e.g., call success rate, SMS send failure ratio, etc.). For a high number of measurement results, the distribution can be replaced by the normal distribution as a first approximation as shown in clause 11.2.6.1.1.

Precondition: To determine the CDF of a binomial distribution with relation to different tests, the single events have to be independent from each other. This means that the probability of a successful outcome of different consecutive tests **must not change**. In consequence, this means a memory-less process where the result of a succeeding test is not related to the outcome of its predecessor(s).

Figure 11-15 shows density functions of binomial distributions.

	Binomial distribution
Notation	$X \sim Bin(n, p)$
Parameters	n Number of tests m Number of successful test outcomes $p = \frac{m}{n}$ Observed probability of successful outcomes $q = 1 - p$ Observed probability of unsuccessful outcomes
PDF	$P(X = k) = \binom{n}{k} p^k (1-p)^{n-k} = b(n, p, k)$ with $k = 0, 1, 2, \ldots, n$
CDF	$P(X \le k_0) = \sum_{k=0}^{k_0} \binom{n}{k} p^k (1-p)^{n-k}$ with $k = 0, 1, 2, \ldots, n$ and $k_0 \ge k$
Expected value	$E\{X\} = n \cdot p$
Variance	$Var\{X\} = n \cdot p \cdot q = n \cdot p \cdot (1-p)$
Remarks	Related to F distribution

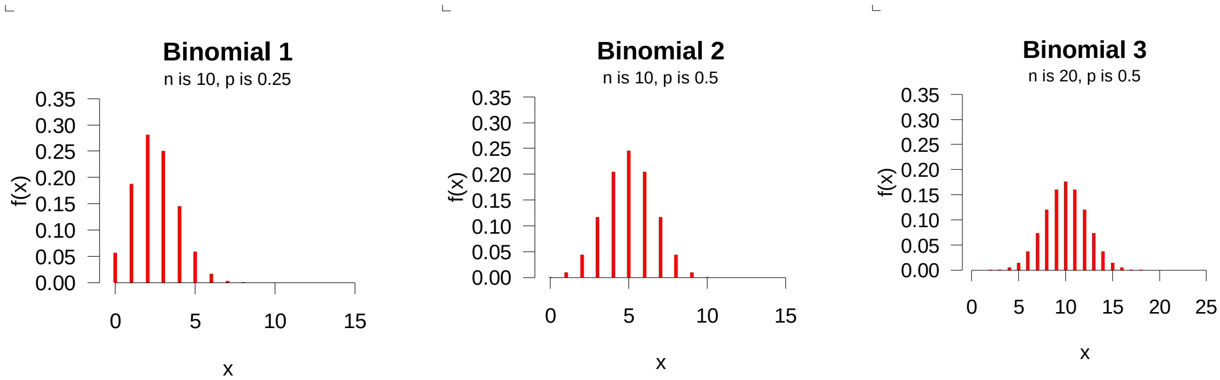


Figure 11-15 – Density functions of binomial distributions

For computation purposes, the following relation between the binomial distribution and the F distribution may be useful:

$$P(X < x) = 1 - P\left(F \le \frac{n-x}{x+1} \cdot \frac{p}{1-p}\right)$$

In this formula, F represents a F distributed random variable with 2 × (x+1), 2 × (n-x), degrees of freedom.

11.2.6.3.3 Geometric distribution

The geometric distribution typically describes the following situation: A number of Bernoulli trials are executed consecutively. Each of these trials has a success probability p. By use of the geometrical distribution, one can determine the probability of a successful outcome of a Bernoulli trial after x unsuccessful outcomes.

Scenarios where this computation may be of interest are for example the occurrence of the first success after x failures, or related to service probing, the number of failed service access attempts before the first successful attempt.

Figure 11-16 shows density functions of geometric distributions.

	Geometric distribution
Notation	$X \sim G(p)$
Parameters	$p \in (0,1)$
PDF	$p(x) = \begin{cases} p(1-p)^x & \text{if } x \in \{0, 1, \ldots\} \\ 0 & \text{otherwise} \end{cases}$
CDF	$F(x) = \begin{cases} 1-(1-p)^{\lfloor x \rfloor + 1} & \text{if } x \geq 0 \\ 0 & \text{otherwise} \end{cases}$
Expected value	$E\{X\} = \frac{1-p}{p}$
Variance	$Var\{X\} = \frac{1-p}{p^2}$
Remarks	

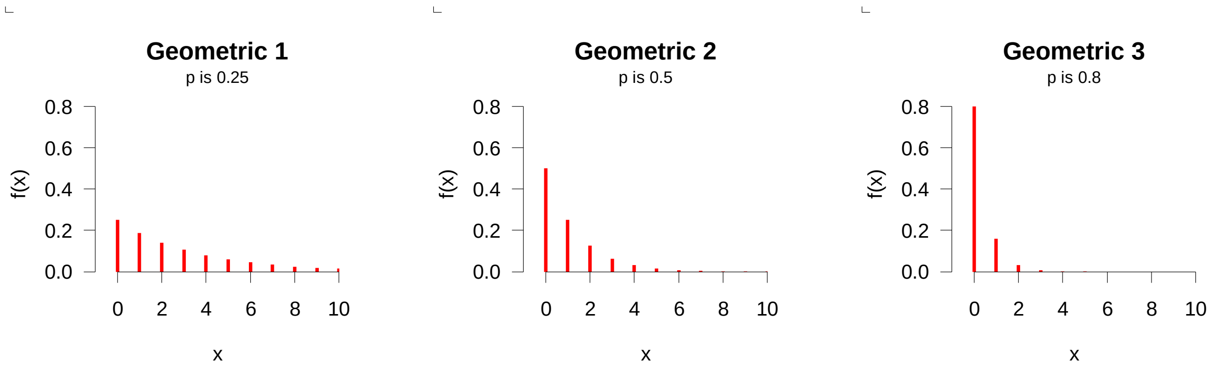


Figure 11-16 – Density functions of geometric distributions

11.2.6.3.4 Poisson distribution

The Poisson distribution is also called "distribution of rare events". Generally, this distribution relates to the number of events within a certain time of period under the precondition that the events occur at a constant rate λ. The Poisson distribution often is used to describe call arrivals in a transmission system, especially the current number of processed service attempts in a system.

Figure 11-17 shows density functions of Poisson distributions.

	Poisson distribution
Notation	$X \sim Po(\lambda)$
Parameters	λ
PDF	$P(X=k)=\frac{\lambda^k}{k!}\exp(-\lambda)$ with $k=0,1,2,\ldots,n$
CDF	$P(X \le k)=\sum_{i=0}^{k}\frac{\lambda^k}{i!}\exp(-\lambda)$ with $k=0,1,2,\ldots,n$
Expected value	$E\{X\}=\lambda$
Variance	$Var\{X\}=\lambda$
Remarks	Related to χ^2 distribution

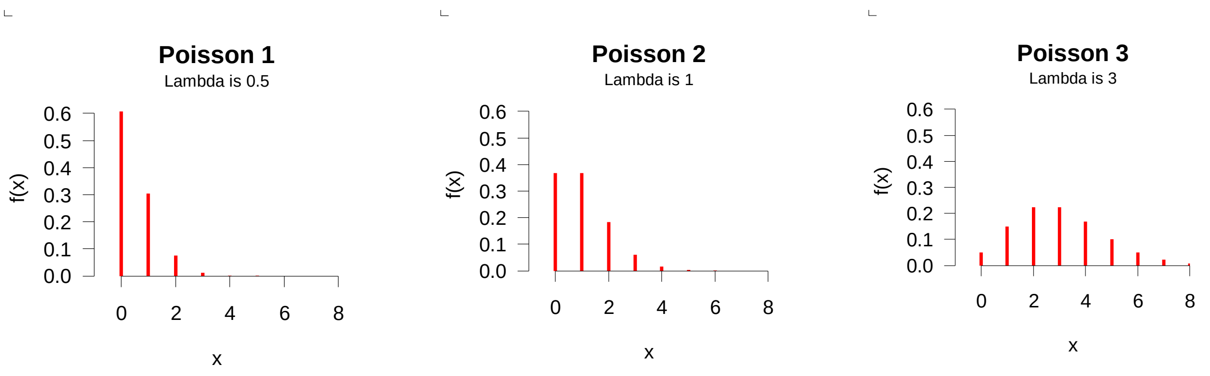


Figure 11-17 – Density functions of Poisson distributions

For computation purposes, the following relation between the Poisson distribution and the χ^2 distribution may be useful:

$$P(X \le x)=1-P(\chi^2 \le 2\lambda)$$

In this formula, χ^2 represents a $\chi_{2x}{}^2$ distributed random variable.

11.2.6.4 Transitions between distributions and appropriate approximations

Depending on the number of available measurement results, different distributions can be applied to handle the results. In this clause, some useful transitions between common distributions and their required conditions are discussed.

11.2.6.4.1 Approximation of binomial distribution by Poisson distribution

The binomial distribution can be approximated by the Poisson distribution if:

- the probability p is small (rule of thumb: p<0.1); and
- the number of executed test cases n is high enough (rule of thumb: $n > 30$).

The approximation of a binomial distributed quantity by a Poisson distribution is given by:

$$P(X = k) \cong \frac{\lambda^k}{k!} \cdot \exp(-\lambda)$$

where the Poisson distribution parameter λ is given by:

$$\lambda = p \times n$$

11.2.6.4.2 Approximation of binomial distribution by normal distribution

If a binomial distribution fulfils the rule of thumb:

$$n \times p \times q \geq 9$$

then it can be approximated by the Normal distribution:

$$B(n, p) \cong N(n \cdot p, n \cdot p \cdot q)$$

The approximation in detail reads:

$$P(X \leq x) \cong \Phi\left(\frac{X - n \cdot p}{\sqrt{n \cdot p \cdot q}}\right)$$

Especially for smaller numbers of n the following approximation may be more favourable:

$$P(x_1 \leq X \leq x_2) \cong \Phi\left(\frac{x_2 - n \cdot p + 0{,}5}{\sqrt{n \cdot p \cdot q}}\right) - \Phi\left(\frac{x_1 - n \cdot p - 0{,}5}{\sqrt{n \cdot p \cdot q}}\right)$$

11.2.6.4.3 Approximation of Poisson distribution by normal distribution

According to the Poisson limit theorem, the Poisson distribution can be approximated by the normal distribution if the distribution parameter λ fulfils the following relation:

$$\lambda = p \times n \geq 9$$

which is quite similar to the transition from binomial to normal distribution.

Then, the approximation reads:

$$P(X \leq k) \cong \Phi\left(\frac{k - \lambda}{\sqrt{\lambda}}\right)$$

11.2.6.5 Truncated distributions

Due to resource constraints of measurement equipment, some measurements have to consider timeout values. By the use of timeouts, the maximum period of time in which measurement results are considered as relevant for the measurement is limited. The resulting density function then is clipped at the right-hand side. Truncation may also occur at both ends of a density function.

For example, if the end-to-end delivery time of some message service is subject of a measurement, the introduction of timeout values may reduce the number of measurement samples. This is because all delivery times which are higher than the defined timeout value are discarded. By discarding some samples, the entirety of data is reduced which means that probabilities describing the measurement may be influenced.

In general, truncation can be described by conditional probabilities. The condition is given by the timeout value. Furthermore, probabilities are then computed under the constraint of the timeout. Truncated normal and Poisson distributions are covered in more detail by [b-Mood].

11.2.6.6 Distribution selection and parameter estimation

If a distribution is sought to describe a given data set, two steps have to be carried out. First, an appropriate distribution family (type of distribution) has to be selected and second, the corresponding parameters specifying this distribution have to be estimated. Test procedures or graphical methods may be applied for the first step, parameter estimation procedures are needed for the second.

11.2.6.6.1 Test procedures

The formulation of tests is covered in detail in clause 11.2.7.1. In this clause, three well-known tests that may be used for checking distributional assumptions are described briefly. It focuses mainly on the fundamental ideas leading to these tests.

All test procedures are based on comparisons between assumed and empirical distribution. That is, from the data on hand, the underlying distribution is guessed and then verified by applying one of the test procedures described in clauses 11.2.6.6.1.1 to 11.2.6.6.1.3.

11.2.6.6.1.1 Chi-square test

The main idea of the Chi-square test is to test whether a set of data comes from a normal distribution by building classes and checking whether the expected number of observations and the number of data in each class are similar. If the deviations between both numbers – in terms of squared differences – exceeds a corresponding χ^2 value, the distribution assumed has to be rejected.

11.2.6.6.1.2 Kolmogorov-Smirnov test

The Kolmogorov-Smirnov test is based on the cumulative distribution functions of the theoretical (assumed) and empirical distribution of the data at hand. The main idea is that the distributional assumption is rejected, if the maximum vertical distance between both cumulative density functions (CDFs) exceeds a critical value.

11.2.6.6.1.3 Shapiro-Wilk test

Shapiro and Wilk suggested a test procedure that is based on quantiles and related to the QQ-Plot introduced in clause11.3.1.3. The main idea of this test is to compare the sum of squared deviations between the points of the QQ-Plot and the best fitting straight line with a given χ^2-value.

11.2.6.6.2 Parameter estimation methods

Most frequently applied methods for parameter estimation are maximum-likelihood or moment estimation methods. For the mean of a normal distribution, both methods yield identical results. For further details see [b-Mood].

11.2.7 Evaluation of measurement data

Related to active service probing, certain issues can become much easier to handle if it is able to describe the gathered data in a very compact way. One possible way to reach this aim is to execute different tests and thereby to check some assumptions. These assumptions are stated before any testing is done. They are called "hypotheses".

From a slightly more theoretical point of view, this topic can be expressed as follows:

With every measurement sample, some information about the investigated process is retrieved. Since commonly the characteristics of the process are unknown, with every piece of additional information (i.e., every sample) the degree of knowledge increases. This knowledge is formalized by the application of statistical tests or the determination of confidence intervals for the distributional parameters of interest.

The idea of statistical tests and some simple examples are presented in clause 11.2.7.1. The construction of confidence intervals and the relation between test and confidence intervals are covered in clause 11.2.7.2.

11.2.7.1 Statistical tests

Statistical tests are introduced by specifying the test components first and afterward distinguishing different test classes and giving examples where appropriate.

11.2.7.1.1 Formulation of statistical tests

Statistical tests are formulated by specifying the following components:

- **(Null-)Hypothesis**: This hypothesis is commonly denoted by H or H_0.

Example 1: H: μ = 60.

- **Alternative hypothesis**: This one is commonly denoted by A or H_1.

Example 2: A: $\mu \neq 60$ or A: μ > 60.

- **Test statistic**: A test statistic is derived so that it is sensitive for deviation from the hypothesis in favour of the alternative. That is, the meaning of the test statistic is to notice if in fact A is true instead of H. Test statistics are commonly denoted by T.
- **Testing rule and critical value**: The testing rule states the condition under which H is rejected in favour of the alternative A. So it represents something like a switching condition.

Exmple 3: "Reject H, if $\bar{x} > c$" or "Reject H, if $|\bar{x}| > c$".

The value c is called "critical value".

- **Type I: Error level** α: The probability of rejection H, although true, is controlled by the value α. The specification of α has direct impact on c and thereby on the testing rule. Commonly, the type I error is restricted to 5 per cent (i.e., α = 0.05) or 1 per cent (i.e., α= 0.01).

A statistical test is carried out by specifying all of the above components, computing the test statistic and comparing it with the critical value. Test results are usually documented by reporting the value of the test statistic as well as the corresponding test result. Alternatively, the so called *p*-value may be reported. This value measures the "significance" or a test result in the way that if the *p*-value is smaller than the error level α, the hypothesis is rejected, otherwise it may not be rejected. A *p*-value corresponds to the smallest α-level for which the test would have been rejected.

11.2.7.1.2 Classes of statistical tests

Commonly, statistical tests are formulated in the way that the alternative hypothesis makes the statement that one wishes to prove in a statistical sense. That is, in general, tests seek to reject a given hypothesis and are therefore formulated accordingly. This is done due to the fact that if the Type I error is specified, a rejection implies that the hypothesis is untrue with user-defined certainty.

However, a number of test procedures exist that differ from the mentioned general test philosophy. Some of these examples are the tests used for selecting distributions (see clause 11.2.6.6). Their purpose is to support the hypothesis. Nevertheless, it is almost impossible to prove any equality hypothesis. Therefore, the test result can either be that there is no hint that the hypothesis is violated or that there is evidence that the assumed distribution is not appropriate. In the following, it is assumed that one wishes to reject the hypothesis in favour of the alternative hypothesis.

Two major classes of tests are distinguished, namely one-sample and two-sample tests.

- If a test is based on only one data set for which a reference alternative is to be checked, this is a one-sample test.
- On the other hand, two data sets may be compared by testing for instance the hypothesis H related to the multimedia messaging service (MMS).

H: MMS-E2E-delivery time [this week] > MMS-E2E-delivery time [last week] against the alternative hypothesis that the MMS-E2E-delivery time was reduced from last week to this week.

Furthermore, tests that are based on distributional assumptions and distribution-free tests are distinguished. Most distribution-based tests are testing for the location and dispersion/variation of an assumed distribution. For two-sample tests, both samples are assumed to be from the same type of distribution, but possibly with different parameters, for instance different location. In contrast, distribution-free tests should be applied, if there is not enough knowledge about the distribution of the data. However, distribution-free tests are in general less powerful, therefore distribution-based tests should be preferred, if appropriate.

11.2.7.1.3 Tests for normal and binomial data

In the following clauses, two of the main use-cases of statistical data are taken into concern. These clauses deal with test for normal distributed and binomial distributed data.

11.2.7.1.3.1 One-sample tests for normal data

If data are from a normal distribution with known variance $\sigma_0{}^2$, i.e., $X_1,...,X_n \sim N(\mu,\sigma_0{}^2)$, three different location tests may be carried out. All of these compare the location of a given sample to a value μ_0 that may be chosen arbitrarily.

- Test for *H:* $\mu = \mu_0$ vs. *A:* $\mu \neq \mu_0$: The corresponding test statistic is given by $T = \left|\sqrt{n}\frac{\overline{x}-\mu}{\sigma_0}\right|$ *H* is rejected, if $T > u_{1-\alpha/2}$.
- Test for *H:* $\mu \leq \mu_0$ vs. *A:* $\mu > \mu_0$: The corresponding test statistic is given by $T = \left|\sqrt{n}\frac{\overline{x}-\mu_0}{\sigma_0}\right|$
  H is rejected, if $T > u_{1-\alpha}$ (for $\alpha < 0.5$).
- Test for *H:* $\mu \geq \mu_0$ vs. *A:* $\mu < \mu_0$: The corresponding test statistic is again given by $T = \left|\sqrt{n}\frac{\overline{x}-\mu_0}{\sigma_0}\right|$
  H is rejected, if $T < u_\alpha = -u_{1-\alpha}$ (for $\alpha < 0.5$).

If data are from a normal distribution, but the variance is unknown, i.e., $X_1,...,X_n \sim N(\mu,\sigma^2)$, the variance has to be estimated from the data and the above "normal-tests" are replaced by student *t*-tests. In this case, the variance estimator:

$$s^2 = \frac{1}{n-1}\sum_{i=1}^{n}\left(x_i - \overline{x}\right)^2$$

is applied. Test statistics are replaced as follows: $T = \left|\sqrt{n}\frac{\overline{x}-\mu_0}{S}\right|$ or $T = \sqrt{n}\frac{\overline{x}-\mu_0}{\sigma_0}$, respectively.

Critical values are given by the quantiles of the *t*-distribution: $t_{n-1,\,1-\alpha/2}$, $t_{n-1,\,1-\alpha}$ or $t_{n-1,\,\alpha}$, respectively.

If instead the variance is unknown and subject of a test, i.e., $X_1,...,X_n \sim N(\mu,\sigma^2)$ with unknown μ and σ^2, the following tests comparing the variance of a given sample to a value σ_0, that may be carried out.

- Test for *H:* $\sigma = \sigma_0$ vs. *A:* $\sigma \neq \sigma_0$: The corresponding test statistic is given by $T = (n-1)/\sigma_0^2\, s^2$ with $s^2 = \frac{1}{n-1}\sum_{i=1}^{n}\left(x_i - \overline{x}\right)^2$. *H* is rejected, if $T > \chi^2_{1-\alpha/2,\,n-1}$ or $T < \chi^2_{\alpha/2,\,n-1}$.
- Test for H: $\sigma \leq \sigma_0$ vs. A: $\sigma > \sigma_0$: The corresponding test statistic is again given by T = (n-1)/ σ_0^2 s^2 with s^2 as given above. H is rejected, if $T > \chi^2_{1-\alpha\, n-1}$.
- Test for H: $\sigma \geq \sigma_0$ vs. A: $\sigma < \sigma_0$: Test statistic and empirical variance are as before. In this case, H is rejected, if $T < \chi^2_{\alpha,\,n-1}$.

11.2.7.1.3.2 Two-sample tests for normal data

In the case of two samples, that are to be compared, two very different situations are distinguished. The two samples can either be collected on the same observational units or can be observed independently. If both samples are from the same units, for example measuring the cut-off call ratio at different network elements before and after a new piece of software is installed, the two samples are called paired and two observations from the same unit will generally be correlated. In this case, the differences between both measurements for each unit are computed and the new observations $D_i = X_i - Y_i$ are assumed to be normal with expectation $\mu_D = \mu_X - \mu_Y$. Then, the above tests for normal data may be applied, for instance to test for $\mu_D = 0$, i.e., $\mu_X = \mu_Y$ to prove that there is a significant difference between both samples.

For independent data from two samples, both assumed to be normally distributed with the same known variance, but possibly different expectations, i.e., $X_1,\ldots,X_n \sim N(\mu_X,\sigma^2)$ and $Y_1,\ldots,Y_m \sim N(\mu_Y,\sigma^2)$, tests to compare both means are given as follows.

The following test statistic T is defined for testing the hypotheses:

$$T = \frac{\bar{x}-\bar{y}}{\sigma\sqrt{\frac{1}{n}+\frac{1}{m}}}$$

- Test for H: $\mu_X = \mu_Y$ vs. A: $\mu_X \neq \mu_Y$: H is rejected, if $|T| > u_{1-\alpha/2}$
- Test for H: $\mu_X \leq \mu_Y$ vs. A: $\mu_X > \mu_Y$: H is rejected, if $T > u_{1-\alpha}$
- Test for H: $\mu_X \geq \mu_Y$ vs. A: $\mu_X < \mu_Y$: H is rejected, if $T < u_\alpha$

If the variance is unknown but assumed to be equal for both samples, the normal distribution is again replaced by a student t-distribution resulting in the following test procedures.

$$T = \frac{\bar{x}-\bar{y}}{s\sqrt{\frac{1}{n}+\frac{1}{m}}} \text{ with } s = \frac{\sum_{i=1}^{n} x_i^2 - \frac{1}{n}\left(\sum_{i=1}^{n} x_i\right)^2 + \sum_{i=1}^{m} y_i^2 - \frac{1}{m}\left(\sum_{i=1}^{m} y_i\right)^2}{n+m-2}$$

- Test for H: $\mu_X = \mu_Y$ *vs*. A: $\mu_X \neq \mu_Y$: H is rejected, if $|T| > t_{1-\alpha/2,\, n+m-2}$.
- Test for H: $\mu_X \leq \mu_Y$ *vs*. A: $\mu_X > \mu_Y$: H is rejected, if $T > t_{1-\alpha,\, n+m-2}$.
- Test for H: $\mu_X \geq \mu_Y$ *vs*. A: $\mu_X < \mu_Y$: H is rejected, if $T < t_{\alpha,\, n+m-2}$.

In general, before carrying out one of the above tests, the assumption of equal variances has to be verified. This can be done by using the following test:

- Test for H: $\sigma_X^2 = \sigma_Y^2$ vs. A: $\sigma_X^2 \neq \sigma_Y^2$: The corresponding test statistic is given by $T = \frac{s_X^2}{s_Y^2}$,

where:

$$S_X^2 = \frac{1}{n-1}\sum_{i=1}^{n}(x_i-\bar{x})^2 \text{ and } S_Y^2 = \frac{1}{m-1}\sum_{i=1}^{m}(y_i-\bar{y})^2$$

11.2.7.1.3.3 Test for binomial data

For binomial data, tests for the probability of success p may be carried out that compare the sample probability to some specified value p_0. Three one-sample tests may be derived by computing the critical values under the three hypotheses.

If m is the number of successful trials, the first hypothesis is rejected, if:

$$m > c_{1-\alpha/2} \text{ or } m < d_{\alpha/2}$$

where:

$$c_{1-\alpha/2} = \min_{k\in\{1,\ldots,n\}} \sum_{i=k+1}^{n} \binom{n}{i} p_0^i (1-p_0)^{n-i} \leq \alpha$$

and

$$d_{\alpha/2} = \max_{k\in\{1,\ldots,n\}} \sum_{i=0}^{k-1} \binom{n}{i} p_0^i (1-p_0)^{n-i} \leq \alpha$$

The second hypothesis is rejected if $m > c_{1-\alpha}$ and the third one if $m < d_{\alpha}$.

An alternative way may be appropriate if large numbers of samples are available (large means $np(1-p) \geq 9$ is fulfilled). In this case, the test statistic:

$$Z = \frac{m - np_0}{\sqrt{np_0(1-p_0)}}$$

can be applied. In this case:

- the first hypothesis is rejected, if $|Z| > u_{1-\alpha/2}$;
- the second one, if $|Z| > u_{1-\alpha}$; and
- the third one, if $|Z| < u_{1-\alpha}$.

11.2.7.1.4 Distribution-free tests for location

If the location for two sets of random variables shall be compared, but there is not enough knowledge for a distributional assumption, distribution-free test may be applied.

11.2.7.1.4.1 Sign tests

In the case of paired samples, the differences between both measurements for each unit are again computed as $D_i = X_i - Y_i$. If both distributions have the same location, the probability of $X_i < Y_i$ should equal the probability of $X_i > Y_i$ and both should equal 0.5. Based on this consideration, the following tests may be carried out.

- Test for *H:* $P(X_i>Y_i) = P(X_i<Y_i) = 0.5$ *vs. A:* $P(X_i>Y_i) \neq o.5$.

- Test for H: $P(X_i>Y_i) \leq 0.5$ *vs. A:* $P(X_i>Y_i) > o.5$
- Test for H: $P(X_i>Y_i) \geq 0.5$ *vs. A:* $P(X_i>Y_i) < o.5$

In all cases, the test statistic T is given as the number of positive differences D_i. This test statistic is a binomial random variable with $p=P(X_i>Y_i)$. Therefore, all of the above stated hypotheses are tested by applying a binomial test as described in more detail in clause 11.2.7.1.3.3.

11.2.7.1.4.2 Sign rank test

For the same situation, another kind of test, namely the sign rank test, may be preferable if the distribution of differences is symmetric around some value δ, that is $P(D_i \leq \delta - a) = P(D_i \geq \delta + a)$ for all real numbers a. In comparison to the previous clause, the sign rank test not only uses the signs of the differences between both measurements, but also the absolute values in terms of their ranks.

For each of the following hypotheses, the test statistic $T = \sum_{i=1}^{n} V_i R(|D_i|)$ with $V_i = 1$, if $D_i > 0$ and $V_i = 0$ otherwise and $R(\cdot)$ the rank operator that sorts the entries and gives rank *1* to the smallest entry and rank *n* to the largest, is used as a basis for the test decision.

- Test for *H*: $\delta = 0$ vs. *A*: $\delta \neq 0$.
- Test for *H*: $\delta \leq 0$ vs. *A*: $\delta > 0$.
- Test for *H*: $\delta \geq 0$ vs. *A*: $\delta < 0$.

For the test statistic, there is a distribution with expectation $E(T) = \frac{1}{4} n(n+1)$ and the variance $Var(T) = \frac{1}{24} n(n+1)(2n+1)$. The quantiles of the resulting distribution are given in statistical text books on nonparametric methods. However, for the case where $n \geq 20$, the distribution of $\frac{T - E(T)}{\sqrt{Var(T)}}$ may be approximated by a standard normal distribution.

11.2.7.1.4.3 Wilcoxon rank sum test

In contradiction to both tests explained before, the rank sum test suggested by Wilcoxon is used for independent samples. It assumes that both samples come from the same kind of distribution, but with a shifted location, that is $X_1,...,X_n$ *and* $Y_1,...,Y_m$ $(n \leq m)$ are independent and have continuous distribution functions F_X and F_Y respectively. These are shifted by δ, i.e., $F_X(x) = F_Y(x+\delta)$. Meaningful hypotheses are the following:

- Test for *H*: $\delta = 0$ vs. *A*: $\delta \neq 0$.
- Test for *H*: $\delta \leq 0$ vs. *A*: $\delta > 0$.
- Test for *H*: $\delta \geq 0$ vs. *A*: $\delta < 0$.

In this situation, for instance a rejection of the third hypothesis, i.e., $\delta < 0$ would imply that the location of the second distribution function is significantly smaller, that is the *y*-values are in general smaller than the *x*-values.

For the above tests, a sensible test statistic is derived by combining both data sets to one sample and computing ranks by ordering the values according to their size. The test statistic *T* is now given by the sum of all ranks for values from the first sample, i.e., the *x*-values. For this test statistic, expectation and variance are given by:

$$E(T) = \frac{1}{2} n(n+m+1)$$

and

$$Var(T) = \frac{1}{12} n\, m\, (n+m+1)$$

respectively. Again, exact critical values for these tests are not easy to derive, but approximations exist. If $n,m \geq 4$ and $n+m \geq 30$, the distribution of $\frac{T - E(T)}{\sqrt{Var(T)}}$ may be approximated by a standard normal distribution.

11.2.7.2 Confidence interval

In contrast to point estimators where a single number is used to summarize measurement data (see methods for estimating moments or quantiles in clause 11.2.5), confidence intervals describe an interval that covers the true parameter value with a certain probability. Usual probability measures are in the 90 per cent range. For example, a confidence interval represents the interval in which the mean of the underlying distribution lies with a probability of 95 per cent or with a probability of 99 per cent.

Confidence intervals are related to statistical tests in the sense that a confidence interval with a given confidence level, for instance 95 per cent contains all confidence levels denoted by 1-α, where α corresponds to the Type I error level for tests.

As a rule of thumb, number of samples within a measurement campaign correlates with the reliability of results. In other word, the higher the number of collected samples, the more precise and trustworthy the results are.

The computation of confidence intervals depends heavily on the assumed kind of distribution. In the following, the computation of confidence intervals is described for the binomial and the normal (Gaussian) distribution.

11.2.7.2.1 Binomial distribution

This clause defines how to compute a confidence interval to the level $1-\alpha$ for p for a binomial distribution.

At first, the computation of a confidence interval $[p_1; p_2]$ according to the binomial distribution depends on the number of tests n which are executed to determine p.

- If the condition $n \cdot p \cdot q \geq 9$ is fulfilled, the binomial distribution can be approximated by the Normal distribution which eases the computation of the according confidence interval.

The values for p_1 and p_2 are then given by:

$$p_1 = \frac{2m + u^2_{1-\frac{\alpha}{2}} - u_{1-\frac{\alpha}{2}} \cdot \sqrt{u^2_{1-\frac{\alpha}{2}} + 4m\left(1-\frac{m}{n}\right)}}{2\left(n + u^2_{1-\frac{\alpha}{2}}\right)}$$

$$p_2 = \frac{2m + u^2_{1-\frac{\alpha}{2}} + u_{1-\frac{\alpha}{2}} \cdot \sqrt{u^2_{1-\frac{\alpha}{2}} + 4m\left(1-\frac{m}{n}\right)}}{2\left(n + u^2_{1-\frac{\alpha}{2}}\right)}$$

with the known parameters m and n from clause 11.2.6.3.2 (Binomial distribution). The term $u_{1-\frac{\alpha}{2}}$ represents the $1-\frac{\alpha}{2}$ quantile of the standard normal distribution $N(0,1)$. Some examples for different confidence levels α and their according $u_{1-\frac{\alpha}{2}}$ quantile values are given in the following table.

Confidence level $1-\alpha$	α	**Term** $1-\frac{\alpha}{2}$	**Quantile** $u_{1-\frac{\alpha}{2}}$
0.9 (≙90%)	0.1	0.95	1.6449
0.95 (≙95%)	0.05	0.975	1.96
0.96 (≙96%)	0.04	0.98	2.0537
0.97 (≙97%)	0.03	0.985	2.1701
0.98 (≙98%)	0.02	0.99	2.3263

Confidence level $1-\alpha$	α	Term $1-\frac{\alpha}{2}$	Quantile $u_{1-\frac{\alpha}{2}}$
0.99 ($\hat{=}$99%)	0.01	0.995	2.5758
0.999 ($\hat{=}$99.9%)	0.001	0.9995	3.2905

The quantile values can be taken from tabulated quantile values of the standard normal distribution respectively the cumulated distribution function of this distribution.

- If the previous condition is not fulfilled, the confidence interval has to be computed with regard to the binomial distribution itself. In this case, the parameters p_1 and p_2 are called "Pearson-Clopper values". In detail, the values p_1 and p_2 represent interval boundaries which fulfil the relations:

$$P(X \geq m) = \sum_{k=m}^{n} \binom{n}{k} p_1^k (1-p_1)^{n-k} = \frac{\alpha}{2}$$

$$P(X \leq m) = \sum_{k=0}^{m} \binom{n}{k} p_2^k (1-p_2)^{n-k} = \frac{\alpha}{2}$$

Using the relation between the binomial distribution and the F distribution with 2 × (x-1), 2 × (n-x) degrees of freedom (see clause 11.2.6.2.3):

$$P(X < x) = 1 - P\left(F \leq \frac{n-x}{x+1} \cdot \frac{p}{1-p} \right)$$

The Pearson-Clopper values can be determined as:

$$p_1 = \frac{m \cdot F_{2m, 2(n-m+1); \frac{\alpha}{2}}}{n-m+1+m \cdot F_{2m, 2(n-m+1); \frac{\alpha}{2}}}$$

$$p_2 = \frac{(m+1) \cdot F_{2(m+1), 2(n-m); 1-\frac{\alpha}{2}}}{n-m+(m+1) \cdot F_{2(m+1), 2(n-m); 1-\frac{\alpha}{2}}}$$

$F_{n_1, n_2; \gamma}$ represents the γ quantile of a F distribution with degrees of freedom n_1 and n_2 which are tabulated in the literature. An approximation for γ quantiles of the F distribution is given in clause 11.2.6.2.3.

11.2.7.2.2 Normal (Gaussian) distribution

In the normal distribution, confidence statements depend on the composition of known and unknown parameters. This means that different computations have to be applied if mean value and/or variance of the distribution are known. If the parameters are not known, they can be estimated by empirical values (see clause 11.2.5). Furthermore, confidence statements can be made related to the expected value, to the variance and to the standard deviation.

To sum up, the estimated empirical values of the normal distribution are:

- Empirical mean $\bar{x} = \frac{1}{n} \sum_{i=1}^{n} x_i$, where $x_i, i = 1,...,n$ are the sample values.
- Empirical variance $s^2 = \frac{1}{n-1} \sum_{i=1}^{n} (x_i - \bar{x})^2$.

- Empirical standard deviation $s = \sqrt{\frac{1}{n-1}\sum_{i=1}^{n}(x_i - \bar{x})^2}$.

Based on these expressions, the following terms are applied to estimate confidence intervals for the mean value, the variance and the standard deviation of a Normal distribution:

- Confidence interval for mean value μ if variance σ^2 is known:

$$\left[\bar{x} - u_{1-\frac{\alpha}{2}} \cdot \frac{\sigma}{\sqrt{n}} ; \bar{x} + u_{1-\frac{\alpha}{2}} \cdot \frac{\sigma}{\sqrt{n}}\right]$$

- Confidence interval for mean value μ if variance σ^2 is unknown:

$$\left[\bar{x} - t_{n-1;1-\frac{\alpha}{2}} \cdot \frac{s}{\sqrt{n}} ; \bar{x} + t_{n-1;1-\frac{\alpha}{2}} \cdot \frac{s}{\sqrt{n}}\right]$$

- Confidence interval for variance σ^2 if mean μ is known:

$$\left[\frac{\sum_{i=1}^{n}(x_i - \varpi)^2}{\chi^2_{n;1-\frac{\alpha}{2}}} ; \frac{\sum_{i=1}^{n}(x_i - \varpi)^2}{\chi^2_{n;\frac{\alpha}{2}}}\right]$$

- Confidence interval for variance σ^2 if mean μ is unknown:

$$\left[\frac{(n-1)s^2}{\chi^2_{n-1;1-\frac{\alpha}{2}}} ; \frac{(n-1)s^2}{\chi^2_{n-1;\frac{\alpha}{2}}}\right]$$

- Confidence interval for standard deviation σ if mean μ is unknown:

$$\left[s \cdot \sqrt{\frac{n-1}{\chi^2_{n-1;1-\frac{\alpha}{2}}}} ; s \cdot \sqrt{\frac{n-1}{\chi^2_{n-1;\frac{\alpha}{2}}}}\right]$$

- Confidence interval for standard deviation σ if mean μ is known:

$$\left[\sqrt{\frac{\sum_{i=1}^{n}(x_i - \varpi)^2}{\chi^2_{n;1-\frac{\alpha}{2}}}} ; \sqrt{\frac{\sum_{i=1}^{n}(x_i - \varpi)^2}{\chi^2_{n;\frac{\alpha}{2}}}}\right]$$

11.2.7.3 Required sample size for certain confidence levels

In this clause, the relationship between the number of acquired measurement samples and the resulting confidence level is in the focus.

Whereas in the usual measurement chain at first the samples are collected and afterwards the confidence level is determined, in some situations the reverse procedure may be necessary. For example, during a measurement period there may exist preconditions which require a certain confidence level which should be reached during the measurements. The unknown parameter is the number of measurement samples which have to be collected to reach this confidence level and must be determined in advance.

Tables of required sample size depending on desired confidence levels are given in clause IX.5. The tables are based on the binomial distribution and the corresponding Pearson-Clopper expressions. Due to this fact, they are available and valid for rather small sample sizes like they occur if manual tests are executed.

The tables provide two kinds of information:

- The limits and the range of the confidence interval of the mean for an increasing number of samples whereas the estimated rate is constant.
- The range ("span") of the confidence interval of the mean for a varying estimated rate whereas the number of samples is constant.

Based on this, one can state in advance the maximum span of the confidence interval based on the number of samples which should be gathered.

11.3 Visualization techniques

In this clause, some useful visualization techniques are presented. This should not be considered as a complete overview over all possible methods, but provides some standard techniques and some non-standard alternatives.

In the following, a distinction is made between the static and the dynamic data. By static data, variables are do not change systematically within the time period under consideration, i.e., which are not subject to seasonal or daily influences. Dynamic data on the contrary are data which vary systematically over the time. Examples are usage data that show a typical curve with high usage during the day (in particular in the afternoon) and low usage at night.

11.3.1 Visualization of static data

Visualization techniques for static data assume that the underlying distribution does not change over the considered time period and try to give a compressed overview over this distribution.

11.3.1.1 Histograms

Histograms compress the information by building classes and counting the number of data values falling into each of the specified classes. The main idea is to represent each class by a bar with area equivalent to the portion of data values included. An example is given in Figure 11-18.

Histograms can be viewed as density estimators since the area of the visualized bars adds up to one, smoothed density estimation curved can also be applied as available in most of the common statistical analysis computer packages. The two plots of example 1 in Figure 11-18 with different bar width illustrate the concept of histograms. Here one bar in the first plot contains the same number of data values than five successive bars in the second plot, therefore the height of one bar in plot one is given by the mean height of the five corresponding bars in plot two. Histograms even allow bar heights greater than one, if the bar width is small, respectively.

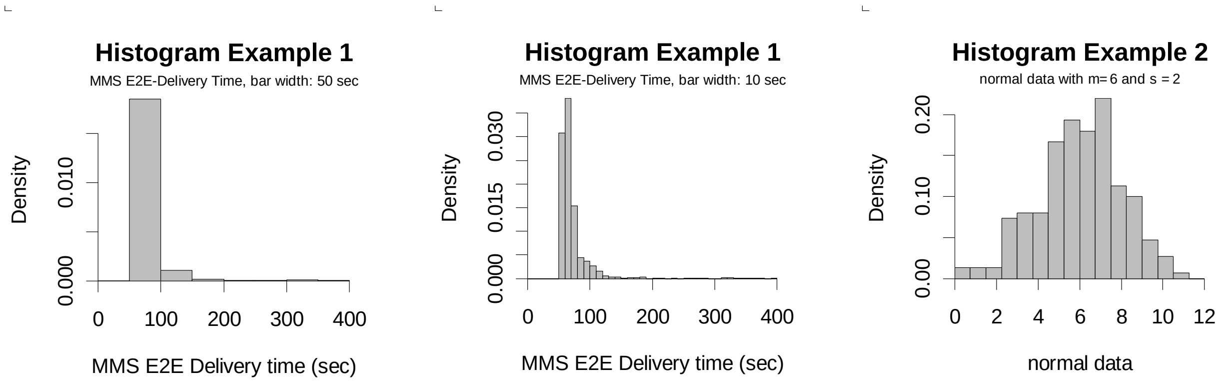


Figure 11-18 – Examples of histograms

11.3.1.2 Barplots

Barplots are suitable for ordinal samples and to visualize the total or relative number of elements from a sample with different values of a characteristic of interest. Barplots are used if the distribution of users to groups with different business states or of trouble tickets with different priorities or other examples of ordinal samples are to be visualized. Since for ordinal samples, the differences between groups are not to be interpreted in a numerical sense, the main difference in comparison to histograms is that the widths of the bars do not have any meaning, only the height corresponds to the total or relative number of elements represented by each bar. Moreover, commonly gaps are left between the bars, to illustrate that ordinal samples are visualized. Examples are given in Figure 11-19 where months and priorities are used as example units.

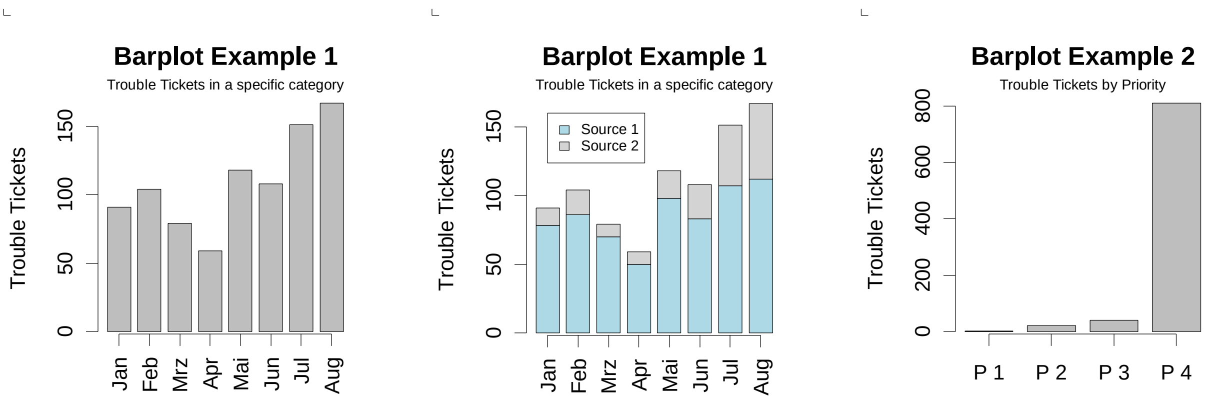


Figure 11-19 – Examples of barplots

11.3.1.3 QQ-Plots

An important tool for checking the normality assumption is the so called quantile-quantile-plot (QQ-Plot). This plot compares the quantiles of two distributions in a scatter plot. In particular the theoretical quantiles of the normal distribution may be compared to empirical quantiles from a sample of interest, but also any other distributional assumptions can be checked, respectively.

In case of a normal QQ-Plot, theoretical quantiles can be taken from the standard normal distribution. The points of the resulting scatter plot should then fall on a straight line with slope corresponding to the empirical standard deviation of the sample. Figure 11-20 gives three example normal QQ-Plots for normal and non-normal samples.

In the first plot, the sample is in fact normal and the normal QQ-Plot also supports the assumption of normal data. For both other plots, non-normal data are simulated to visualize the normal QQ-Plot in cases where the assumptions are violated. In the second example, the entire distribution disagrees with the normal assumption while in example three, only the right tail of the distribution does not agree with the normality assumption.

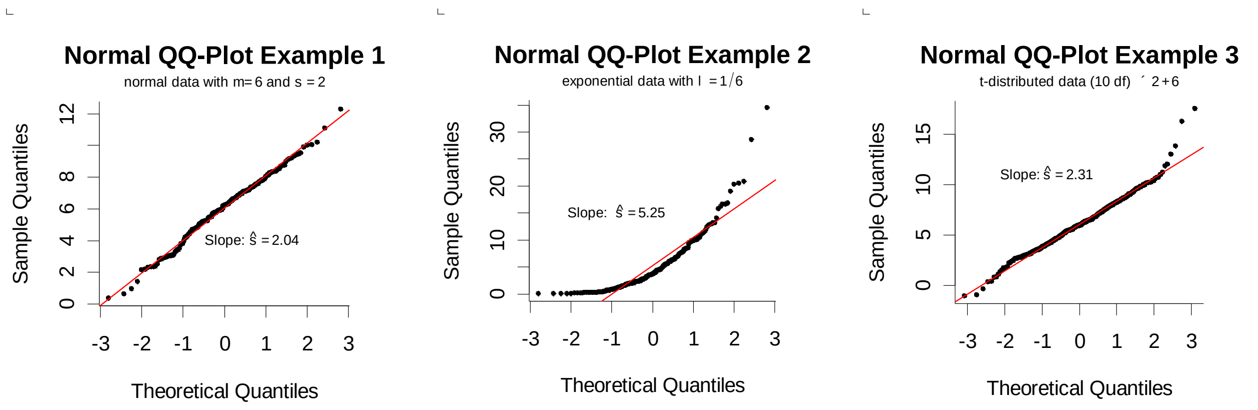


Figure 11-20 – Examples of normal QQ-Plots

11.3.1.4 Boxplots

Boxplots, as the name suggests, consist of a box and some additional shapes called whiskers. These visualize the data information compressed in only a few numbers based on quantiles of the empirical distribution. The end-points of the box are given by the 25% and 75%-quantile (also called quartiles), the horizontal line is given by the median of the data (50%-quantile). Therefore the box contains 50 per cent of the given data. The whiskers (in the example plots represented by dotted lines) extend to the most extreme data point which is no more than 1.5 times the interquartile range (between the 25%-quantile and the 75%-quantile) from the box. All data points outside this interval are individually added and may be viewed as outliers. Figure 11-21 gives some boxplot examples.

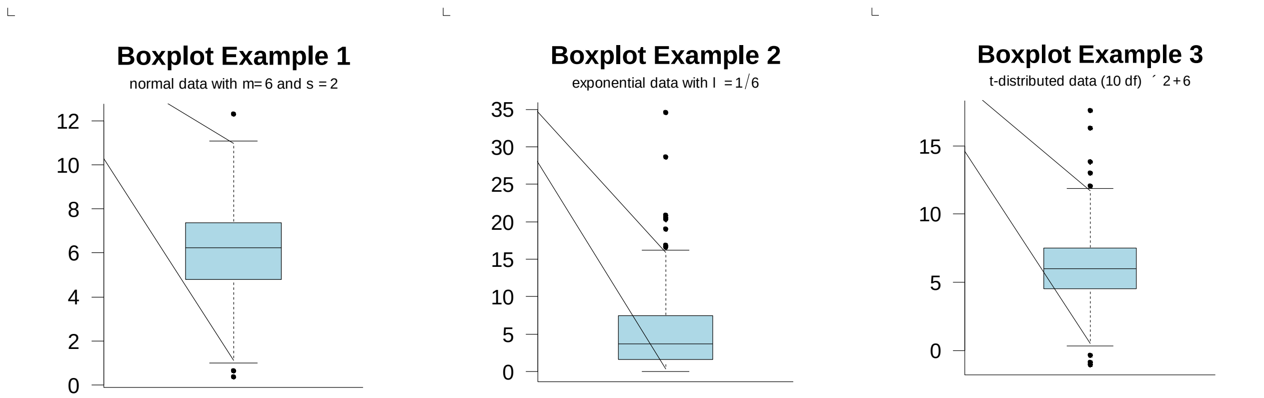


Figure 11-21 – Boxplot examples

11.3.2 Visualization of dynamic data

For dynamic data, visualization techniques should take the dynamic aspect into account. This can be done by visualizing single data points or by using aggregated data values or summary statistics like the mean, respectively. In addition, visualizations as introduced for static data can be compared over time. Boxplots as described in clause 11.3.1.4 are an adequate tool for characterizing changes over time and will be addressed in clause 11.3.2.2.

If summary statistics are applied, a chronological classification of the data is needed. This can be done by summarizing a given number of succeeding data points or by summarizing data of a given time period like an hour or a day. However, data within a time period or group should be as homogeneous as possible, i.e., changes of the parameter of interest should not be hidden by large classification groups for instance due to long time intervals.

11.3.2.1 Line diagrams

Line diagrams may be based on single data points or values of a summary statistic like the mean. They only provide a visual comparison of the data points over time without any kind of assessment. This can be achieved by adding control limits yielding control charts as will be described in clause 11.6.2. In Figure 11-22, examples of line diagrams are given. If the measurements are not equidistant in time, points of measurement should be specified by points in addition to the connecting line.

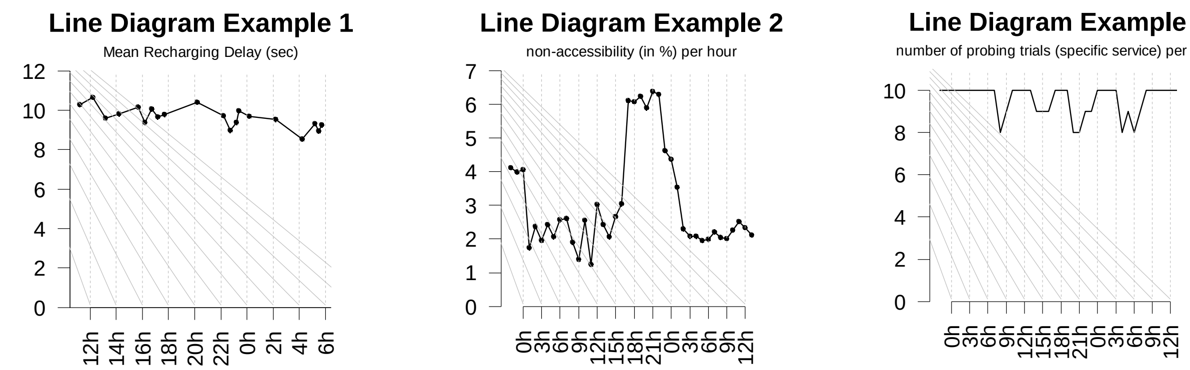


Figure 11-22 – Examples of line diagrams

11.3.2.2 Temporal changing boxplots

Instead of a single summary statistic, boxplots may be used as a summarizing tool and visualized and compared over time. Boxplots are not only appropriate for comparing empirical distributions over time, but also for unordered groups like the comparison of delay or delivery times for different service providers or vendors. These boxplots are then called parallel boxplots. Examples for both cases are given in Figure 11-23.

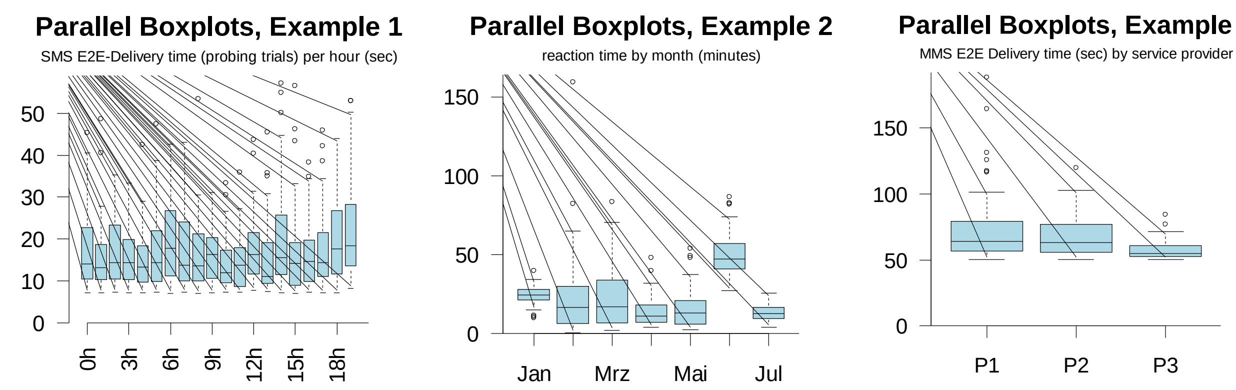


Figure 11-23 – Examples of parallel boxplots

11.3.2.3 MMQ-Plots

Median-mean-quantile plots (MMQ-Plots) visualize the serial of each of the three statistics median, mean and 95%-quantile over time in a common plot. The 95%-quantile characterizes the upper tail of the empirical distribution, while mean and median as measures for location allow conclusions about outliers which will only affect the mean due to its non-robustness. Examples of MMQ-Plots are given in Figure 11-24.

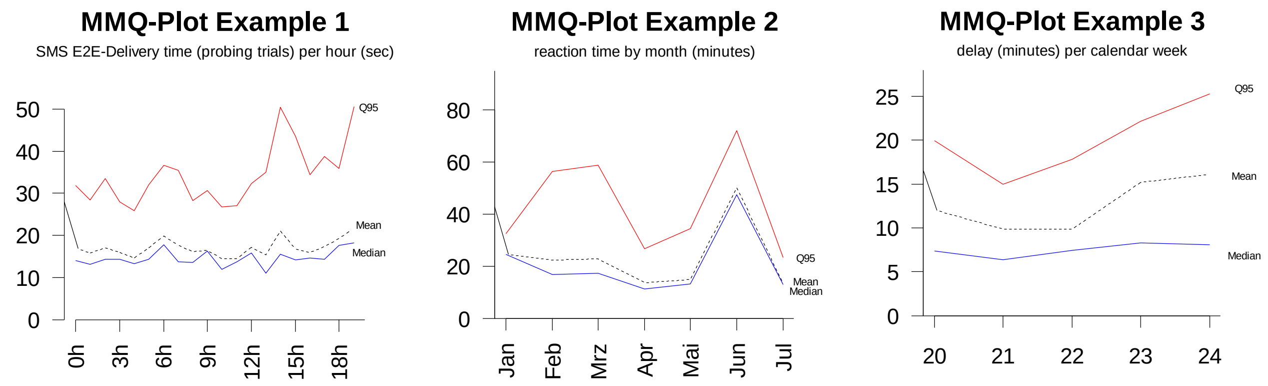


Figure 11-24 – Examples of MMQ-Plots and their temporal behaviour

11.4 Time series modelling

Beneath stationary processes, on the one hand temporal changes are very interesting. On the other hand, there are many cases where an appropriate description of the changes in a system over time has to be handled. Both cases are covered by the so called time series and their appropriate methods.

For example, if measurements in a mobile network are executed for a period of one month with regard to the transmitted traffic volume, a periodic behaviour will be observable. Depending on the hour of the day and on the day of the week, different traffic situations are expected. Example of a daily traffic pattern is given in Figure 11-25.

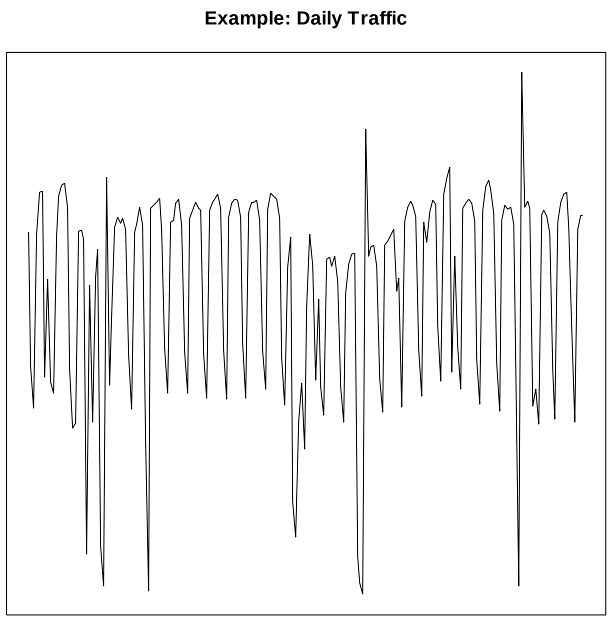


Figure 11-25 – Example of daily traffic

With respect to the time series, four different main areas can be identified:

1) Descriptive characterization
   - This method is based on the principle of "let the data speak". A very basic procedures are applied to achieve a description of time series which is as exact and detailed as possible. Especially the method of extracting different components with respect to different time scales is presented.

2) Modelling
   - Time series are interpreted as a realization of a stochastic process which means a sequence of dependent random variables. Under the stationary assumption (i.e., the main characteristics of the process are not varying over time), methods using so called auto-regressive moving average (ARMA) processes are in the focus.

3) Prognosis
   - If it is assumed that the found stochastical model is valid, it is possible to state the future behaviour.

4) Monitoring
   - Methods in this area are used to model variables which describe technical processes. The aim is to enable the controlling and monitoring of the related processes. Specialized visualization techniques, so called control charts, allow the deployment of these mechanisms in the operational realm. Their main advantage is the fact that no further detailed knowledge about statistical methods is required.

11.4.1 Descriptive characterization

A time series is an amount of observations x_t which are ordered in ascending order by a time index t. The observations are interpreted as realizations of a random variable X_t. In general, it is assumed that at the point of time the analysis is done, a certain history of observations is available. The history is formally described as a finite amount of parameters N.

$$x_1, x_2, ..., x_N$$

In a more practical manner, the observations are represented by certain measurement results which are collected over time and which are analysed in their order of occurrence. Furthermore, the observations can be distinguished according to their timely occurrence: A stochastical process can be observed at fixed points of time which leads to equally spaced intervals between the observations. Another way to observe a process is by executing permanent measurements which deliver measurement results related to some events, so called event data. In effect, the time intervals between consecutive events may vary heavily. In this case it may be appropriate to approximate the situation at fixed points of time. This allows using mechanisms which are designed for discrete time series.

11.4.1.1 Empirical moments

Similar to the handling of one-dimensional measurement results, descriptive characteristics (clause 11.2) can be deployed to describe the main characteristics of a time series. In particular, the arithmetic mean value or the variance and the standard deviation are addressed by this issue.

However, these global parameters of time series are only applicable if there is no systematic change in the series, the so called stationary time series. In these cases, a movement in a certain direction (i.e., a trend) is not allowed. Concerning non-stationary time series, it might be useful to fragment the series in smaller parts. Then, the partial time series can be assumed to be approximately stationary. This allows using some procedures with a local meaning which are presented in clause 11.4.1.4.

Beyond this, the question arises if dependencies exist between different observations at different points of time. Corresponding to the covariance, the autocovariance function:

$$c_j = \frac{1}{N}\sum_t (x_t - \bar{x})(x_{t+j} - \bar{x})$$

and the autocorrelation function:

$$r_j = \frac{c_j}{c_0}$$

are defined to measure linear dependencies between succeeding observations of a process. Both functions are defined as functions of the temporal difference (lags) j = – (N-1), .., –1, 0, 1, ..., (N-1) between the observations.

The graphical representation of the autocorrelation function r_j is named correlogram. Correlograms are of high relevance for finding cyclic (i.e., periodic) structures in the gathered measurement data.

Figure 11-26 gives an example of a correlogram.

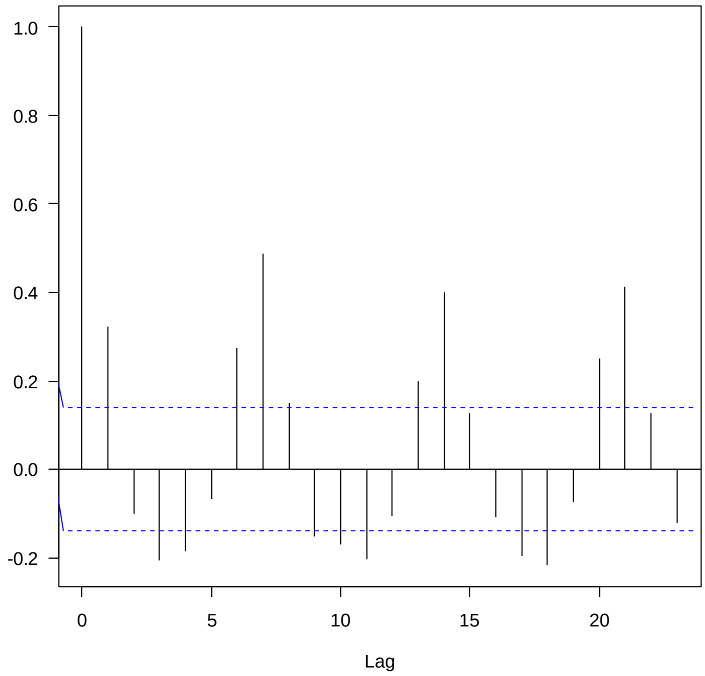


Figure 11-26 – Example of correlogram

Furthermore, the autocovariance function again depends on the stationary character of the measurement results because its definition assumes the existence of constant mean values.

11.4.1.2 Decomposition of time series

The reflection of the example given in the last clause shows the following: Data which is related to the behaviour of the users leads to a mixture of short-term cycles like days and long-term cycles which change on an annual basis. This means that the daily changes are overlaid by, for example, weekly changes as well as seasonal or yearly modifications.

Now, the aim to reach by the decomposition of time series is the following: The time series should be decomposed to be able to identify the long-term trend of a process. The question which should be answered is: Are there any long-term movements behind the different layered cyclic processes? Especially with respect to management decisions, this information can be of a very high importance.

In general, time series are based on two different assumptions:

- Additive time series:

$$x_t = T_t + K_t + S_t + R_t \text{ for } t = 1, \ldots, n$$

- Multiplicative time series:

$$x_t = T_t \times K_t \times S_t \times R_t \quad \text{for } t = 1, \ldots.., n$$

The different parts are in detail:

- T_t (*Trend*) represents the systematic long-term changes of the mean level of a time series.
- The economic component K_t includes long-term changes in the model which need not to be regular in any way. The combination of T_t and K_t often is concentrated in terms of a smooth component G_t.
- Cyclical changes are represented by the term S_t which is the seasonal component of the process.
- R_t stands for an irregular behaviour of the process which is not known in advance. This component is assumed to be part of a random process which oscillates around the zero level.

If seasonal changes occur with the same amplitude in each period, the additive time series model should be taken into account. However, if seasonal changes causes change of their amplitude with every observation period while they keep their general behaviour, the multiplicative approach may be the better choice.

Figure 11-27 depicts an example of decomposition in different components.

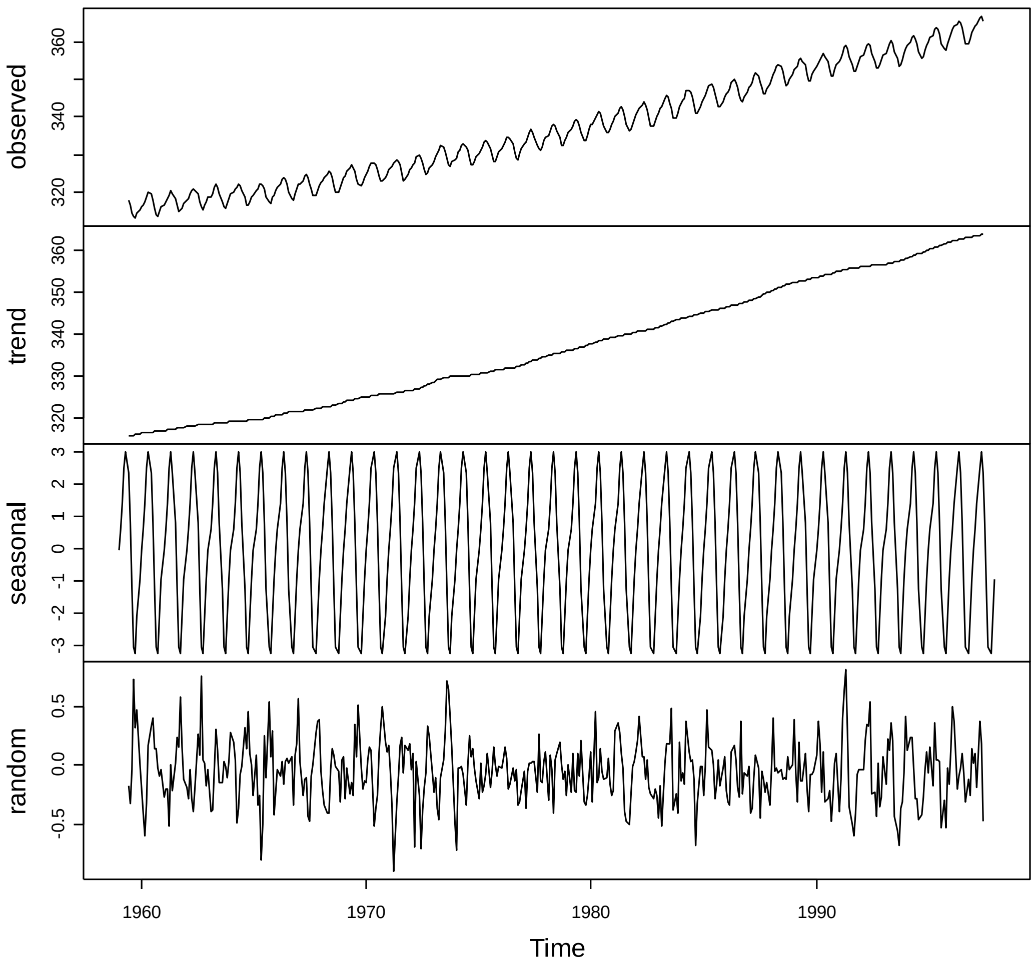


Figure 11-27 – Example of decomposition in different components

In general, there is no statement on how to process a given time series in an optimal way. Therefore, different approaches or modelling types may lead to different results. Particularly, two different approaches can be distinguished:

- Global component model: The time series is mapped to a global model which is valid for all clauses of the time series and which is adapted to the specific time series. The estimation of the trend component is usually done by the adaption of linear and non-linear regression models based on the method of minimized square values.
- Local component model: In this model, the time series is split up in different clauses. For every clause, a certain local model with partial meaning can be developed. The concatenation of all the local models represents the complete time series. The trend estimation is normally done by filtering mechanisms and non-linear procedures.

Both different models are discussed in the following clauses.

11.4.1.3 Determination of the trend component

The trend component describes the long-term behaviour of a process. Due to the importance that the trend component may have with regard to the management view, the choice of the appropriate model is one of the main issues. The use of an incorrect model has a far reaching influence with respect to the quality of the component model.

Furthermore, wrong assumptions may lead to misinterpretations. For example, if a linear trend model is introduced, the conclusions drawn from such a model are restricted to a linear character. It is not possible to describe processes which own a more complex function with such a simple model which may result in obtaining wrong conclusions.

11.4.1.3.1 Trend function types

Different types of trend functions are possible. All of them introduce some unknown coefficients a_i which must be determined by calculation or estimation. The subsequent clauses introduce different approaches and their characteristics.

11.4.1.3.1.1 Linear trend function

The most well-known approach to model a trend function is a linear function. It is assumed that the observations x_t depend on the time index t in a linear manner. This relation can be formalized by the expression:

$$x_t = a_1 + a_2 t$$

It is very easy to interpret this model since the sign of a_1 represents if the time series increases (positive sign) or decreases (negative sign) over time.

11.4.1.3.1.2 Polynomial trend function

Extending the linear approach, the polynomial approach assumes that a time series can be described as the composition of different functions *m*:

$$x_t = a_1 m_1(t) + a_2 m_2(t) + \ldots + a_k m_k(t)$$

$m_i(t)$ are arbitrary known functions. It is important that the combination of all the single expressions $a_i m_i(t)$ is linear.

A very simple approach is to define the m_i functions as polynomials of rank (i-1). Then, the approach reads:

$$x_t = a_1 + a_2 t + \ldots + a_k t^{k-1}$$

According to the theory, p + 1 points of a function can be perfectly approximated if a polynomial of rank p is used. This means it is possible to reach a perfect approximation between model and any time series in any case. However, there are two serious disadvantages:

- Resulting models are very complex and cannot be interpreted in a simple manner (compared with the basic trend model).
- The assimilation takes only the available data into account. Therefore, it is not possible to make statements about the behaviour in the future.

Both effects are considered as overfitting effects.

11.4.1.3.1.3 Non-linear trend models

Lots of different non-linear trend models are available. Because of difficulties to describe all models in a general manner, this clause concentrates on some important cases with a very special meaning:

1) Exponential models:

$$x_t = e^{a_1 m_1(t) + a_2 m_2(t) + \ldots + a_k m_k(t)}$$

2) Power models:

$$x_t = m_1(t)^{a_1} \cdot m_2(t)^{a2} \cdot \ldots m_k(t)^{a_k}$$

Both models can be reduced to linear models if a logarithmic operation is applied. Then, the multiplication respective exponentiation is reduced to a simple addition.

3) Logistic models.

In many use cases, it can be assumed that natural limits exist which can be reached by a time series if observations are done over a longer period of time. For example, the growth of users in a network follows a function which is shaped like an S. In other words, these processes are constrained by saturation effects.

Formally, these time series can be described the following approach:

$$x_t = \frac{a_1}{a_2 + e^{-a_3 t}}$$

In this case the values of the time series converge to the saturation at the value G = a_1/a_2.

11.4.1.3.2 Trend estimation

The common principle behind the different presented approaches is to determine the unknown parameters a_i. Generally, this is done by estimating the minimization of a squared expression based on a difference. The difference is built by comparing the measurement value x_t with the valid approximation given by the chosen approach. Afterwards, the resulting difference is squared and summed up. For the polynomial approach, the according overall expression reads:

$$Q = \sum_{i=1}^{N} \left(x_t - a_1 m_1(t) + a_2 m_2(t) + \ldots + a_k m_k(t)\right)^2 \rightarrow \text{Minimum}$$

Now, the task is to minimize the expression for Q.

To solve the minimization problem, partial derivatives are calculated. In detail, Q is derived with respect to each of the parameters a_i:

$$\frac{\partial Q}{\partial a_i} \overset{!}{=} 0,\ i = 1, \ldots, n$$

A system of so called normal equations is the result of this calculation. Under the assumption of linear independency of the different functions m_i, a closed solution exists so that the different parameters a_i can be identified.

Related to non-linear models, the minimization approach leads to normal equations that cannot be solved explicitly. A further leading approach which might solve these problems is based on linear approximations and is called the Gauss-Newton procedure. Additional information can be found in [b-Bates].

Figures 11-28 and 11-29 show respectively examples of a linear model and a polynomial model.

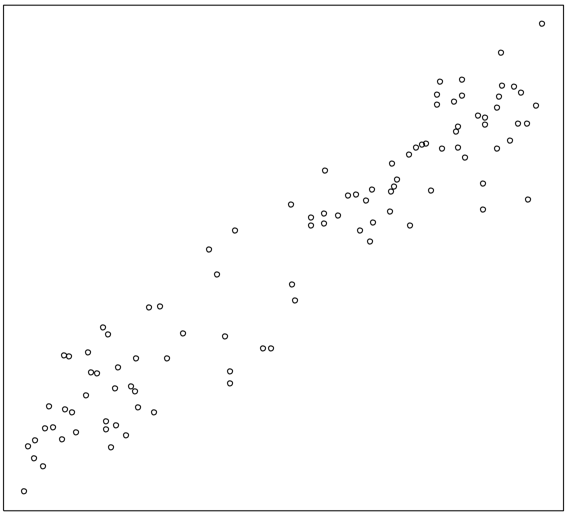


Figure 11-28 – Example of a linear model

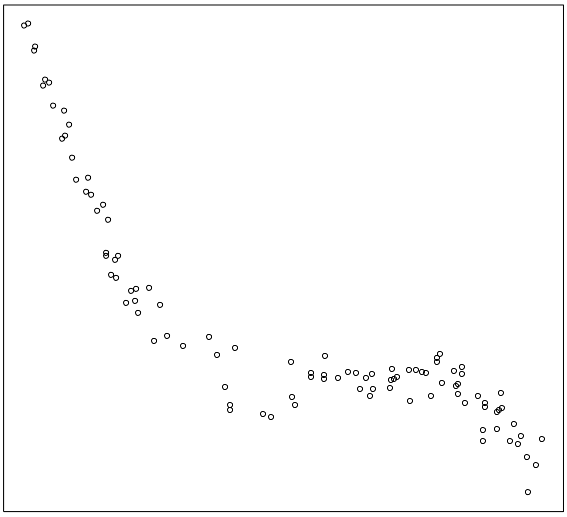


Figure 11-29 – Example of a polynomial model

11.4.1.3.3 Transformation of time series by filtering

Besides the determination of global trends the identification of local trends within the time series is important. The identification of local trends corresponds to the smoothing of a time series by applying filtering functions. One main advantage of this procedure lies in the fact that low-ordered polynomials already lead to acceptable results. This simplification reduces the required computational power.

On the other hand, the main disadvantage of this method is caused by the possibility to get a high number of describing parameters without finding an easy to handle closed solution or model description. In other words, the outcome of this approach may be a smoothened time series, but no model description.

11.4.1.3.3.1 Linear filters

A very simple approach to reach smoothing effects is the application of a sliding window to a time series. This corresponds to the calculation of a moving average. In general, this approach can be formally described as follows:

A linear filter L is a transformation of a time series x_t into a time series y_t with respect to the relation:

$$y_t = Lx_t = \sum_{i=-q}^{s} a_i x_{t-i} \quad i = s+1, \ldots N-q$$

Where, (a_{-q}, ..., a_s) symbolize different weights.

The simplest approach is done by "simple moving average". According to the notation given above, it reads:

$$a_i = \frac{1}{2q+1}, \, i = -q, \ldots q$$

The smoothing effect increases with the number of values taken into account which is the case for increasing values of q.

Reducing the condition for the weighting parameters of the filter to the standardization $\sum a_i = 1$, it is possible to prove that the local approximation based on polynomials is equivalent to the filtering method:

1) The simple moving average is the same as a local trend estimation of the data (x_{i-q},..., x_{i+q}).

2) The filter represented by the polynomial:

$$y_t = \frac{1}{35}\left(-3x_{t-2} + 12x_{t-1} + 17x_t + 12x_{t+1} - 3x_{t+2}\right)$$

represents a local trend estimation according to the squared minimization approach which is based on a polynomial of second order.

Figures11-30 to 11-32 show examples for filtering with linear and polynomial filters.

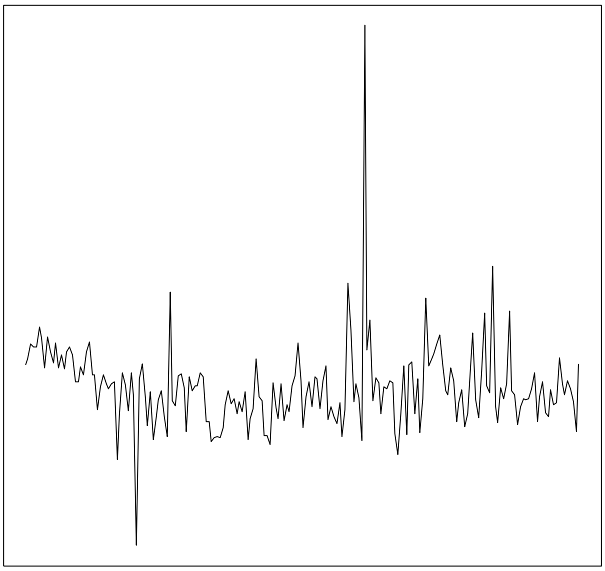


Figure 11-30 – Example of linear filter with q = 7

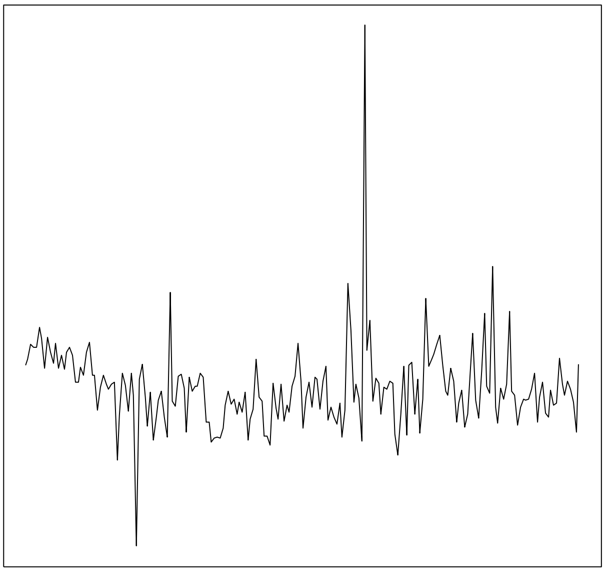


Figure 11-31 – Example of linear filter with q = 20

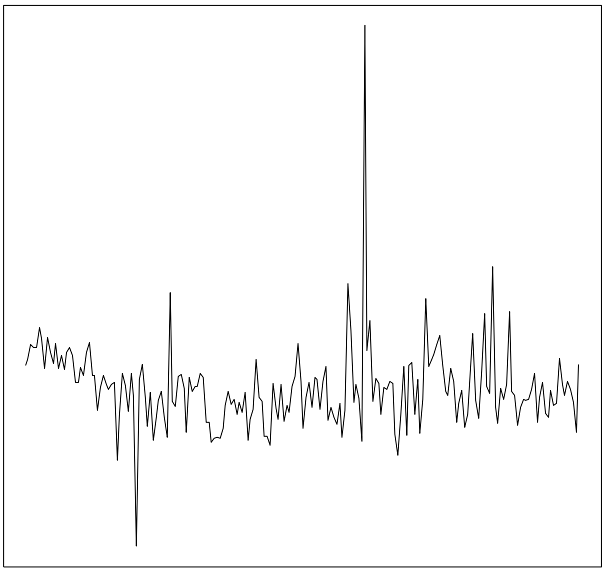


Figure 11-32 – Example of polynomial filter

11.4.1.3.3.2 Exponential filters

Linear filters always use a constant number of weights. Furthermore, a different approach can be interesting which takes into account that older values may be less interesting than newer ones. This is realized by decreasing the weights of older values whereas newer values lead to a higher weighting and is known as an exponential approach. This approach reads in recursive description:

$$y_{t+1} = (1-a)\sum_{i=0}^{\infty} a^i x_{t-i}$$

and is equivalent to the formula:

$$y_{t+1} = ax_t + (1-a)y_t$$

Both expressions are stated in such a way that they can be read as a prediction for the next point of time.

From this equation it can be seen that exponential smoothing also overcomes another limitation of moving averages: older values are weighted with decreasing weights. That is, since a is a number between 0 and 1, the weights [a, $a(1\text{-}a)$, $a(1\text{-}a)^2$, etc.] show a decreasing magnitude. These are the reasons why exponential smoothing has gained such wide acceptance as a forecasting method.

Rearranging the terms in the above equation would result:

$$y_{t+1} = y_t + a(x_t - y_t)$$

In this form, the new forecast is simply the old forecast plus a times the error in the old forecast ($x_t - y_t$). It is evident that when a has a value close to 1, the new forecast will include a substantial adjustment for any error that occurred in the preceding forecast. Conversely, when a is close to 0, the new forecast will not show much adjustment for the error from the previous forecast. Thus, the effect of a large and a small a is analogous to the effect of including a small number of observations in computing a moving average versus including a large number of observations in a moving average.

Figures 11-33 and 11-34 show examples of exponential filters with a = 0.1 and a=0.5 respectively.

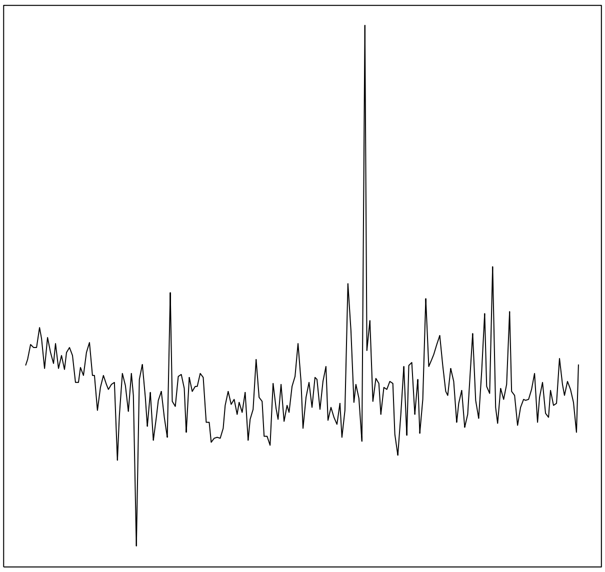


Figure 11-33 – Example of exponential filter with a = 0.1

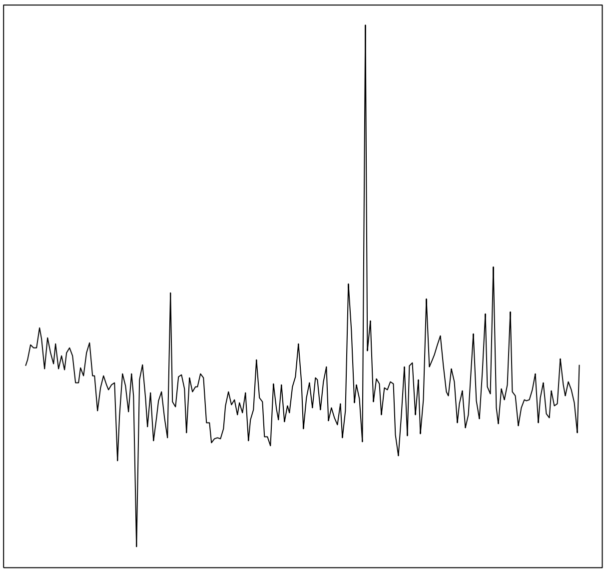


Figure 11-34 – Example of exponential filter with a = 0.5

11.4.1.4 Seasonal component

Data series which are related to the user's usage essentially contain seasonal figures, which imply cyclical fluctuations with regular characteristics. Interesting intervals in this area are yearly, monthly and daily periods.

Based on the two following views, it may be possible to eliminate the seasonal influences in the available data. Considering practical issues, the latter view is mostly preferred.

- Retrospective view: How would the data have been in the past if no seasonal influences were overlaid?
- Prospective view: What is the long term tendency with respect to the smooth component?

As an example of all the different possible procedures, the so called "mean phase" procedure is explained. This procedure is one of the easiest of the available procedures. It is suitable to achieve the elimination of a fixed seasonal component of a time series without a trend component. This means within the data of the time series no trend component is allowed. It must have been removed before by a trend elimination mechanism.

The procedure can be subdivided into four different steps. Generally, it is assumed that the time series $x_1, ..., x_N$ can be separated in different parts, the so called phases *p*, each with a length of *n* data elements. Formally, this relation is given by:

$$(x_t), t = 1, \ldots N \rightarrow (x_{i,j}), i = 1, \ldots p, j = 1, \ldots, n$$

The first index *i* describes the number of the segment or the phase whereas the second index *j* represents the consecutive number within this phase. For example, if a data series contains data of a period of 5 years on a monthly basis, it can be described by the parameters *p* = 5 (representing 5 phases, each for one year) and *n* = 12 (representing the 12 months within each year).

The following calculations implement the already mentioned four steps to achieve data without underlying seasonal impacts:

1) Calculation of the average of different phases:

$$\bar{x}_j = \frac{1}{p} \sum_{i=1}^{p} x_{i,j}$$

2) Calculation of the total average:

$$\bar{x} = \sum_{j=1}^{n} \bar{x}_j$$

3) Calculation of seasonal indices (seasonal factors):

$$s_j = \frac{\bar{x}_j}{\bar{x}}$$

Related to the last mentioned calculation, the averaged phases $\bar{x}_j$ are set into relation to the total average $\bar{x}$. In the example, the average of January is for example divided by the total average. This step is done for all of the different monthly averages. If January data is much above average, then $s_i > 1$ is the result, and if January data are much below average, $s_i < 1$ will be the result.

4) Calculation of seasonally adjusted values:

$$y_{i,j} = \frac{x_{i,j}}{s_j}, i = 1, \ldots p, j = 1, \ldots n$$

This step concludes the basic calculation scheme related to the determination of the seasonal component.

11.5 Data aggregation

Depending on the objective, i.e., for monitoring or reporting purposes, different types of aggregation may be of interest. First, different granularities in time and space may be needed. Second, weighting of data may be considered for instance to generate the QoS perceived by the user. This may be more or less straightforward, if the smallest time granularity is evenly spaced and the full information at any time point or interval is available, i.e., there are no missing data. However, data aggregation becomes more challenging if the event data are considered, like data from active probing systems, or if data are missing and substitution algorithms are needed for defining meaningful results at higher aggregation levels.

In the following, after presenting some basic aggregation operators, different data sources and corresponding structures are distinguished, temporal and spatial aggregation levels are defined and estimation methods for incomplete data are suggested. Based on a summary of desirable attributes of aggregation procedures, an aggregation strategy is suggested and discussed. Subsequently, weighted aggregations are motivated and weighting methods are introduced.

11.5.1 Basic data aggregation operators

Most common data aggregation operators are **sums** and **means**. Sums are applied if the total number of events, usage, etc., within a given time period is of interest, while means are commonly applied if some kind of averaging is needed. However, in particular for data out of time-interval measurements, means may not lead to interpretable and convincing aggregation results, therefore other summarizing statistics like the minimum, maximum, median and other quantiles of data are also used as aggregation operators. For quality of services measures like accessibilities, retainabilities and so on, ratios in the sense of fractions are used.

The combination of aggregation operators in a row might raise problems, even if the same operator is used at each single step. If the data basis is complete, the combination of sums on different aggregation levels, that is summing sums of lower levels for a higher level aggregation result, ensures a meaningful interpretation. If the considered time intervals are equidistant in addition, the same holds true for the combination of means on different aggregation levels. Minimum and maximum operators are also examples where this kind of combination is possible and meaningful. However, for other quantiles like the median or Q95, it is not recommended to base aggregations on higher levels on the results of lower aggregation levels, since for instance the median of subgroup-medians is not guaranteed to be near the overall median. Aggregation methods for fractions will be discussed later.

If different aggregation operators are combined one after another, the resulting values should be interpreted with great care. For instance the minimum and maximum of mean values from lower aggregation levels should not be mistaken as the range of the original data values. However, one can think of many examples where this kind of combination yields meaningful results that are of interest, for instance if different base station controllers (BSCs) are compared with regard to QoS parameters, the median or quantiles of the values for all BSCs may be used as cut-points to identify BSCs performing particularly good or bad.

11.5.2 Data sources, structures and properties

In the following, a distinction between raw data that may result from data sources with different attributes and parameters that are defined based on these raw data is made.

11.5.2.1 Raw data

For measuring QoS, data from a number of different data sources are used. These "raw data" are based on different measurement systems with possibly different granularities, differences due to release changes of the underlying systems and so on. Therefore, raw data often come with a number of attributes that need to be taken into account for aggregation. Here performance data and event data are considered, although other data sources and types could also provide valuable information about QoS (like fault management data).

In the situation that not all data are available – which is a common problem not only in mobile communications – raw data are rarely substituted or adjusted, but are stored with the full source information to allow suitable parameter definition and estimation. This is often only possible by applying all available reference knowledge, for instance which or how many cells were supposed to deliver data for a given time period.

11.5.2.1.1 Performance data

Most performance data are given by network element (NE) counters. Due to different underlying systems or releases, these may be available in different temporal granularities, like values for 15 minutes from one system and values for hours from another system, respectively. Here basic operations are needed to enable a common data format in order to ensure that values are comparable and basic aggregations are needed for total NE counter values independent of underlying systems. In addition to results of basic aggregations, the total number of aggregated values or even additional detailed reference information needs to be stored.

Examples for performance data are the number of calls per hour per cell, the total number of call minutes per quarter-hour per cell or the number of started WAP-sessions per hour per cell.

11.5.2.1.2 Event data

Billing data are one example of event data that may provide information about the QoS. On the other hand, results from active probing systems are also in general not equally spaced over time. This may be due to varying duration of trials, possibly depending on network performance or other influences. Also there may be reasons to do a larger number of tests regarding a specific service over a given period of time, for instance if new services are launched.

Event data do not provide information of a time interval, but just a number of results, each for a corresponding point in time. To allow statements about time periods, it is either possible to use all original data points for defining and aggregating parameters for each time interval of interest, or as an alternative, relatively small time intervals have to be defined for which a first, "basic", aggregation step is carried out which then allows higher aggregations independent of the original data.

11.5.2.2 Key performance indicators/parameters

A particular feature of a key performance indicator (KPI or parameter) – in comparison to raw data – is given by the fact that KPIs are defined in an abstract manner, thereby allowing a common interpretation for parameters computed from different data sources and on various aggregation levels. Usually, there are two possible reasons for a parameter to be identified as a KPI, either:

- the KPI is a function aggregation of different parameters; or
- the KPI represents a very important quality measure related to the user's perspective.

In the latter case, data aggregation is not necessarily implied.

Parameters are defined to serve specific purposes like answering questions about QoS or performance by utilizing raw data or basic aggregations of raw data. This might also include combinations of different data by mathematical operations like ratios. Unlike raw data, parameters are independent of the underlying software releases or system vendors. One could also define them as being independent of different underlying systems, if appropriate.

Parameters based on performance data are for instance the cut-of-call ratio, which is based on two different NE counters, namely the number of unintentionally terminated calls divided by the number of successful call attempts times 100%. Data from active probing systems allow the definition of parameters like the recharging-failure-ratio, SMS-E2E-failure-ratios and so on.

For the definition and computation of parameters, rules for handling missing data are needed. Therefore, methods for data substitution become a major point when talking about parameter computation and aggregation and will be covered in some detail after defining aggregation hierarchies of interest.

11.5.3 Aggregation hierarchies

Aggregation hierarchies are commonly divided into temporal and spatial aggregation hierarchies, where temporal in fact refers to different time intervals while the term spatial may also cover aggregations over items with similar properties with respect to an attribute of interest.

11.5.3.1 Temporal aggregation

Temporal aggregation levels that should be used for a given parameter will depend on the intended use of the parameter as well as on the raw data frequency. For most parameters, sensible aggregations levels will be some or all of the ones given in the following set:

- quarter-hour;

- hour;
- day;
- week;
- month;
- quarter year;
- calendar year; and
- business year.

In addition, incomplete temporal aggregation levels can be defined, for instance cumulative values for the current week based on the daily values that are available so far. This is of particular interest for parameters that are based on ratios or mean values because these may be interpreted directly. For the interpretation of incomplete parameters that are based on sums, the number of covered time units has to be taken into account.

11.5.3.2 Spatial aggregation

Spatial aggregation originally refers to aggregation levels from the smallest possible units like radio cells (or even sectors) in mobile communications up to the entire network. This can be done from a technical point of view for instance by taking the order "cell – BSC – MSC – (branch) – entire network", or from a regional point of view by "cell – state/region – entire network".

As mentioned before, the term "spatial" may also be used in the context of summarizing cells with similar properties regarding an attribute of interest, like all cells that cover motorways, or the position of a cell in terms of the surrounding area, whether it belongs to a large city, a small town or a rural area. In these cases, spatially incoherent units are aggregated.

11.5.4 Parameter estimation methods

The ideal situation of full information is rarely met in practice. Network elements or branches failing to deliver data in time are common reasons for missing data. Since in most situations missing values as parameter values are unacceptable, even if parts of the raw data are missing, data estimation methods are needed. Depending on the situation, projection, substitution or combined estimation methods are suitable.

11.5.4.1 Projection method

The easiest method of data substitution is to project the available data to the entire time interval of interest. For example, if a fraction of 90 per cent of the expected data measuring the quality of a specific service within one hour is available and an hour-value is sought, these 90 per cent of the data are viewed as being representative for the hour of interest. If the aggregation contained of cumulating the entries, the value achieved by the available data has to be multiplied by a factor 100/90. If aggregated values are mean (or median) values, the mean (or median) of the available data is used as an aggregated value. If minimum or maximum values (or other quantiles besides the median) are sought as aggregated values, more sophisticated estimation methods should be applied like maximum likelihood methods.

Provided that a high percentage of data is available and there are no systematic reasons for missing data, the above procedure is sufficiently reasonable. However, an analysis of the times and circumstances of missing data might be of interest to identify the reasons, if missing values are becoming more frequent or appear suspect that there might be an underlying pattern.

If only a low percentage of data is available for a time period of interest, for instance less than 50 per cent, the above projection procedure is more than questionable. In particular if the parameter of interest is subject to dynamic changes over time, the results may be heavily biased. As an example, consider usage data where a high percentage of missing data consists of records that should have been taken at night. If the usage is then estimated by projecting the available results, the usage is extremely overestimated. Therefore, it seems sensible to define a "critical percentage" of data that need to be available for applying the above method. This percentage should depend on the specific service which is evaluated and on the needed precision of the aggregation results.

11.5.4.2 Substitution method

If the estimation of parameters can or should not be based on the available data (i.e., a large number of data is missing or the data are not credible for some reason), substitution methods would apply to values from the previous time periods. This can either be done by using the last available value for the parameter of interest, which is only sensible for static parameters that are not subject to dynamic changes over time, or by using the last available value of a comparable time period with similar properties like the same day-type (weekday/Saturday/holiday or Sunday) or the same day of the week and the same time.

11.5.4.3 Application of estimation methods

Common problems that complicate the application of the methods suggested above are given by:

1) Unavailability of reference data: The number of missing data is needed for deciding which method should be used and for the application of the projection method.
2) Determination of values for substitution: Comparable time intervals have to be defined and substitution values may be stored in a database, which needs to be updated and maintained. In addition, calendar information about holidays/working days, weekdays and so on is needed.
3) Parameters that are defined as ratios: Either numerator and denominator are estimated separately based on the available information for each part, or the ratio is estimated as a whole by using only data with information about both, numerator and denominator. In the situation of full information, there is no difference between both possibilities, in case some data are available for one part of the ratio and not for the other, both strategies yield different results.

Referring to data aggregation, the question arises, at which aggregation level estimation should take place. Is it acceptable to use estimated values as a basis for higher aggregations? Data aggregation procedures combining both introduced methods are derived in the following, originating from a summary of desirable attributes of aggregation procedures.

11.5.4.4 Attributes of aggregation operators

Aggregation methods may be evaluated according to the following attributes:

1) The result should be meaningful, that is as near as possible to the true value of interest. In particular, NULL-values are not sensible as a result of a higher level aggregation. In addition, all information about missing values should be used to take non-available data into account, to avoid biased parameter values. (Moreover, the variance of parameter values caused by estimation methods should be as small as possible.)
2) Aggregation results should be reproducible and understandable. In particular, at higher aggregation levels no estimation procedure should be used so that results on a higher aggregation level are in accordance with values of the underlying level.
3) Aggregation results should not depend on the used aggregation paths, i.e., there should be no difference of results, if spatial or temporal aggregation steps are interchanged as well as direct aggregation and aggregation with intermediate steps should not lead to different results. Independence of paths refers to aggregation calculations.
4) Results should be consistent. On a given aggregation level, individual aggregation results should agree with total result, i.e., the sums for different branches should add up to the total sum for the company and so on. Consistency refers to aggregation results.

5) The applied calculation procedures should be rather easy. This also implies independence of the past values like those from the previous time periods.

6) Independence of network references like assignment of results to network elements.

In general, it is not possible to meet all requirements at the same time. Easy methods may lead to non-sensible results, while methods that contain estimation procedures often rely on values from the previous time periods or network references and may be more sophisticated. In particular the requirements 1 and 5/6 are contradicting as estimation methods ignoring network references and past values will presumably often lead to worse results compared to methods that take into account all available information.

One idea to combine the above requirements to a certain extent is to define a smallest temporal and spatial aggregation level for which the data are completed by estimation procedures (for missing parameter data) or basic aggregations (for event data and parameter data with different lowest levels), like per hour per cell. This yields an equally spaced "complete" data pool and therefore simplifies all further aggregation steps and in particular ensures consistency of results and independence of aggregation paths.

One major disadvantage of this method is the fact that estimation procedures have to be applied on low aggregation levels which rely heavily on reference data and good substitution values or projection procedures. For parameters that are dynamic over time, time series methods as covered in clause 7 should be considered, which then implies more complicated calculation procedures for low aggregation levels and therefore might take some computation time.

11.5.5 Weighted aggregation

In many situations, in particular if the QoS perceived by the user is of interest, simple aggregations of the available information or estimated values are not very meaningful. A better approach would be to take into account, e.g., how many users are affected if a service fails. That leads to the idea of weighted aggregation, where for instance the usage can be applied for weighting, respectively. It should be noted, however, that weighted aggregation methods will in general lead to non-consistent results in the sense of property 4 from clause 11.5.4.4.

11.5.5.1 Perceived QoS

Depending on the point of view and the corresponding intention of a parameter of interest, it appears reasonable to only consider users view instead of the network view, e.g., by taking the usage of a service into account. Depending on the applied aggregation procedure, this may have already been done implicitly. For instance, if the aggregated cut-off call ratio for a particular week is considered, different aggregation procedures imply different weightings (it is assumed that values are available hourly):

1) If the cut-off call ratio is stored for hourly intervals and the weekly aggregation is done by averaging all values for the week of interest, no weighting is carried out and each hour viewed as being equally important. This does not correspond with the user's perspective. (Note that a geometric rather than an arithmetic mean should be applied for averaging ratios.)

2) If the numerator and denominator are stored for hourly intervals and the weekly aggregation is done by first summing all entries for the numerator and denominator separately and then dividing both numbers, an implicit weighting is carried out. Since high usage time intervals contribute a larger number of call attempts than low usage intervals, thereby the user's perspective is expressed.

When applying the first method, one should consider using weighted means instead of simple means. Depending on the type of parameter, weighted arithmetic means are computed according to:

$$\bar{x}_w = \sum_{i=1}^{n} x_i w_i$$

where $\sum_{i=1}^{n} w_i = 1$, weighted geometric means are given by:

$$\tilde{x}_w = \left(\prod x_i w_i\right)^{1/n}$$

where $\prod w_i = 1$. (For unweighted arithmetic or geometric means, all weights are given by $w_i \equiv 1/n$ or $w_i \equiv 1$, respectively.)

Weights can either be based on true or average usage curves, but also on strategic reasons or any other procedure in accordance with the aim of the analysis. An average usage curve may for instance be achieved by averaging over the past 30 working days, the past five Saturdays and the past five Sundays or holidays or by applying some kind of time series modelling and forecasting methods. Weights based on usage curves are then computed as:

$$w_i = \frac{u_i}{\sum u_i} \text{ or } w_i = \frac{u_i}{\prod u_i}$$

respectively, where u_i is the true or estimated usage within time period i.

If the second method from above is applied, weighting is done implicitly with the actual usage curve. However, other problems arise in particular regarding missing data handling as mentioned in clause 11.5.4.3. For each time period, the percentage of missing data might be of interest for applying projection or substitution methods and to ensure that the cut-off call ratio does not exceed 1, e.g., the number of unintentionally terminated calls should not exceed the number of successful call attempts, one might only want to consider data pairs where both numbers are known. When using the first method, this could be avoided by estimating only on an hourly basis.

Remark: For ratios, higher level aggregations are commonly achieved by applying the second method because of the implicit weighting, which is more intuitive.

Data from active probing systems are generally not weighted implicitly, since probing frequencies are commonly non-representative for user behaviour. In this context, the idea of weighting might even be of importance in more than one respect.

1) Since data from active probing systems are not equally spaced, a weighting of each trial result by the time between two trials in some way could be considered. This can either be realized by defining (rather small) time intervals for which all trials done within this interval are summarized without weighting or alternatively by computing using half of the time interval between the last trial and the current one and half of the time interval between the current trial and the next one as a basis for weighting. If such weighting is considered, an upper bound for the defined underlying intervals should be considered and strategies for the situation that the active probing system does not work or data are not delivered for a longer time period are to be thought of (estimation or NULL-values, depending on the situation and parameters of interest).

2) A second – and probably more important – way of weighting results from active probing system is the usage weighting for achieving the perceived QoS as explained before.

If both types of weighting are applied, combined weights are computed as $w_i = \frac{u(t_i)}{\sum u(t_i)}$, where $u(t_i)$ is the usage within time period t_i assigned to probing trial i according to the distribution of trials over time (1st weighting), either for a basic aggregation level for further aggregation or for the desired aggregation level directly.

11.5.5.2 Weighted quantiles

For duration values as results from active probing trials, quantiles represent meaningful aggregation results on higher aggregation levels. From the above weighting discussion the necessity of determining weighted quantiles arises. Due to the calculation of quantiles based on ordered data values, a weighting similar to those for mean values is not applicable. Instead, a **replication** algorithm could be used for computing weighted quantiles. This algorithm simply repeats each value according to an assigned weight and calculates the desired quantile of the resulting new data set. (If weights are irrational, sensible rounding is needed.)

Example: The original (ordered) data set of ten MMS-E2E delivery times is given by 51, 55, 60, 61, 65, 70, 71, 72, 72, 80 seconds. These measurements have been taken at different times of the day and therefore get a weight of 1, if taken at night, 2, if taken in the morning or late in the evening and 4, if taken between noon and 8 p.m. for instance. Accordingly, weights are therefore given by 1, 4, 2, 2, 1, 4, 4, 2, 1 and 4 resulting in a data set with 25 data: 51, 55, 55, 55, 55, 60, 60, 61, 61, 65, 70, 70, 70, 70, 71, 71, 71, 71, 72, 72, 72, 80, 80, 80 and 80. Quantiles from the original and the replicated data set will in general lead to slightly different results.

If weighting according to some kind of usage curve is aimed, this curve can be used as a replication function or replication curve and represents the basis for defining the needed weights. To simplify the computation, weights may be defined by identifying the minimum of the replication function $r_{\min}$ and to define the weights according to:

$$w_i = round\left({r_i}/{r_{\min}} \right)$$

If a uniform concept for weighting of any kind of parameter is sought, the approach based on replication functions might also be used for means or non-accessibilities or other parameters of interest. Differences for instance between conventionally weighted means and means weighted by replication curves are only due to the applied rounding step for the latter approach.

11.5.6 Additional data aggregation operators

In the following, some additional data aggregation operators are covered, adding to those mentioned in clause 11.5.1, that are in some sense individual regarding their attributes and/or applications.

11.5.6.1 MAWD and BH

In particular for network maintenance an aggregation operator of interest is the monthly average working day (MAWD). This operator can be viewed as being an aggregation result as well as a possible weighting function for other aggregation operators.

The monthly average working day of a given data set is computed by filtering all data from working days within a given month first and then computing the mean value for each hour of the day over all data from corresponding hours. The result is therefore given by a vector of 24 entries, each corresponding to one hour of the day (0000 hour – 0100 hour, ..., 2300 hour – 2400 hour).

Based on the MAWD, the busy hour (BH) is defined as the hour in which the MAWD vector takes its maximum value. In mathematical notation, this is $argmax\ (m)$, where $m = (m_1, ..., m_{24})^T$ is the vector resulting from applying the MAWD operator.

11.5.6.2 AVGn

The class of parameters averaging operator regarding n days (AVGn) is applied for similar reasons as the BH operator mentioned in clause 11.5.6.1. Both try to identify peaks of high usage or traffic, where the BH operator considers hours where the highest usage is observed on average, while the AVGn-operators are interested in the maximum usage or traffic for a given calendar week. The mean of the n largest values realized on n different days (n between 1 and 5 or 7, depending on intended use) is defined as AVGn.

11.6 Assessment of performance indices

11.6.1 Estimation of performance parameters based on active service probing systems

End-to-end service probing systems yield valuable information about services and systems that may not be provided by the network elements alone. Active probing systems are used to view the user's perspective of the quality of service, i.e., the perceived QoS. Typical parameters that may be computed based on active probing systems are failure ratios, accessibilities and end-to-end-delivery times for a number of different services.

One characteristic of active probing systems is that the tests are often done more or less equally distributed over time for utilizing the equipment as exhaustingly as possible. In this respect, they fail to reflect the user's perspective, since derogation during the day will be a lot more severe than after midnight due to lower volume of traffic for almost all services at night.

From a statistical point of view, end-to-end active probing systems try to estimate the real behaviour of a service by taking a defined number of samples. Therefore, the results of measurement activities have to be interpreted as the current view on a certain service and need not necessarily represent user experience. Depending on the number of considered samples, the connection between observed measurement results and unknown real behaviour may vary.

11.6.2 Monitoring concepts

To ensure that network problems are noticed and remedied as quickly as possible, monitoring concepts based on active probing results are important tools for an efficient alarming system. Such monitoring tools may be based on control charts or other alarming rules.

11.6.2.1 Control charts

Control charts are based on the assumption that if the service under study works properly, data gained from active probing systems follow a stable distribution with the given parameters. From previous observations, the underlying distribution may be identified and parameters have to be estimated. Control charts are now set up based on statistical considerations such that in case of network problems, i.e., the process is "out of control", an alarm is created. On the other hand, false alarms have to be avoided, that is as long as the process is "in control", no alarming should occur.

Control charts generally visualize the development of a quality characteristic of interest over time similar to a line diagram as shown in Figure 11-22. Further, a target line and control and or warning limits are added to the plot. The target line represents the line around which data values are expected. Warning and control limits may be used to define identifying realizations indicating that the process it "out of control". Different types of control charts were invented for different types of data.

11.6.2.1.1 Shewhart control charts

If data are normal or mean values are considered (central limit theorem, compare section), Shewhart charts [b-Nist 1] for normal data may be applied. In this case, the current short-term data is compared against an underlying data model which represents the long-term behaviour. Based on this model, it is possible to define the usual or "normal" situation. This is required to pay attention to unusual situations. Shewhart control charts are widely used in different sections of the industry.

11.6.2.1.2 CUSUM and EWMA charts

Two other approaches can be used to introduce some kind of weighting into control charts. The cumulated sum (CUSUM) approach [b-Nist 2] uses sums data up over time and therefore indicates the behaviour over a greater period of time. A slightly different approach is represented by exponentially weighted moving average (EWMA) charts [b-Nist 3] where older values gain less influence than newer data does.

11.6.2.2 Other alarming rules

Furthermore, the deviation between the long-term data model and the short-term monitoring data should lead to consecutive actions if a critical state is reached. This relation is defined as "alarming rules". One example for the alarming rules is the set of the Western Electric Company (WECO) rules [b-Nist 4].

11.6.3 Methods for evaluation of objectives

Commonly objectives are formulated in terms of target values for important parameters. Then the evaluation of objectives could mean to assess to which extend these aims have been achieved within a given time-period (i.e., month or business year). If there is only one important parameter, this is a rather easy task. However, if a number of pre-defined parameters are to be combined in an overall measure and in addition different groups (i.e., branches or departments) are to be compared regarding their performance, the main issue for evaluation will be to define a common measurement scale for all parameters. This allows the combination to an overall evaluation index of some kind and thereby a comparison of groups is facilitated.

In the following, two methods are described that allow the evaluation of objectives, namely the desirability approach and the loss function approach. Both approaches rely on definitions of target values and specification limits for the parameters. In this context, parameter values are denoted by $y_i, i = 1,\ldots,P$ and target values are denoted by $\tau_i, i = 1,\ldots,P$. Specification limits are given as upper specification limit (USL_i) and/or lower specification limit (LSL_i) for each parameter under consideration $i = 1,\ldots, P$. (It might also be sensible to consider lower and upper target values, if the target is given as an interval instead of a numerical value.)

11.6.3.1 Desirability functions

Desirability functions use a transformations of the values y_i to the interval [0, 1] based on system requirements by defining a transformation function based on target values and specification limits. Desirability functions are piecewise defined continuous linear functions where desirability values of 0 are assigned to parameter values y_i outside the specification limits, realizations on target get desirability values of 1 and outcomes between target and specification limits are assigned by a linear connection or a power transformation thereof.

The principle of desirability is best explained by providing example desirability functions as summarized in Figure 11-35.

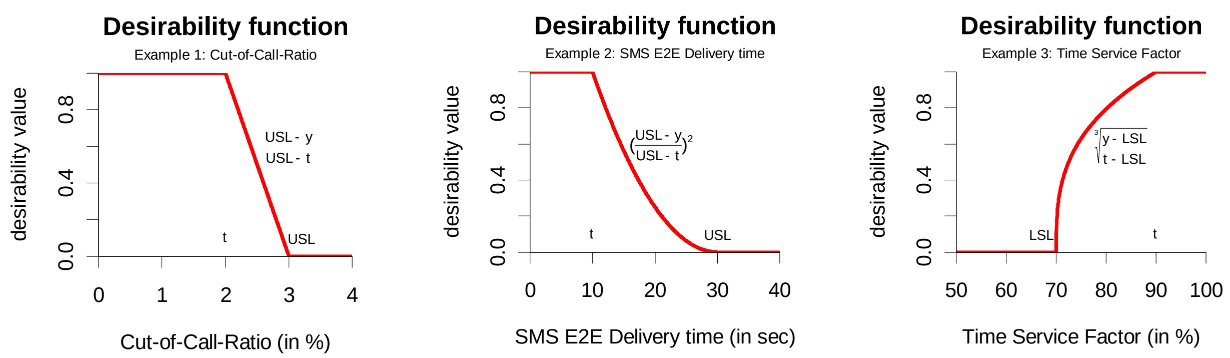


Figure 11-35 – Examples of different desirability functions

11.6.3.2 Loss functions

Loss functions in contrary evaluate a realized parameter value in terms of costs arising due to deviation from target or realization outside the specification limits. Therefore, values within the interval [0, ∞] will be achieved. The main issue for the specification of loss functions is the assignment of a rising costs. The loss of earnings if services are not fully usable may be stated rather easily, but quantifying the image loss and corresponding costs might be a much more difficult task.

For each parameter of interest, the arising loss for a value y_i is given by $L(y_i) = c(y_i - \tau_i)^2$ or alternatively $L(y_i) = c\min\left((y_i - USL_i)^2, (y_i - LSL_i)^2\right)$, where $c \in IR$ quantifies the arising cost.

Mainly, normal distributed values are in the focus if loss functions are discussed. In general, the area which is covered by the lower and upper tails of the normal (or Gaussian) distribution is in the main interest. These branches violate the guaranteed specification levels USL for the upper tail and LSL for the lower tail. All values in these areas represent defects referring to the observed process. The underlying theory specifies rules how to set the limits and how to proceed with asymmetric cases. One useful hint for further research in this area is the "six sigma approach" which is wide-spread in the industry.

12 Network based quality of service measurements

12.1 Network measurement basics

12.1.1 Point of control and observation (PCO)

The point of control and observation (from now on called "point of observation" or PCO) is the location where the measurement is actually performed. The location can be either inside the network or at the end-point. The measurements should be done using standardized interfaces and protocols.

Possible points of observation for QoS parameters covered in this Recommendation are:

- Inside nodes in the network (e.g., RNC, base station, switch, etc.)
- Observations in the terminal:
  - End-point test tool; or
  - Measurements that are reported back from the terminal to the network

Figure 12-1 shows points of observation, points of recording and measurement reporting.

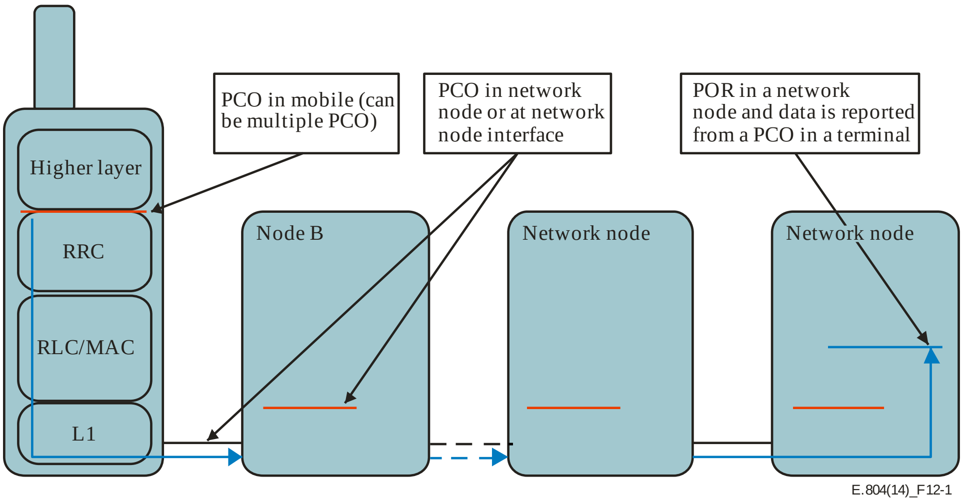


Figure 12-1 – Points of observation, points of recording and measurement reporting

12.1.2 Point of recording (POR)

The point of recording (POR) is where the QoS parameters are recorded. The POR can be the same as the PCO or another point inside the terminal or the network. If the PCO and the POR are not the same, the measurement data must be reported from the PCO to the POR. Examples of such reporting are described in Appendix XI.

12.2 Measuring QoS parameters in the network

12.2.1 General overview

The quality of service measurements should be done as much as possible in the same way inside the network as they are done at the end-point with a test tool. Many types of measurements can be done with the same trigger points (for instance, the reception of a certain protocol message) independent of the point of measurement, but the measurement result might differ slightly depending on where in the call or transmission chain the PCO is located.

12.2.2 Service accessibility QoS parameters

Accessibility QoS parameters reflects the ability to initiate or intentionally terminate a connection or a service. The parameters can be divided into the following groups:

- **Failure parameters**: Reflects the outcome of attempts to initiate a connection or a service. For mobile-originating (MO) cases the network might sometimes be unaware of some of the attempts, and network-based parameters can be expected to give a slightly more positive view of the network condition, as compared to the corresponding endpoint-test parameters. For mobile-termination (MT) initiation attempts the network is normally fully aware of these, and network-based parameters should therefore correspond well to the same parameters as measured by an endpoint test tool (assuming that MT initiation attempts are known and controlled by the endpoint test tool). As most initiation procedures require a successful two-way communication during the initiation phase, the accessibility parameters measured for MO and MT endpoint test calls should normally not differ too much, and thus the network-based parameters for the MT case can be seen as a good approximation of the total network state.
- **Time parameters**: Reflects the time needed to initiate a connection or a service. As these parameters are only defined for successful attempts the network can see the message flow, and can measure the time elapsed to initiate the connection or the service.

  The difference in parameter values as compared to the corresponding endpoint test measurements depend on where in the network the time measurements are done, but normally the time elapsed in the radio link and the processing time in the mobile are not included in the network-based parameters, making them more optimistic.

  If the excluded radio delay is stable or small compared to the total delay, the network-based measurements can still give a good picture on the state of the network.

  The estimated value of the excluded delay parts (for instance the radio delay) should be added to (or noted together with) the measured time parameter.

12.2.3 Service retainability QoS parameters

Retainability QoS parameters reflect the ability to retain, or keep a service up and running. Typical examples of retainability parameters are cut-off ratio and session failure ratio. Retainability parameters can be measured at the end-point but in general also inside the network. Measurements inside the network in general do not need any additional measurement data from the end-point.

12.2.4 Service integrity QoS parameters

Service integrity QoS parameters reflect the quality of a service that has been successfully set up and is in use. As the integrity parameters are only measured for successfully connected services, the network will always be aware of the ongoing service, and can measure its performance.

Depending on the type of service used, different types of integrity parameters are calculated:

- Media quality parameters
- Response time parameters
- Data rate parameters

12.2.4.1 Media quality parameters

In an end-point test scenario, the media quality parameters can be measured in the mobile device by using objective quality algorithms, for instance [ITU-T P.862.1] for speech quality measurements. For the network-based quality measurements other methods are used:

- Parameters related to the service quality can be measured at different nodes in the network and translated to a service quality parameter by using a parametric or bitstream quality model, like [ITU-T P.564] or similar model for video and multimedia service. The resulting quality parameter will reflect the state at the measurement nodes, and be an estimate of the end-user quality.
- Parameters related to the service quality can be measured in the mobile, and reported back to POR in a node in the network (see Appendix XI). The calculation node combines the reported parameters with other relevant information collected from the network about the ongoing service, and calculates the service quality parameter. The resulting quality parameter will be closely correlated to the end-user experience.

12.2.4.2 Response time parameters

Some services are characterized by real time events or real time interaction between the users, and the response time needs to be short enough to provide a good service experience. A typical service example is PoC which is dependent on short delays between user input and system acknowledgement.

The time elapsed between certain protocol events can normally be measured in different network nodes. Some parts of the response time (typically the radio part) will not be included in the network-based measurements, so the resulting response time parameters will only be an estimate of the end-user experience.

However, if the radio delay is stable or small compared to the total delay, the network-based measurements can still give relevant information about the end-user experience. It might also in some cases be possible to make a separate measurement or estimate of the radio delay.

Any estimated value of the excluded delay parts (for instance the radio delay) should be added to (or noted together with) the measured time parameters.

12.2.4.3 Data rate parameters

Data rate parameters, for services such as FTP, web browsing and e-mail, can be measured at different nodes in the network. Normally these services are using acknowledged radio bearers, and the network-based data rate measurements should thus be closely correlated to the corresponding endpoint test measurements.

12.3 Comparing network and end-point test measurements

Except for the differences due to different PCO for end-point tests tools and network measurements, there are other factors that affect the result. For instance the user behaviour might not be the same, as in the following example:

There is a radio network coverage problem for a series of tunnels in a newly built motorway area. Mobile users frequently get their calls dropped when they pass this area of the motorway. Initially when the motorway opened, the call drop rate measured in the network was high, since most of the calls were dropped. However, after some time the mobile users frequently passing this problem area will learn that the call will drop, and start to avoid making calls when passing that point. The drop rate in the network will go down but the problem has actually not been solved.

If the same area is also monitored by, e.g., automatic end-point test equipment the drop rate will not go down, since the end-point test tool calling frequency is not reduced due to the coverage problem.

The network measurement results and the end-point test tool measurements are both correct, yet they might have a very different drop rate. This illustrates that even if the same thing is measured in both cases, the results should not always be expected to be the same.

Another important difference between end-point test tools and network measurements is the possibility to get a good geographical position for the end-point test tools data, while network data typically is limited to cell level resolution. Thus even if network data in many cases can measure the same parameters, specific problems discovered in a cell might still need further support from endpoint testing to pinpoint where in the cell the problem occurs.

Appendix I

Examples for measuring trigger points

(This appendix does not form an integral part of this Recommendation.)

- SMS service:
  - Layer 3 messages:
    - Start SMS service attempt: generating random access (chan_request SDCCH) at mobile equipment.
    - Successful SMS service attempt receiving cp_data (rp_ack) at mobile equipment.
    - Receiving SMS on mobile equipment 2: receiving cp_data (rp_ack) at mobile equipment.

Additional examples are for further study.

Appendix II

Streaming explanations

(This appendix does not form an integral part of this Recommendation.)

Real time protocol (RTP)

The real time protocol (RTP) is used for the transmission of real time data, e.g., audio, video, simulation data over multicast or unicast network services. No QoS functionality is implemented.

RTP is designed to be independent from the underlying transport and network layers. For a complete description refer to [b-IETF RFC 3550].

Real time control protocol (RTCP)

The real time control protocol (RTCP) is known as the control protocol for the RTP. It allows the monitoring of the data delivery and provides a minimal control and identification functionality. RTCP is designed to be independent from the underlying transport and network layers.

For a complete description of the RTCP refer to [b-IETF RFC 3550].

Real time streaming protocol (RTSP)

The real time streaming protocol (RTSP) is used for the overall control of the streaming session.

For a complete description of the RTSP refer to [IETF RFC 2326].

Most important methods of RTSP:

- DESCRIBE: The DESCRIBE method retrieves the description of a presentation or media object identified by the request URL from a server. It may use the Accept header to specify the description formats that the client understands. The server responds with a *description* of the requested resource. The DESCRIBE reply-response pair constitutes the media initialization phase of RTSP [IETF RFC 2326].
- SETUP: Causes the server to allocate resources for a stream and start an RTSP session [IETF RFC 2326].
- PLAY: Play is sent from the client to the server and informs the server to start the transmission of data as specified by the SETUP method [IETF RFC 2326].
- PAUSE: Sent from client to server. Temporarily halts the stream transmission without freeing server resources. These resources can only be freed after a specified time [IETF RFC 2326].
- RECORD: This method initiates recording a range of media data according to the presentation description [IETF RFC 2326].
- TEARDOWN: Frees resources associated with the stream. The RTSP session ceases to exist on the server [IETF RFC 2326].

The following syntax for the hyperlink is used in order to access streaming content on the server:

protocol://address:port/path/file

Protocol	Used protocol. E.g., rtsp://
Address	Address of the used streaming server
Port	Port used by the server for answering request
Path	Path to the file to be streamed
File	The streaming file to be reproduced and its extension

Appendix III

Push to talk over cellular information

(This appendix does not form an integral part of this Recommendation.)

Figures III.1 to III.4 visualize signal flows of typical PoC sessions. The figures include the signal flows on the transport layer as well as some restricted information on the application layer. To keep the flows concise, some signals are not shown. So, it is possible to obtain signal flows universally valid for different kinds of PoC sessions. Figures III.1 to III.4 also show particularities using unconfirmed indication with media buffering as well as differences between pre-established and on-demand PoC sessions.

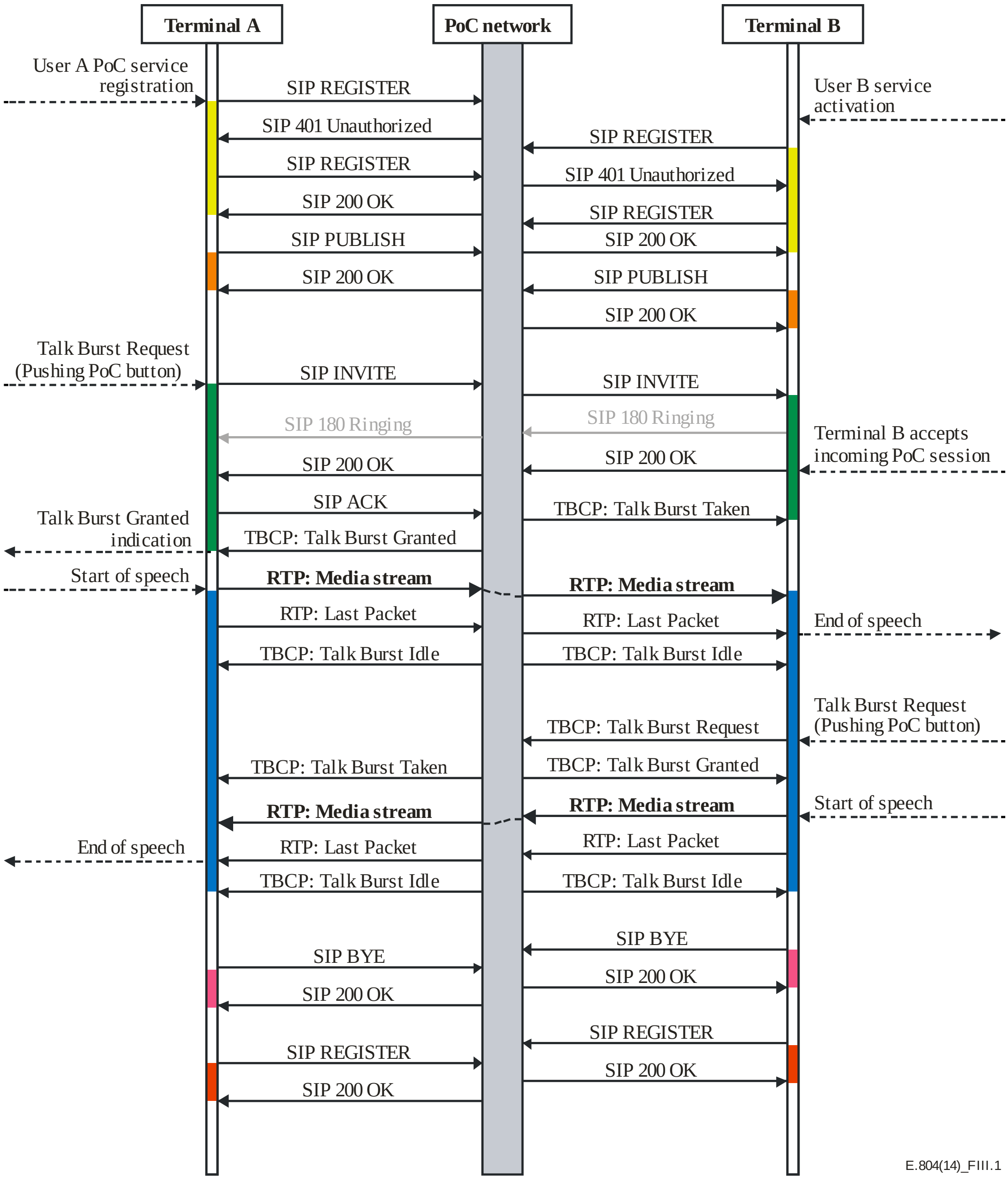
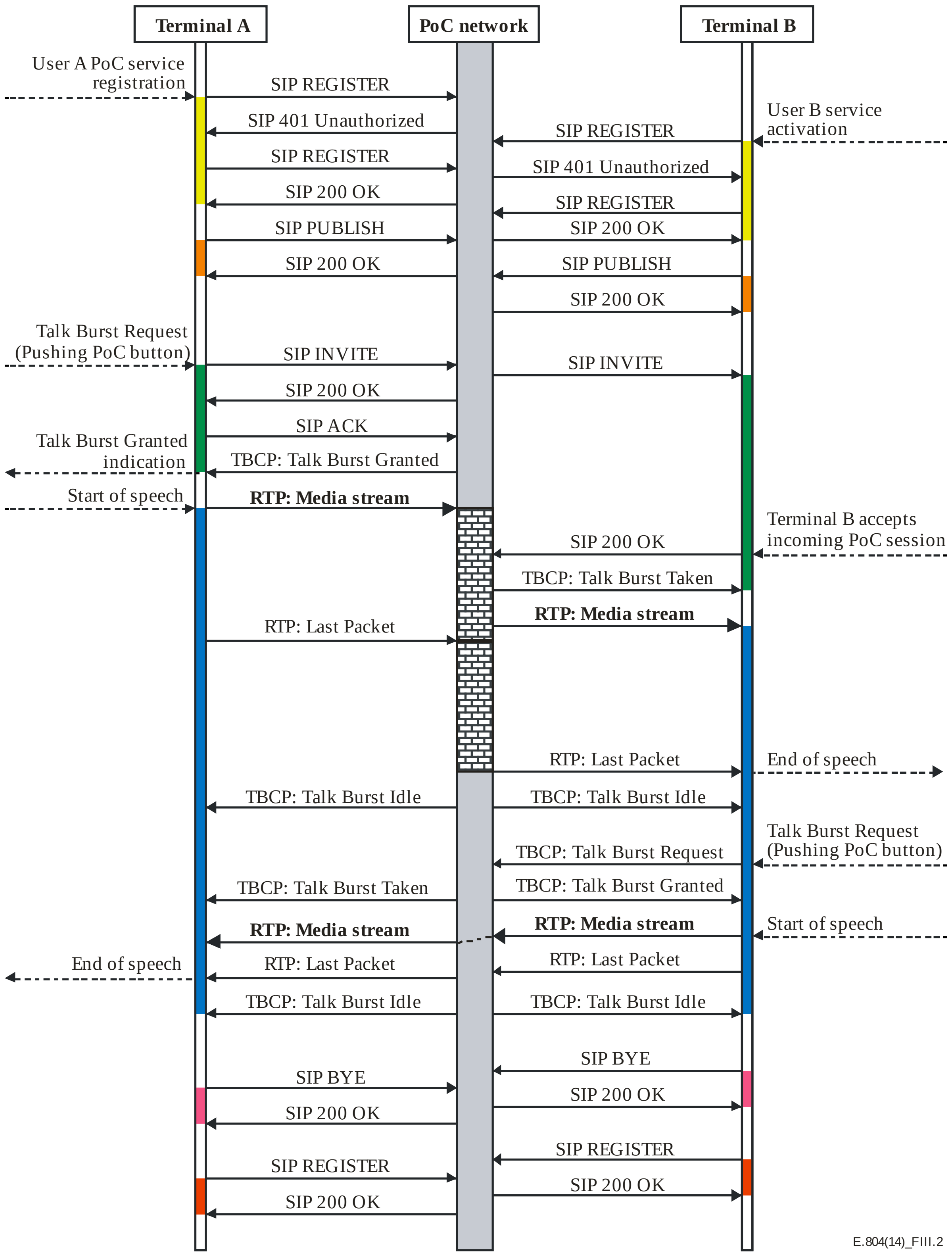


Figure III.1 – On-demand PoC session with manual-answer



NOTE – The PoC server supports media buffering and send the Talk Burst confirm message after receiving automatic-answer message.

Figure III.2 – Unconfirmed on-demand ad hoc PoC group session with automatic-answer

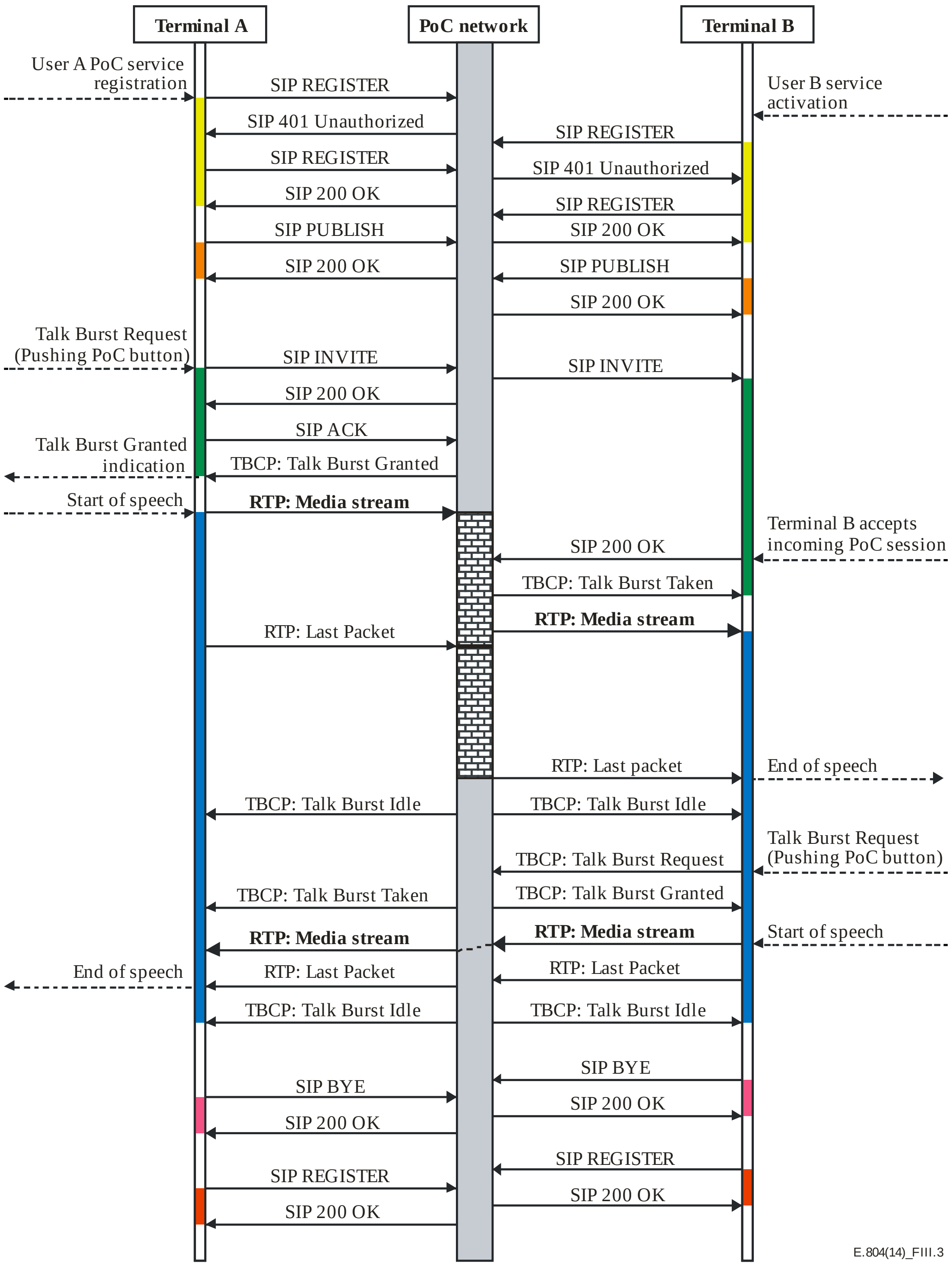


Figure III.3 – Confirmed pre-established session with manual-answer

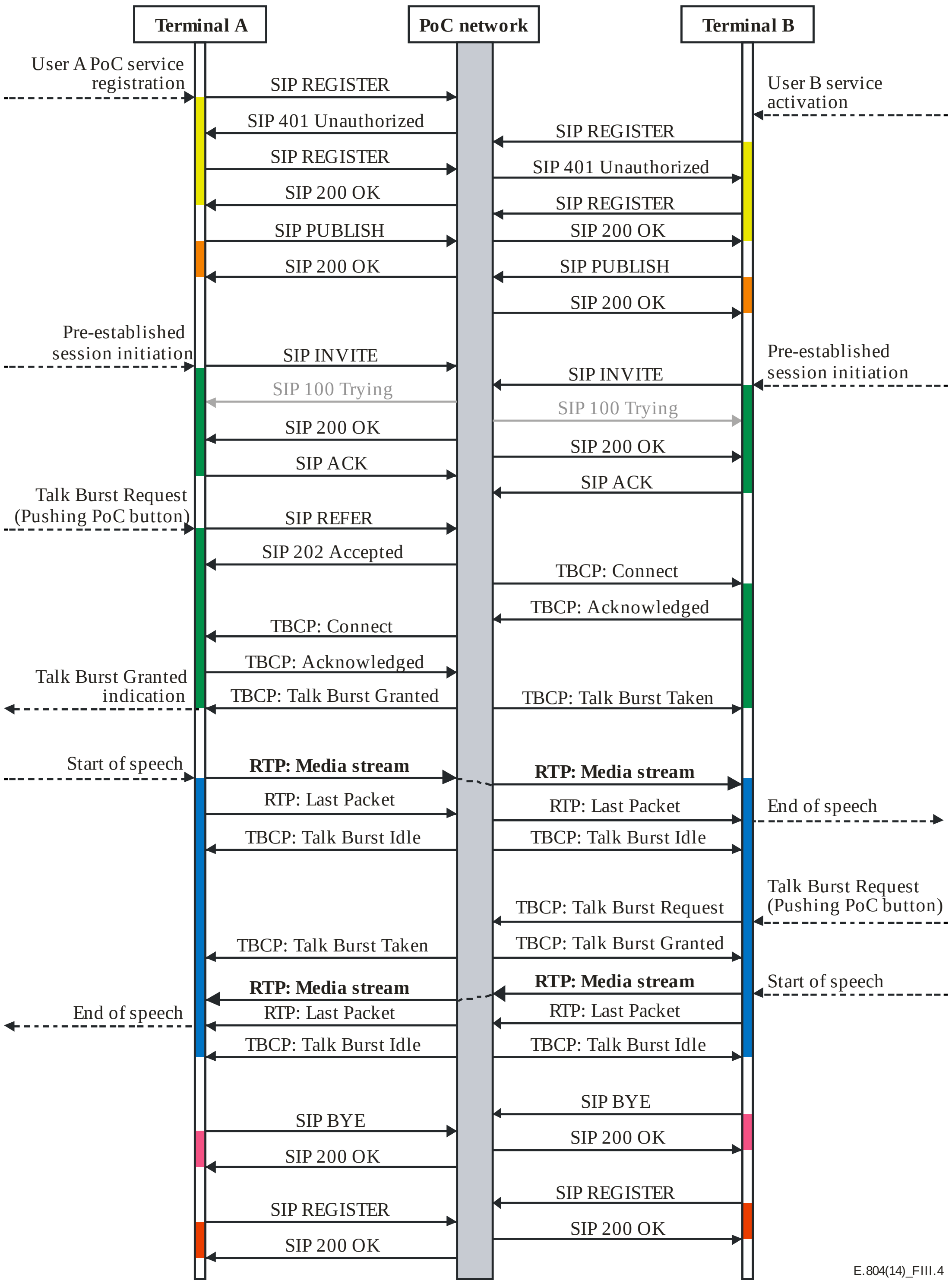


Figure III.4 – Unconfirmed pre-established session with automatic-answer

III.1 Signal grouping

This clause defines groups of signals which will in the following be referred to as building blocks of PoC signal flows, or just building blocks. These building blocks are derived from [b-ETSI TS 123 246], [OMA-1] and [OMA-2] representing only parts of a complete signal flow as seen in Figures III.1 to III.4. Here, different building blocks of the same kind correspond to the same QoS group. The aim of the definition of such building blocks is to give detailed information on the different signal flows.

Remark:

- In the QoS parameter defining clause of this Recommendation, most signal flows shown are less detailed. The reason for this is that these flows are only used to visualize the relevant trigger points of the corresponding QoS parameter with respect to their occurrence over time.

The relationship between building blocks and QoS groups is depicted in the following table. In contrast to the signal flows given to illustrate QoS parameter definition, only flows leading to a positive result are given. The only exception from this is the signal flow for a queued talk burst request which is added for sufficiency.

A distinction has been made between on-demand and pre-established PoC sessions since here different building blocks are needed. Crosses are indicating the blocks needed for the corresponding QoS group. For simplicity some crosses are greyed (i.e., shown with lighter colour). These crosses indicate that a choice between confirmed and unconfirmed indication has to be made.

Further parameters for the "Session SETUP" are the following:

- Session SETUP alternative 1: confirmed with auto-answered on terminating side.
- Session SETUP alternative 2: confirmed with manual answered on terminating side.
- Session SETUP alternative 3: unconfirmed with auto-answered on terminating side.
- Session SETUP alternative 4: unconfirmed with manual answered on terminating side.

Remarks:

- Only the QoS groups relevant to the building blocks are shown in Table III.1.
- Building blocks not related to any QoS group are omitted in Table III.1.
- Building blocks can be identified by their number as specified in Table III.1.

Table III.1 – Assignment of PoC session parts to building blocks

Building blocks (below) and QoS groups (right hand side)		REG	PUB	REG long	On-demand							Pre-established								DeREG
					INIT	Session SETUP				PtS	LEAVE	NEGO	INIT	Session SETUP				PtS	LEAVE	
						1	2	3	4					1	2	3	4			
1	PoC service registration	X	X	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x
2	PoC publish		x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	
3a	PoC on-demand session Initiation, confirmed				x	x	x			x										
3b	PoC on-demand session initiation, unconfirmed				x			x	x	x										
3c	PoC pre-established session media parameters negotiation											x	x	x	x	x	x	x		
3d	PoC pre-established session initiation, confirmed												x	x	x			x		
3e	PoC pre-established session initiation, unconfirmed												x			x	x	x		
4a	PoC on-demand session initiation, user B auto-answer					x		x		x										
4b	PoC on-demand session initiation, user B manual-answer						x		x	x										
4c	PoC Pre-established session initiation, user B auto-answer													x		x		x		
4d	PoC pre-established session initiation, user B														x		x	x		
5a	Media stream from user A to PoC server									x								x		
5b	Media stream from PoC server to user B, without buffer									x								x		
5c	Media stream from user B to user A, without buffer									x								x		
6b	Talk burst request									x								x		
6c	Queued talk burst request									x								x		
7a	Leaving PoC session (on-demand)										x									
7b	Leaving PoC session (pre-established)																		x	
8	Deregistration																			x

III.2 PoC service registration

Figure III.5 shows the PoC service registration signal flow.

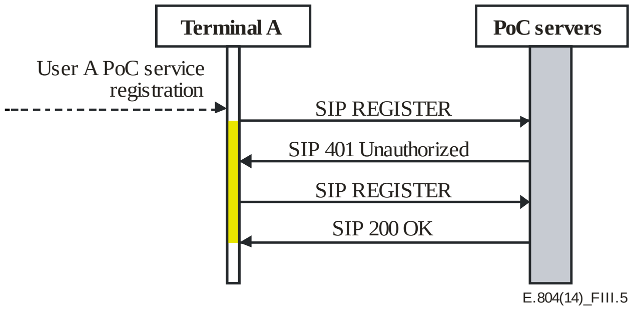


Figure III.5 – PoC service registration signal flow

III.3 PoC publish

Figure III.6 shows the publish signal flow.

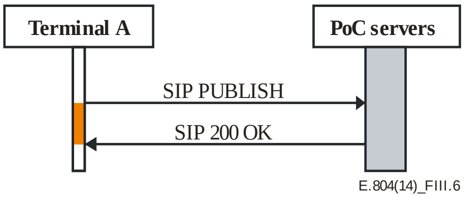


Figure III.6 – PoC publish signal flow

III.4 PoC session initiation, originating part

Figure III.7 shows the PoC on-demand session initiation signal flow (confirmed, originating part).

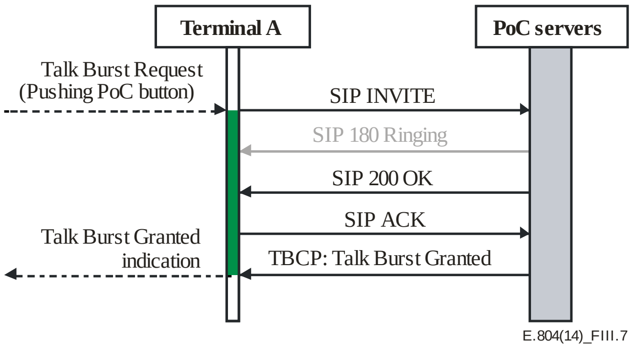


Figure III.7 – PoC on-demand session initiation signal flow (confirmed, originating part)

Figure III.8 shows the PoC on-demand session initiation (unconfirmed, originating part).

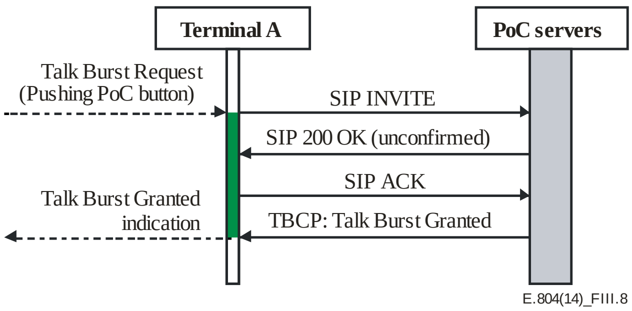


Figure III.8 – PoC on-demand session initiation signal flow (unconfirmed, originating part)

Figure III.9 shows the PoC pre-established session media parameters negotiation (originating part).

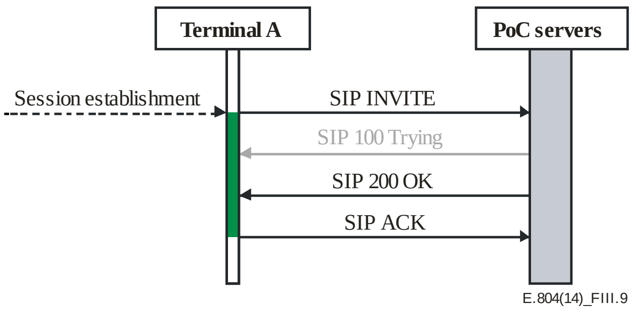


Figure III.9 – PoC pre-established session media parameters negotiation signal flow (originating part)

Figure III.10 shows the PoC pre-established session initiation signal flow (confirmed, originating part)

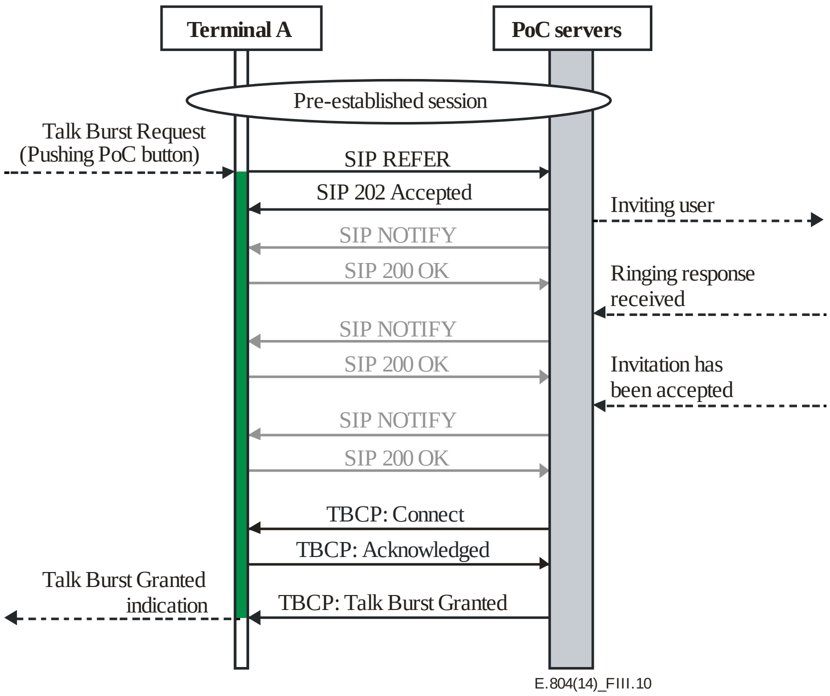


Figure III.10 – PoC pre-established session initiation signal flow (confirmed, originating part)

Figure III.11 shows PoC pre-established session initiation (unconfirmed, originating part).

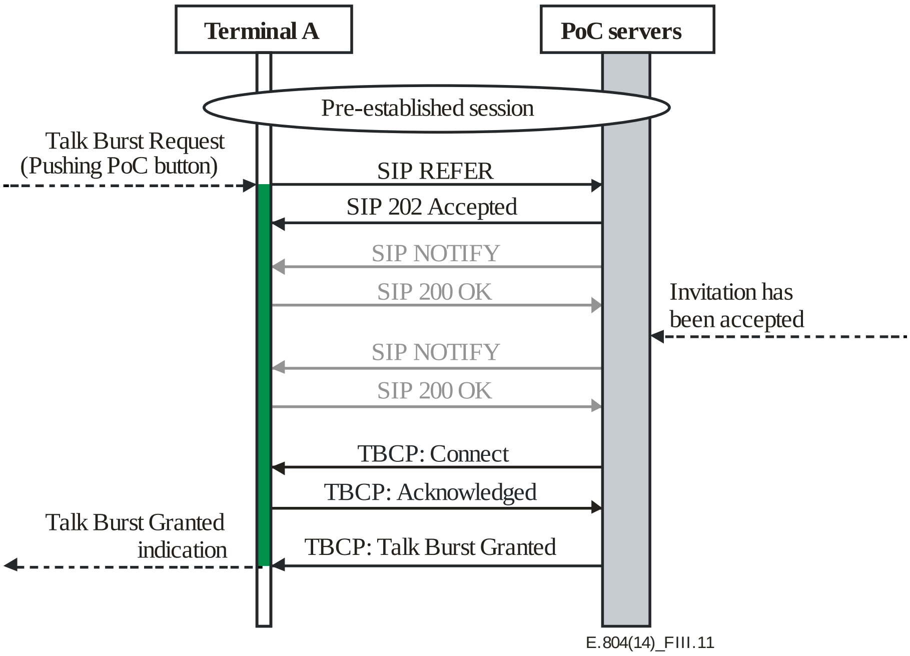


Figure III.11 – PoC pre-established session initiation signal flow (unconfirmed, originating part)

III.5 PoC session initiation, terminating part

Figure III.12 shows the PoC on-demand session signal flow (automatic answer terminating part).

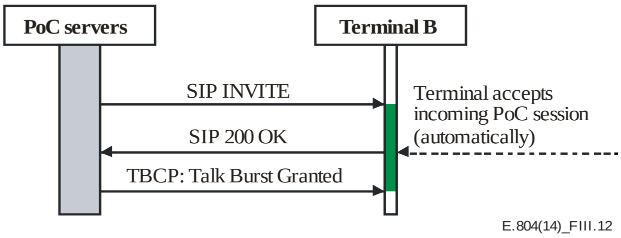


Figure III.12 – PoC on-demand session signal flow (automatic answer, terminating part)

Figure III.13 shows the PoC On-demand session signal flow (manual answer, terminating part).

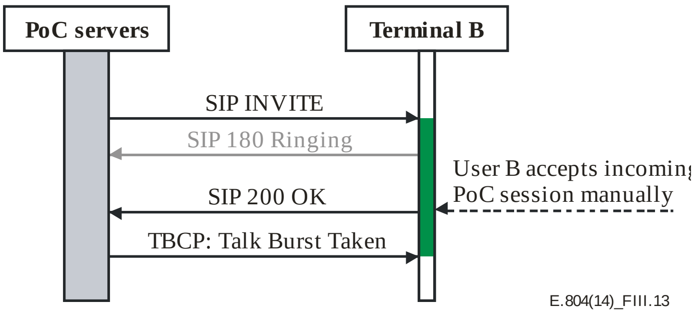


Figure III.13 – PoC on-demand session signal flow (manual answer, terminating part)

Figure III.14 shows PoC a pre-established session signal flow (automatic answer, terminating part).

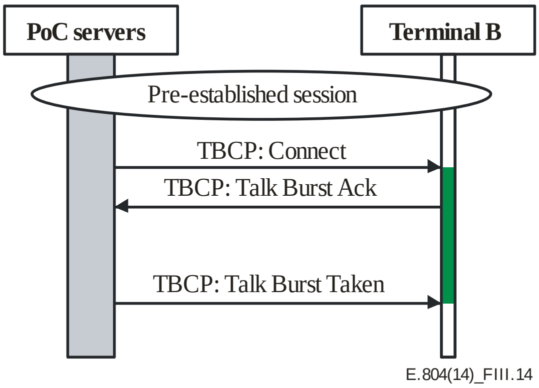


Figure III.14 – PoC pre-established session signal flow (automatic answer, terminating part)

Figure III.15 shows the PoC pre-established session signal flow (manual answer, terminating part).

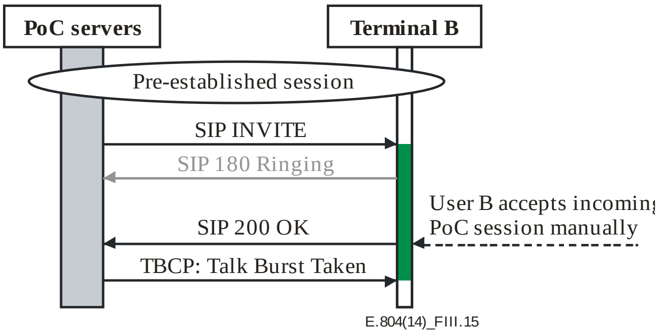


Figure III.15 – PoC pre-established session signal flow (manual answer, terminating part)

III.6 Media Streaming

Figure III.16 shows the signal flow for the first media stream from user A to PoC server.

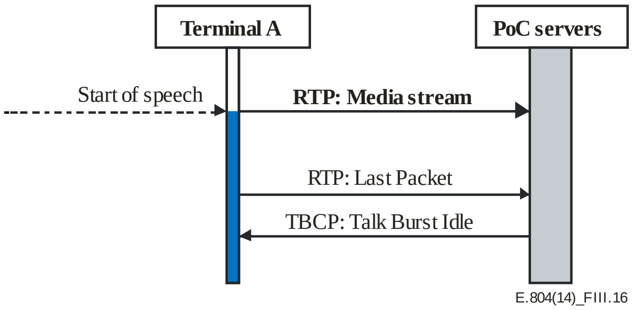


Figure III.16 – Signal flow for first media stream from user A to PoC server

Figure III.17 shows the signal flow for the first media stream from PoC server to user B (without media buffering).

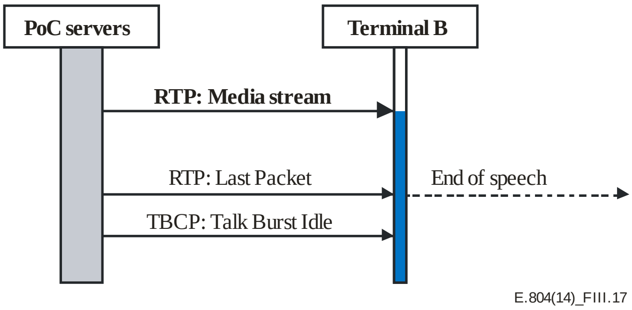


Figure III.17 – Signal flow for first media stream from PoC server to user B (no media buffering)

Figure III.18 shows the last media stream from user B to user A via PoC network (without media buffering), including Talk Burst Request of user B.

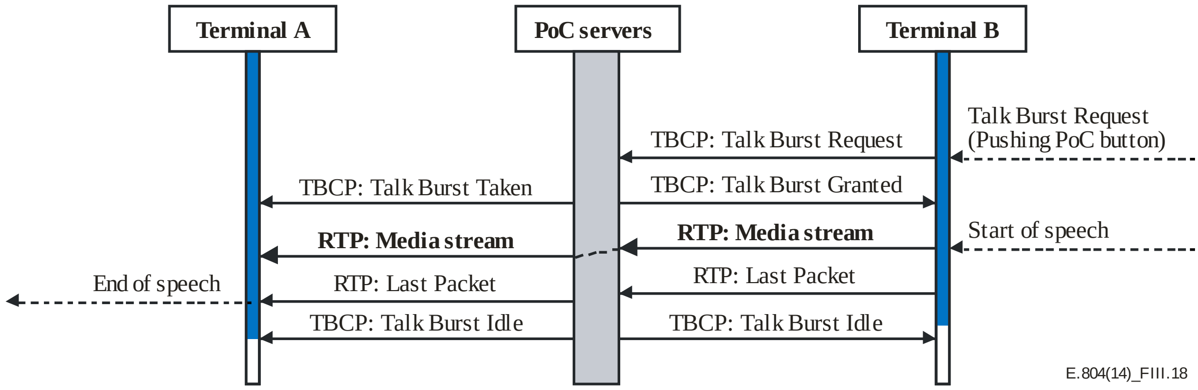


Figure III.18 – Signal flow for last media stream from user B to user A via PoC network (no media buffering)

III.7 Talk burst request

Figure III.19 shows the signal flow for implicit talk burst request (on-demand session initiation).

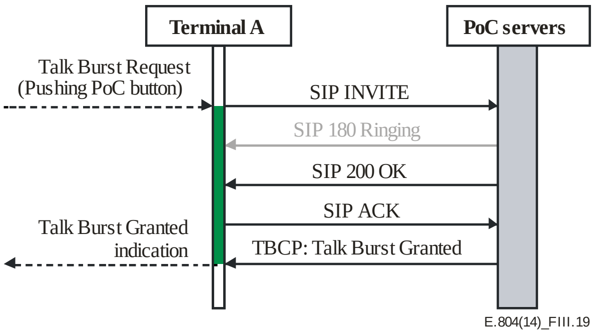


Figure III.19 – Signal flow for implicit talk burst request (on-demand session initiation)

Figure III.20 shows the signal flow for explicit talk burst request.

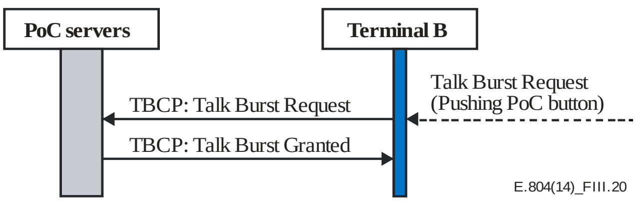


Figure III.20 – Signal flow for explicit talk burst request

Figure III.21 shows the signal flow for queued talk burst request.

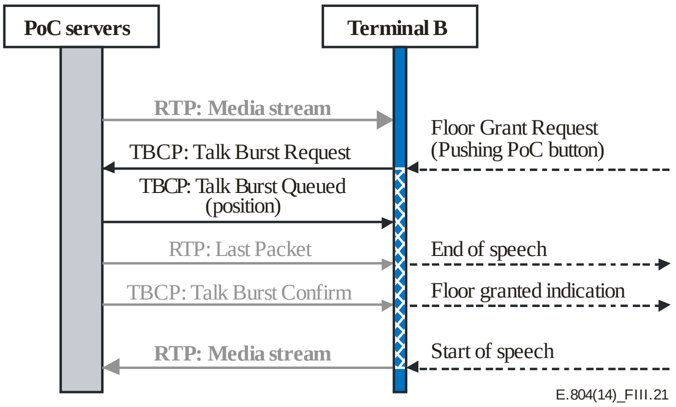


Figure III.21 – Signal flow for queued talk burst request

III.8 Leaving PoC session

Figure III.22 shows the signal flow for leaving on-demand PoC session.

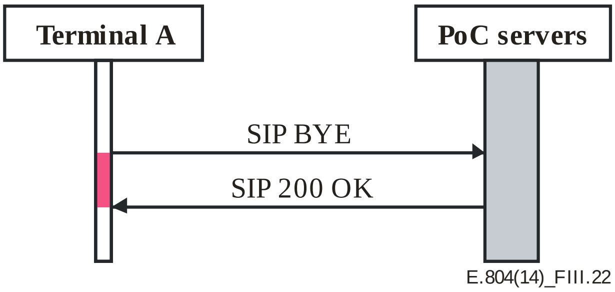


Figure III.22 – Signal flow for leaving on-demand PoC session

Figure III.23 shows the signal flow for leaving pre-established PoC session.

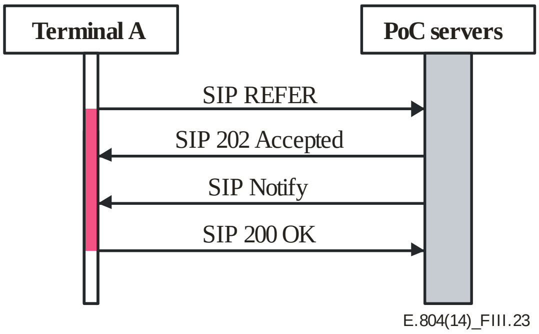


Figure III.23 – Signal flow for leaving pre-established PoC session

III.9 Deregistration

Figure III.24 shows the PoC service deregistration signal flow.

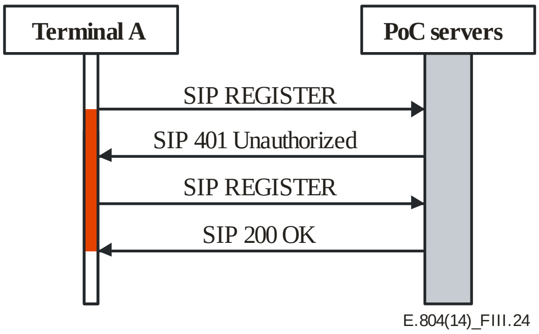


Figure III.24 – PoC service deregistration signal flow

Appendix IV

QoS parameter export

(This appendix does not form an integral part of this Recommendation.)

IV.1 Overview

Goal of this file format is to have a common interface to be able to interchange measurement results between systems of different vendors. Data should be provided on the highest possible level of granularity to support the QoS parameters defined in this Recommendation by keeping the possibility to select by time, geographical position and operator information.

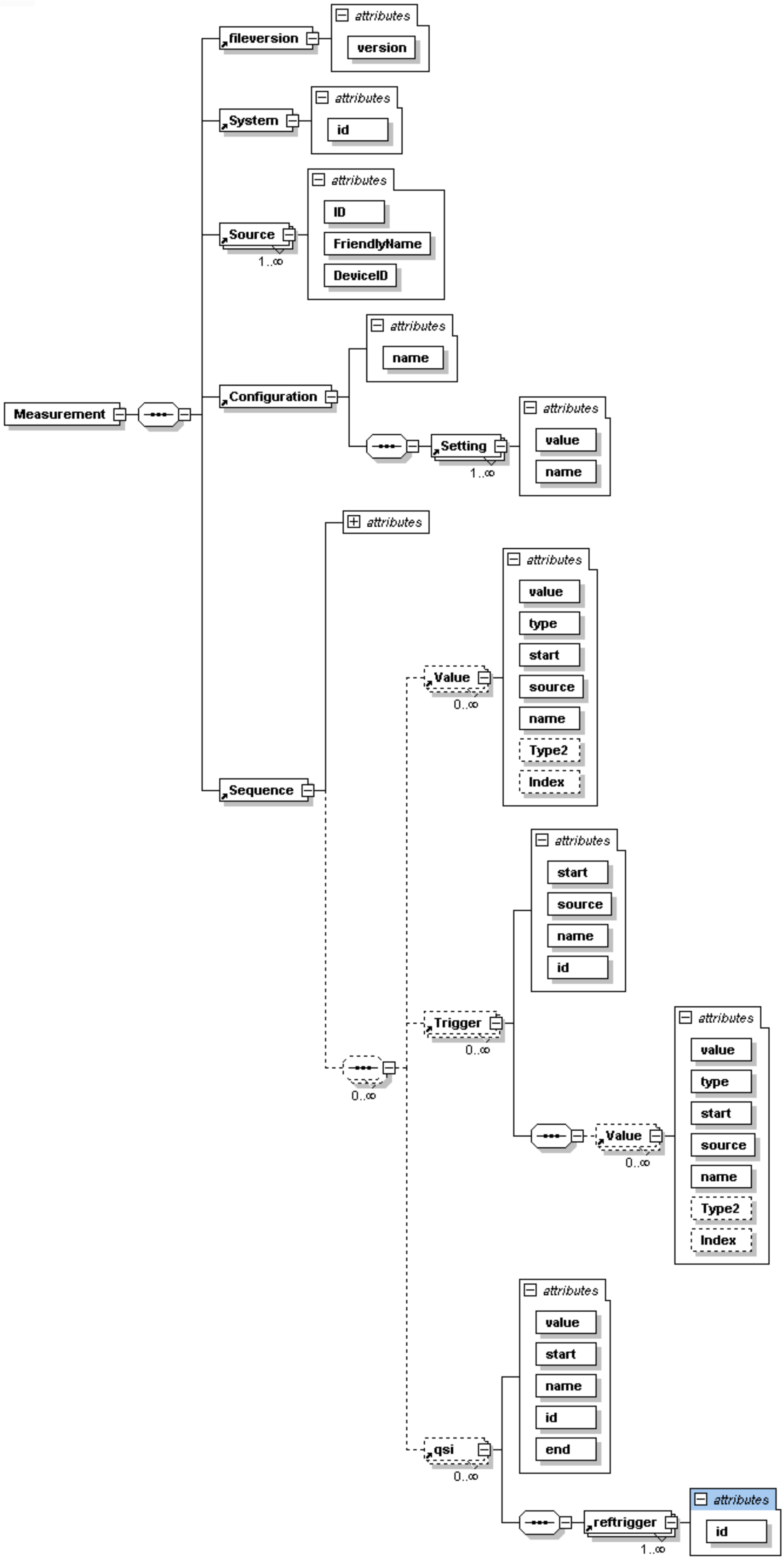
The file format is XML. This Recommendation describes the use of XML bodies and tags.

All relevant IDs for trigger points and QoS parameters can be found in clause 6.

IV.2 XML bodies

IV.2.1 <measurement>

This is the root of the XML document.


Measurement
fileversion
attributes
version
System
attributes
id
Source
1..∞
attributes
ID
FriendlyName
DeviceID
Configuration
attributes
name
Setting
1..∞
attributes
value
name
Sequence
attributes
0..∞
Value
0..∞
attributes
value
type
start
source
name
Type2
Index
Trigger
0..∞
attributes
start
source
name
id
Value
0..∞
attributes
value
type
start
source
name
Type2
Index
qsi
0..∞
attributes
value
start
name
id
end
reftrigger
1..∞
attributes
id

IV.2.2 Configuration

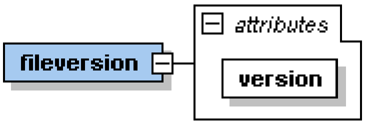
IV.2.2.1 <fileversion>

Version of the file definition.

Version	Version number of the document [b-ETSI TS 102 250-4]. E.g., 2.2.1.

Example: < fileversion version = "2.2.1"/>

XML schema:



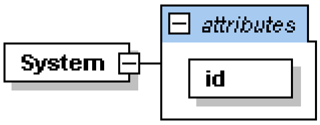
IV.2.2.2 <system>

Name of the measurement system

Id	A unique id of the measurement system.

Example: <system id = "A Measurement System"/>

XML schema:



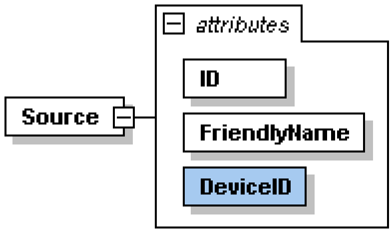
IV.2.2.3 <source>

A source can be any source of measurement data.

Id	A unique id of the data source within the XML file.
FriendlyName	The friendly name of the device.
DeviceID	Unique device ID, e.g., IMEI/IMSI.

Example: <source id = "1" FriendlyName ="Mobile A" DeviceID="1234567890" >

XML schema:



IV.2.2.4 <setting>

A setting is one special parameter of the measurement set-up of a logical part of the system.

Name	Name of the setting.
Value	Value of setting.

Example: <Setting name="POST_ATTACH_PAUSE" value="15 000"/>

XML schema:

See clause IV.2.2.5.

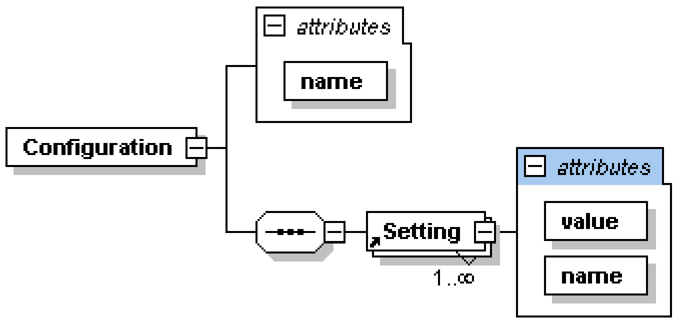
IV.2.2.5 <configuration>

The configuration container holds all settings concerning a logical part of the system.

Name	Name of the settings container.

Example: < configuration name="Service"/>

XML schema:



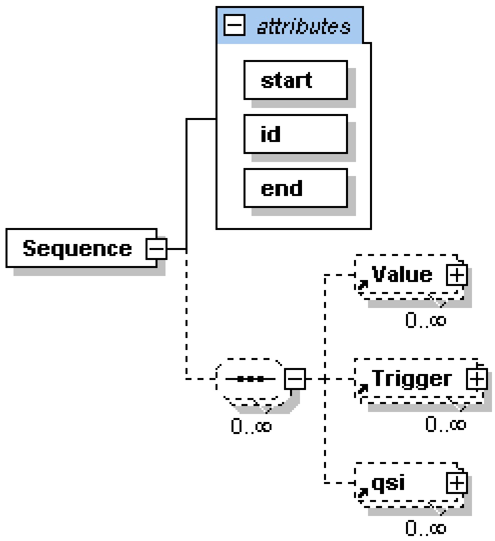
IV.2.3 Measurement results

IV.2.3.1 <sequence>A sequence contains all relevant data which is necessary for one measurement cycle. The tag is mandatory.

Start	Start time of the sequence.
End	End time of the sequence.
Id	Id of the sequence.

Example: <Sequence start="2007-01-25T15:18:25.405+01:00" end="2007-01-25T15:23:35.982+01:00" id="0005883"/>

XML schema:



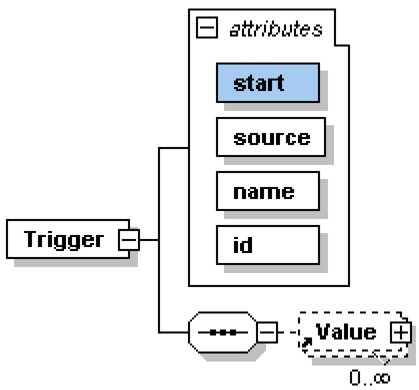
IV.2.3.2 <trigger>

A trigger point as defined in [b-ETSI TS 102 250-1]. The tag is mandatory.

Start	Time when trigger occurred.
Name	Trigger name, as specified in [b-ETSI TS 102 250-1].
Id	Id of trigger.
Source	ID of source.

Example: <Trigger name="attach attempt" start="2007-01-25T14:19:16.222" id="ID-0-51" source="1"/>

XML schema:



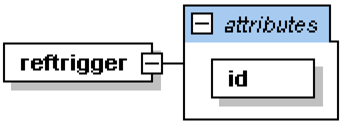
IV.2.3.3 <reftrigger>

A reference to an already reported trigger point. This allows to report the full information of the trigger point once and to refer to this report later. The tag is optional.

Id	Id of trigger.

Example: <REFTRIGGER ID=" ID-1-53" />

XML schema:



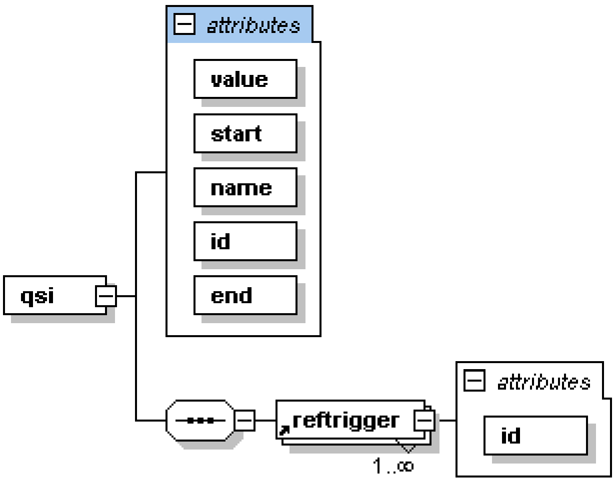
IV.2.3.4 <qsi>

A quality sequence indicator is a quality of service parameter of the test sequence, as defined in [b-ETSI TS 102 250-1]. The tag is optional.

Start	Start time of qsi calculation.
End	End time of qsi calculation.
Name	Name, as specified in [b-ETSI TS 102 250-1].
Id	Id of qsi.
Value	Value of QSI.

Example: <QSI name="CCR" start="2007-01-25T14:19:16.222" end="2007-01-25T14:19:19.358" value="0" id="ID-1-53"/>

XML schema:



IV.2.3.5 <value>

A measurement value can be any value, which was measured during the measurement cycle. The tag is optional.

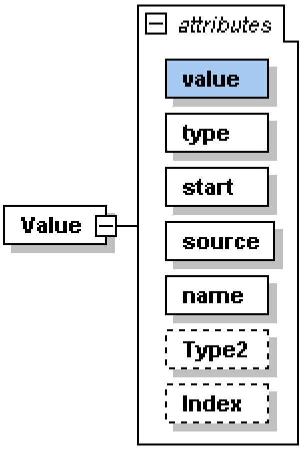
Start	Timestamp.
Name	Value name.
Value	Value of value.
Type	Type of value attribute.
Type2	Optional attribute, which may be necessary for interpreting the value.
Source	ID of source.
Index	Index of an array.

Type	Description
NUMBER	Number without unit for general purpose
BLOB	Binary dump, the values is a HEX dump, e.g.,: A1B204
STRING	Text for general purpose
BOOL	Boolean

Example 1: <Value name="GSM_MOBILE~SERVERREPORT~MCC" start="2007-01-25T14:18:26.647" value="262" type="NUMBER" source="1" />

Example 2: <Value name="GPS_SYSTEM~LLAPOSITION~LONGITUDE" start="2007-01-25T14:18:26.647" value="11.6" type="NUMBER" source="1"/>

XML schema:



IV.2.4 Data dictionary

The following data dictionary defines measurement values which can be added to the export file. If one of these values becomes a trigger point, it should not be removed from the list. The trigger point should be defined as additional information.

Other, not defined, values may be exported as well. Their names have to be marked clearly by the prefix USERData~. E.g., USERData~newvalue1.

NOTE – If the vendor, e.g., wants to add a version, it can create an item:

- USERData~Version.

IV.2.4.1 GPS

Name	Unit	Remarks
GPS_SYSTEM~LLAPOSITION~LATITUDE	°	Latitude (WGS84)
GPS_SYSTEM~LLAPOSITION~LONGITUDE	°	Longitude (WGS84)
GPS_SYSTEM~LLAPOSITION~ALTITUDE	m	Altitude
GPS_SYSTEM~LLAPOSITION~GPS_DISTANCE	m	Driven distance
GPS_SYSTEM~SPEEDINFO~SPEED	km/h	Speed
GPS_SYSTEM~SPEEDINFO~HEADING	°	Heading

IV.2.4.2 GSM

Name	Unit	Remarks
GSM_MOBILE~SERVERREPORT~BCCH	ChanNr	BCCH
GSM_MOBILE~SERVERREPORT~BSIC	–	BSIC
GSM_MOBILE~SERVERREPORT~RxLevF	step	Rx Lev full
GSM_MOBILE~SERVERREPORT~RxLevS	step	Rx Lev sub
GSM_MOBILE~SERVERREPORT~RxQualF	–	Rx Qual full
GSM_MOBILE~SERVERREPORT~RxQualS	–	Rx Qual sub
GSM_MOBILE~SERVERREPORT~C1	–	C1 criteria
GSM_MOBILE~SERVERREPORT~C2	–	C2 criteria
GSM_MOBILE~SERVERREPORT~SVR_AVG_C_I	dB	Average C/I
GSM_MOBILE~SERVERREPORT~CELLBARFLAG	–	Cellbar flag
GSM_MOBILE~SERVERREPORT~MCC	–	MCC
GSM_MOBILE~SERVERREPORT~MNC	–	MNC
GSM_MOBILE~SERVERREPORT~LAC	–	LAC
GSM_MOBILE~SERVERREPORT~CI	–	Cell identity
GSM_MOBILE~SERVERREPORT~TIMESLOT	–	Used timeslot
GSM_MOBILE~SERVERREPORT~TIMINGADVANCE	step	Timing advance
GSM_MOBILE~SERVERREPORT~TSC	–	Trainings sequence
GSM_MOBILE~SERVERREPORT~TXPOWER	–	Tx power
GSM_MOBILE~SERVERREPORT~DTX	–	DTX on/off
GSM_MOBILE~SERVERREPORT~HOPPING	–	Hopping on/off
GSM_MOBILE~SERVERREPORT~TRAFFICCHANNEL	–	Used traffic channel, if hopping is off
GSM_MOBILE~SERVERREPORT~MAIO	–	MAIO
GSM_MOBILE~SERVERREPORT~HSN	–	HSN
GSM_MOBILE~SERVERREPORT~SERVINGCELL	–	Name of the serving cell

Name	Unit	Remarks
GSM_MOBILE~SERVERREPORT~FER_Full	%	FER full
GSM_MOBILE~SERVERREPORT~FER_Sub	%	FER sub
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_RA_COLOUR	–	RA colour
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_RXLEVEL	step	GPRS Rx Lev
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_C31	–	C31 criteria
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_C32	–	C32 criteria
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_BCCH	–	BCCH of neighbour, array for n neighbours
GSM_MOBLE~PACKET_SERVERREPORT~P_SVR_N_NCC	–	BCC of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_BCC	–	NCC of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_RA_COLOUR	–	RA Colour of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_RX_LEVEL	step	GPRS RxLev of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_C31	–	C31 Criteria of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_C32	–	C32 Criteria of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_GPRS_RX_LEV_ACCESS_MIN	step	GPRS_RX_LEV_ACCESS_MIN of neighbour, Array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_GPRS_MS_TXPWR_MAX_CCH	–	GPRS_MS_TXPWR_MAX_CCH of neighbour, Array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_HCS_THR	step	HCS_THR of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_PRIORITY_CLASS	–	PRIORITY_CLASS of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_PSI1_REPEAT_PERIOD	–	PSI1_REPEAT_PERIOD of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_PBCCH_LOCATION	–	PBCCH_LOCATION of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_GPRS_INDICATOR	–	GPRS_INDICATOR of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_PBCCH_Indicator	–	PBCCH_Indicator of neighbour, array for n neighbours

Name	Unit	Remarks
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_GPRS_PENALTY_TIME	–	GPRS_PENALTY_TIME of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_GPRS_RESELECT_OFFSET	–	GPRS_RESELECT_OFFSET of neighbour, array for n neighbours
GSM_MOBILE~PACKET_SERVERREPORT~P_SVR_N_GPRS_TEMPORARY_OFFSET	–	GPRS_TEMPORARY_OFFSET of neighbour, array for n neighbours
GSM_MOBILE~MEASUREMENTREPORT~MMR_N_BCCH	–	BCCH of neighbour, array for n neighbours
GSM_MOBILE~MEASUREMENTREPORT~MMR_N_RXLEV	step	Rx Lev of neighbour, array for n neighbours
GSM_MOBILE~MEASUREMENTREPORT~MMR_N_BSIC	–	BSIC of neighbour, array for n neighbours
GSM_MOBILE~MEASUREMENTREPORT~MMR_N_C1	–	C1 Criteria of neighbour, array for n neighbours
GSM_MOBILE~MEASUREMENTREPORT~MMR_N_C2	–	C2 Criteria of neighbour, array for n neighbours
GSM_MOBILE~MEASUREMENTREPORT~NEIGHBOURCELL	–	Name of neighbour, array for n neighbours
GSM_MOBILE~LAYER3~L3MESSAGE	–	Byte stream of message

IV.2.4.2 UMTS

Name	Unit	Remarks
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_RSCP_CPICH	dBm	RSCP
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_RSSI_UTRA	dBm	RSSI
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_EcNo_CPICH	dB	Ec/No of CPICH
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_Tx_POWER	dBm	Tx power
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_TCH_BLER	%	TrCH BLER

Name	Unit	Remarks
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_UL_INTERFERENCE	dBm	UL interference
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_DF_DL	SF	Spreading factor downlink
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_DF_UL	SF	Spreading factor uplink
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_UL_ARFCN		UARFCN UL
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_DL_ARFCN		UARFCN DL
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_SC	–	Scrambling code from 0…8191.
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_HSDPA	–	HSDPA is active
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_HSUPA	–	HSUPA is active
UMTS_MOBILE~UMTS_MOBILE_LAYER_1~ UMTS_MOBILE_LAYER_1_NODEB	–	Name of current best Node B
UMTS_MOBILE~UMTS_MOBILE_LAYER_3~ UMTS_MOBILE_L3_RRC_STATE	–	"Unknown", "CELL FACH", "CELL DCH", "CELL PCH", "URA PCH"
UMTS_MOBILE~UMTS_MOBILE_LAYER_3~ UMTS_MOBILE_L3_SC_NAME	–	Name of the cell
UMTS_MOBILE~UMTS_MOBILE_LAYER_3~ UMTS_MOBILE_L3_CELL_ID	–	16 bit cell ID
UMTS_MOBILE~UMTS_MOBILE_LAYER_3~ UMTS_MOBILE_L3_CELL_ID_28BIT	–	28 bit cell ID
UMTS_MOBILE~UMTS_MOBILE_LAYER_3~ UMTS_MOBILE_L3_LAC	–	LAC

Name	Unit	Remarks
UMTS_MOBILE~UMTS_MOBILE_LAYER_3~ UMTS_MOBILE_L3_MNC	–	MNC
UMTS_MOBILE~UMTS_MOBILE_LAYER_3~ UMTS_MOBILE_L3_MCC	–	MCC
UMTS_MOBILE~UMTS_MOBILE_LAYER_3~ UMTS_MOBILE_L3_URA_ID	–	URA ID
UMTS_MOBILE~UMTS_MOBILE_AS~ UMTS_MOBILE_AS_SC_NAME	–	Name of the cell
UMTS_MOBILE~UMTS_MOBILE_AS~ UMTS_MOBILE_AS_COUNT	–	when in CELL_DCH
UMTS_MOBILE~UMTS_MOBILE_AS~ UMTS_MOBILE_AS_DL_UARFCN		UARFCN
UMTS_MOBILE~UMTS_MOBILE_AS~AS_SC	–	Scrambling code, array for n members of the active set
UMTS_MOBILE~UMTS_MOBILE_AS~AS_EcIo	dB	Ec/No, array for n members of the active set
UMTS_MOBILE~UMTS_MOBILE_AS~AS_RSCP	dBm	RSCP, array for n members of the active set
UMTS_MOBILE~UMTS_MOBILE_AS~AS_NODEB	–	Name of Node B, array for n members of the active set
UMTS_MOBILE~UMTS_MOBILE_NS~UMTS_MOBILE_NS_COUNT	–	Number of cells in neighbour set
UMTS_MOBILE~UMTS_MOBILE_NS~NS_SC_0	–	Scrambling code, array for n members of the neighbour set
UMTS_MOBILE~UMTS_MOBILE_NS~NS_EcIo_0	dB	Ec/No, array for n members of the neighbour set
UMTS_MOBILE~UMTS_MOBILE_NS~NS_RSCP_0	dBm	RSCP, array for n members of the neighbour set

Name	Unit	Remarks
UMTS_MOBILE~UMTS_MOBILE_NS~NS_STATE_0	–	Current state of neighbour. Array for n members of the neighbour set. Possible values: "Active", "Serving", "Monitored", "Detected", "Undetected", "Not listed or detected"
UMTS_MOBILE~UMTS_MOBILE_NS~NS_NODEB_0	–	Name of Node B, array for n members of the neighbour set
UMTS_MOBILE~UMTS_MOBILE_NS~NS_UARFCN_0	–	UARFCN of Node B, array for n members of the neighbour set
UMTS_MOBILE~UMTS_MOBILE_FINGER_INFO~ UMTS_MOBILE_FINGER_INFO_1_Eb_Io	dB	Ec/No of finger, array for n fingers
UMTS_MOBILE~UMTS_MOBILE_FINGER_INFO~ UMTS_MOBILE_FINGER_INFO_1_SC	–	Scrambling code of finger, array for n fingers
UMTS_MOBILE~UMTS_MOBILE_FINGER_INFO~ UMTS_MOBILE_FINGER_INFO_1_TO	–	Offset of finger, array for n fingers
UMTS_MOBILE~UMTS_MOBILE_RRC_MSG~ UMTS_MOBILE_RRC_MSG_CONTENT	–	See clause IV.2.4.3

IV.2.4.3 Layer 3 message PDU types

The attribute type 2 of the value tag has to be set to one of the following values:

Type of PDU	Type 2
umts_r6_DL_DCCH_Message_PDU	1
umts_r6_UL_DCCH_Message_PDU	2
umts_r6_DL_CCCH_Message_PDU	3
umts_r6_UL_CCCH_Message_PDU	4
umts_r6_PCCH_Message_PDU	5
umts_r6_DL_SHCCH_Message_PDU	6
umts_r6_UL_SHCCH_Message_PDU	7
umts_r6_BCCH_FACH_Message_PDU	8
umts_r6_BCCH_BCH_Message_PDU	9
umts_r6_MCCH_Message_PDU	10
umts_r6_MSCH_Message_PDU	11
umts_r6_HandoverToUTRANCommand_PDU	12
umts_r6_InterRATHandoverInfo_PDU	13
umts_r6_InterRATHandoverInfo_r3_add_ext_IEs_PDU	14
umts_r6_RRCConnectionSetupComplete_r3_add_ext_IEs_PDU	15
umts_r6_UECapabilityInformation_r3_add_ext_IEs_PDU	16
umts_r6_UE_CapabilityContainer_IEs_PDU	17
umts_r6_UE_RadioAccessCapabilityInfo_PDU	18
umts_r6_UL_PhysChCapabilityFDD_r6_PDU	19

Type of PDU	Type 2
umts_r6_TFC_Subset_ID_With3b_PDU	20
umts_r6_TFC_Subset_ID_With5b_PDU	21
umts_r6_TFC_Subset_ID_With10b_PDU	22
umts_r6_DL_CCTrChTPCList_PDU	23
umts_r6_Event1b_r6_PDU	24
umts_r6_MasterInformationBlock_PDU	25
umts_r6_SysInfoType1_PDU	26
umts_r6_SysInfoType2_PDU	27
umts_r6_SysInfoType3_PDU	28
umts_r6_SysInfoType4_PDU	29
umts_r6_SysInfoType5bis_PDU	30
umts_r6_SysInfoType6_PDU	31
umts_r6_SysInfoType7_PDU	32
umts_r6_SysInfoType8_PDU	33
umts_r6_SysInfoType9_PDU	34
umts_r6_SysInfoType10_PDU	35
umts_r6_SysInfoType11_PDU	36
umts_r6_SysInfoType12_PDU	37
umts_r6_SysInfoType13_PDU	38
umts_r6_SysInfoType13_1_PDU	39

Type of PDU	Type 2
umts_r6_SysInfoType13_2_PDU	40
umts_r6_SysInfoType13_3_PDU	41
umts_r6_SysInfoType13_4_PDU	42
umts_r6_SysInfoType14_PDU	43
umts_r6_SysInfoType15_PDU	44
umts_r6_SysInfoType15_1_PDU	45
umts_r6_SysInfoType15_2_PDU	46
umts_r6_SysInfoType15_3_PDU	47
umts_r6_SysInfoType15_4_PDU	48
umts_r6_SysInfoType15_5_PDU	49
umts_r6_SysInfoType16_PDU	50
umts_r6_SysInfoType17_PDU	51
umts_r6_SysInfoType18_PDU	52
umts_r6_SysInfoTypeSB1_PDU	53
umts_r6_SysInfoTypeSB2_PDU	54
umts_r6_MBMS_PreferredFreqRequest_r6_PDU	55
umts_r6_ToTargetRNC_Container_PDU	56
umts_r6_TargetRNC_ToSourceRNC_Container_PDU	57
umts_r6_SRNC_RelocationInfo_v3h0ext_IEs_PDU	58

IV.2.4.4 High speed downlink packet access (HSDPA)

Name	Unit	Remarks
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_DEMOD_VALIDS	%	Decode success rate
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_DSCH_PASSED	%	DSCH passed rate
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_DSCH_FAILED	%	DSCH failed rate
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_DSCH_DTX	%	DSCH DTX rate

Name	Unit	Remarks
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_RE_TRANSMISSION	%	Retransmission rate
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_QPSK	%	SCCH QPSK rate
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_16QAM	bit	SCCH 16QAM rate
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_MAX_TBS	bit	Max transport block size
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_MIN_TBS	bit	Min transport block size
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_AVG_TBS	–	Average transport block size
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_MAX_CHAN	–	SCCH Max # of codes
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_MIN_CHAN	–	SCCH Min # of codes
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_AVG_CHAN	–	SCCH average # of codes
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_SCCH_TP	kbit/s	SCCH throughput when served
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_SCCH_TP_SCHEDULED	kbit/s	SCCH throughput when scheduled
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_SCCH_TP_MAC	kbit/s	MAC throughput
UMTS_MOBILE~UMTS_HSDPA_DECODE_SUM~ UMTS_MOBILE_HSDPA_SCCH_BLER	%	BLER
UMTS_MOBILE~UMTS_HSDPA_ACK_NACK_STAT~ UMTS_MOBILE_HSDPA_ACK_NACK_SAMPLES		Count of ACK/NACK samples

Name	Unit	Remarks
UMTS_MOBILE~UMTS_HSDPA_ACK_NACK_STAT~ UMTS_MOBILE_HSDPA_ACK_PERCENTAGE	%	ACK rate
UMTS_MOBILE~UMTS_HSDPA_ACK_NACK_STAT~ UMTS_MOBILE_HSDPA_NACK_PERCENTAGE	%	NACK rate
UMTS_MOBILE~UMTS_HSDPA_ACK_NACK_STAT~ UMTS_MOBILE_HSDPA_DTX_PERCENTAGE	%	DTX rate
UMTS_MOBILE~UMTS_HSDPA_CQI_STAT_2~ UMTS_MOBILE_HSDPA_CQI2_SAMPLES	–	Count of CQI samples
UMTS_MOBILE~UMTS_HSDPA_CQI_STAT_2~ UMTS_MOBILE_HSDPA_CQI2_TIME_SPAN	ms	Time span of measurement
UMTS_MOBILE~UMTS_HSDPA_CQI_STAT_2~ UMTS_MOBILE_HSDPA_CQI2_MIN	–	Min. CQI
UMTS_MOBILE~UMTS_HSDPA_CQI_STAT_2~ UMTS_MOBILE_HSDPA_CQI2_MAX	–	Max. CQI
UMTS_MOBILE~UMTS_HSDPA_CQI_STAT_2~ UMTS_MOBILE_HSDPA_CQI2_MIN_CODE	–	Min. number of codes
UMTS_MOBILE~UMTS_HSDPA_CQI_STAT_2~ UMTS_MOBILE_HSDPA_CQI2_MAX_CODE	–	Max. number of codes
UMTS_MOBILE~UMTS_HSDPA_CQI_STAT_2~ UMTS_MOBILE_HSDPA_CQI2_TP	kbit/s	Requested throughput
UMTS_MOBILE~UMTS_HSDPA_CQI_STAT_2~ UMTS_MOBILE_HSDPA_CQI2_AVG	–	Average CQI
UMTS_MOBILE~UMTS_HSDPA_CQI_STAT_2~ UMTS_MOBILE_HSDPA_CQI2_AVG_CODE	–	Average number of codes
UMTS_MOBILE~UMTS_HSDPA_CQI_STAT_2~ UMTS_MOBILE_HSDPA_CQI2_QPSK	%	Requested QPSK rate
UMTS_MOBILE~UMTS_HSDPA_CQI_STAT_2~ UMTS_MOBILE_HSDPA_CQI2_16QAM	%	Requested 16QAM rate

IV.2.4.5 High speed uplink packet access (HSUPA)

Name	Unit	Remarks
UMTS_MOBILE~UMTS_HSUPA_EDPCH_SUMMERY~ UMTS_MOBILE_HSUPA_HAPPY_RATE	%	HappyRate
UMTS_MOBILE~RN_UMTS_MOBILE_HSUPA_EDCH_SUMMERY_EV~UMTS_MOBILE_HSUPA_CHANBIT_RATE	kbit/s	Channelbit rate
UMTS_MOBILE~RN_UMTS_MOBILE_HSUPA_EDCH_SUMMERY_EV~UMTS_MOBILE_HSUPA_TBS_RATE	kbit/s	TBS bit rate
UMTS_MOBILE~RN_UMTS_MOBILE_HSUPA_EDCH_SUMMERY_EV~UMTS_MOBILE_HSUPA_1ST_TX_TBS_RATE	kbit/s	1st Tx TBS bit rate
UMTS_MOBILE~UMTS_HSUPA_EDPCH_SUMMERY~ UMTS_MOBILE_HSUPA_TTI	–	TTI, 2 ms or 10 ms
UMTS_MOBILE~UMTS_HSUPA_EDPCH_SUMMERY~ UMTS_MOBILE_HSUPA_MIN_RE_TX	%	Min. retransmission
UMTS_MOBILE~UMTS_HSUPA_EDPCH_SUMMERY~ UMTS_MOBILE_HSUPA_MAX_RE_TX	%	Max. retransmission
UMTS_MOBILE~UMTS_HSUPA_EDPCH_SUMMERY~ UMTS_MOBILE_HSUPA_AVG_RE_TX	%	Avg. retransmission
UMTS_MOBILE~UMTS_HSUPA_EDPCH_SUMMERY~ UMTS_MOBILE_HSUPA_MIN_NDATA	–	Min. Channelbits
UMTS_MOBILE~UMTS_HSUPA_EDPCH_SUMMERY~ UMTS_MOBILE_HSUPA_MAX_NDATA	–	Max. Channelbits
UMTS_MOBILE~UMTS_HSUPA_EDPCH_SUMMERY~ UMTS_MOBILE_HSUPA_AVG_NDATA	–	Avg. Channelbits
UMTS_MOBILE~RN_UMTS_MOBILE_HSUPA_EDCH_SUMMERY_EV~UMTS_MOBILE_HSUPA_MIN_TBS	–	Min. TBS
UMTS_MOBILE~RN_UMTS_MOBILE_HSUPA_EDCH_SUMMERY_EV~UMTS_MOBILE_HSUPA_MAX_TBS	–	Max. TBS
UMTS_MOBILE~RN_UMTS_MOBILE_HSUPA_EDCH_SUMMERY_EV~UMTS_MOBILE_HSUPA_AVG_TBS	–	Avg. TBS

Name	Unit	Remarks
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_NUMBER	–	# of cells in the cell set
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_COMB_ACK_TP	kbit/s	Comb. Ack'd throughput
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_SCH_COMB_ACK_TP	kbit/s	Scheduled throughput
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_COMB_ACK	%	Comb. ACK rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_COMB_NACK	%	Comb. NACK rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_NS_ACK_RATE	%	NS ACK rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_COMB_HICH_SUCCESS	%	Comb. HICH success rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_COMB_UP	%	Comb. UP rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_COMB_DOWN	%	Comb. DOWN rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_COMB_HOLD	%	Comb. HOLD rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_MIN_SG_INDEX	–	Min. SG index
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_MAX_SG_INDEX	–	Max. SG index
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_AVG_SG_INDEX	–	Avg. SG index
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_MIN_LUPR_INDEX	–	Min. LUPR index

Name	Unit	Remarks
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_MAX_LUPR_INDEX	–	Max. LUPR index
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_AVG_LUPR_INDEX	–	Avg. LUPR index
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_DTX	%	DTX rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_HAPPY	%	Happy rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_MIN_RE_TX_COUNT	–	Min. retransmission count
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_MAX_RE_TX_COUNT	–	Max. retransmission count
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_AVG_RE_TX_COUNT	–	Avg. retransmission count
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_LTD_PWR	–	Ltd. by power rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_LTD_SG	–	Ltd. by SG rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_LTD_BUFFER	–	Ltd. by buffer Occ. rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_LTD_MUX	%	Non-serving HOLD rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~ UMTS_MOBILE_HSUPA_CS_LTD_HARQ	%	Non-serving DOWN rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~HSUPA_RLS_SC_0	–	Serving cell SC
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_ID_0	–	Serving cell RLS IDX
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_RSCP_0	dBm	Serving cell RSCP
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_EcIo_0	dB	Serving cell Ec/Io

Name	Unit	Remarks
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_ACKD_TP_0	kbit/s	Serving cell ACK'd throughput
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_ACK_0	–	Serving cell ACK rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_NACK_0	–	Serving cell NACK rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_HICH_SUC_0	–	Serving cell HICH success rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_UP_0	–	Serving cell UP rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_DOWN_0	–	Serving cell DOWN rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_HOLD_0	–	Serving cell HOLD rate
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_MIN_AG_0	–	Serving cell Min. AG value
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_MAX_AG_0	–	Serving cell Max. AG value
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_AVG_AG_0	–	Serving cell Avg. AG value
UMTS_MOBILE~UMTS_HSUPA_CELLS~HSUPA_RLS_SC_1	–	Non-serving CellSC , Array of 0..3 cells
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_ID_1	–	Non-serving CellRLSIDX, array of 0..3 cells
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_RSCP_1	dBm	Non-serving CellRSCP, array of 0..3 cells
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_EcIo_1	dB	Non-serving CellEcIo, array of 0..3 cells
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_ACKD_TP_1	kbit/s	Non-serving CellACKdThroughput, array of 0..3 cells
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_ACK_1	–	Non-serving CellACKRate, array of 0..3 cells
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_NACK_1	–	Non-serving CellNACKRate, array of 0..3 cells
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_HICH_SUC_1	–	Non-serving CellHICHSuccessRate, array of 0..3 cells
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_UP_1	–	Non-serving CellUPRate, array of 0..3 cells
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_DOWN_1	–	Non-serving CellDOWNRate, array of 0..3 cells
UMTS_MOBILE~UMTS_HSUPA_CELLS~RLS_HOLD_1	–	Non-serving CellHOLDRate, array of 0..3 cells
UMTS_MOBILE~UMTS_HSUPA_RETX_STATISTIC~ UMTS_MOBILE_HSUPA_1_RX	%	Rate of n^{th} transmission was successful, array of 0…15

Appendix V

Parameters with an impact on the effect of blocking

(This appendix does not form an integral part of this Recommendation.)

V.1 What is blocking?

Blocking is a measure of the ability of the receiver to receive a wanted input signal in the presence of an unwanted input signal, without exceeding a given degradation. The degradation is measured in reduction of sensitivity of the receiver (up to complete loss). In case of GSM, the effect only happens if the two test mobiles use the same time slot or fractions of it.

Further information on the definition of blocking can be found in clause 14.7.1.1 of [b-ETSI ETS 300 607-1].

V.2 Which parameters have an impact on the effect of blocking?

- **The selectivity of the receiver (hardware)**

  Minimum requirements for selectivity are defined in the corresponding standards. Better selectivity of a receiver increases the costs of the receiver.

- **The level of the unwanted signal**
- The level of the wanted signal

  The wanted signal is transmitted from the base station to the test mobile. The level of the received signal depends on propagation.

V.3 The standards

Rx blocking analysis for [b-ETSI ETS 300 607-1], clause 14.7.

Frequency	Blocking level GSM900 in dB	Blocking level GSM1800 in dB
835 MHz to < 915 MHz	0	
> 1 000 MHz to 12.75 GHz	–23	
100 KHz to 1 705 MHz		0
> 1 920 MHz to 1 980 MHz		–10

Rx blocking analysis for WCDMA, Table 7.7 of [b-ETSI TS 125 101], out of band blocking (extract).

Parameter	Unit	Frequency range 3
DPCH_Ec	dBm/3.84 MHz	–114
Îor	dBm/3.84 MHz	–103.7
I blocking (CW)	dBm	–15
Fuw	MHz	1 < f < 2 025 2 255 < f < 12 750
UE transmitted mean power	dBm	20/18 (for Power class 3/2)

Possible interactions:

Receiver (wanted signal)	Transmitter (unwanted signal)	Rec. blocking at (dBm)	Transmit. Pwr. (dBm)	Min att. no blocking (dB)
GSM 900	GSM 900	0	33	33
GSM 900	GSM 1800	-23	30	53
GSM 900	UMTS	-23	18	41
GSM 1800	GSM 900	0	33	33
GSM 1800	GSM 1800	-10	30	40
GSM 1800	UMTS	-10	18	28
UMTS	GSM 900	-15	33	48
UMTS	GSM 1800	-15	30	45
UMTS	UMTS	-15	18	33

V.4 The situation

V.4.1 One test mobile transmits, the other one receives

The two test mobiles are connected to the same combiner. Will there be any problems? The combiner isolates the two test mobiles by 20 dB and attenuates the wanted (and the transmitting) signal by 8 dB. In order to prevent blocking, the isolation between the units, using the same time slot, should be higher than 53 dB. When the wanted signal is weak the receiver will show blocking effects. The weak wanted signal will be suppressed by the strong unwanted and reception will no longer be possible. If another channel with sufficient radio level (RxLev) is available, then the test mobile may change the serving cell. If there is no stronger wanted signal available, then the call will drop.

V.4.2 Antennas on the roof of a car

Since the same technical specification of the receiver is applicable, the problem of potential blocking remains basically the same. In order to be on the safe side, the distance between the two test mobiles should be 5 m to 10 m (Interpreting the standards, the 53 dB). If the distance between the two antennas is greater than approximately 1 m, then the isolation of the test mobiles will be higher than the isolation by means of a combiner.

But, if no combiner is used then the wanted and transmitted signals are not attenuated.

(Worst case: weak wanted signal, high transmitting power unwanted signal and overlapping GSM time slots). In wideband code division multiple access (WCDMA) technology the necessary isolation has to be 48 dB (2.5 to 5 m). There is no time slot overlap.

V.4.3 Conclusion

For practical purposes, distances of approximately 5 m between antennas are not practical, even at higher cost.

V.5 Possible solutions

Depending on the purpose of the test (urban vs. rural, coverage vs. benchmarking), the method chosen for combining the test mobiles should be different. In case of benchmarking tests in the urban areas, the use of combiners might not significantly influence the results due to the small cell design, resulting in very high dynamics of the network and many handovers. In rural areas, the range of the system is a dominant issue. Additional loss of the wanted signal in a combiner leads to a considerable risk of losing calls. This is due to the combination of a weak wanted signal with the high output power of the transmitting test mobile.

V.5.1 Attenuators

Refer to [ETSI TR 102 581] for recommendations regarding the usage of attenuators.

V.5.2 Recommendations

V.5.2.1 Benchmarking tests

The most reliable results will be obtained if no combiners are used. However, when using combiners in an urban environment, no significant influence is to be expected.

V.5.2.2 Coverage tests

The use of combiners and/or attenuators is not recommended (8 dB attenuation reduces the coverage down to 25 to 50 per cent). In order to avoid time slot collisions, only one network should be tested at any one time.

V.5.3 Other equipment involved

V.5.3.1 Scanners

Although scanners have an outstanding dynamic range, the filters used have limitations. This makes the scanner vulnerable to blocking.

V.5.3.2 GPS receiver 1.2/1.5 GHz

GPS receivers usually use high gain antennas. This fact makes them sensitive for blocking (the distance between the satellite and the GPS receiver is approximately 20 000 km).

V.5.3.3 Risk for scanner and GPS receiver

Malfunction of scanner and GPS might be possible, depending on the transmit power of the test equipment. Antennas for scanners and GPS receiver should be spaced to the practical maximum.

V.5.4 Mixed service GSM/WCDMA

It is recommended to use separate antennas for GSM and WCDMA, with maximum possible spacing.

Appendix VI

Reference SMS or SDS message

(This appendix does not form an integral part of this Recommendation.)

Content integrity of single messages for the {SMS | SDS} is ensured by mechanisms on lower protocol layers of GSM, UMTS and TETRA networks, respectively. Thus, there is – from an end-to-end testing perspective – no need to implement content integrity checking mechanisms on top of the {SMS | SDS}. Therefore, no reference message is provided by this Recommendation.

Appendix VII

Content integrity checking

(This appendix does not form an integral part of this Recommendation.)

Content integrity checking can be achieved by placing meta-information about the expected content in the retrieved documents and check if the content description matches the received content. If the description is put in the text payload of the content, it should survive compression and transcoding of the content during transportation from the server to the client.

VII.1 HTTP

The text of the main reference page should contain the following description language in the main file of a test webpage (typically index.html). The text should substitute text present before in order to keep the length of the reference page constant. In order to avoid misinterpretation of the content description by common client software, established standards like XML or HTML should be avoided.

The structure of the reference description is as follows:

(% TAGNAME VALUE, [VALUE] %)

Before each information element (IE), an opening mark should be used, followed by a blank. The opening mark is a "(%" (bracket open, followed by a per cent sign). The IE itself is identified by a tag called TAGNAME. Below is a list of all valid TAGNAMES. Each IE has one or more parameters, called VALUES, separated by a comma. After the IE and its parameters a closing mark should be used. The closing mark is a "%)" (per cent sign, followed by a bracket close). The opening and closing tags are separated from the TAGNAME and its parameters by a single blank space. The complete element should not be included within any HTML-format construct but should be placed in pure text payload.

TAGNAME PARAMETERS

NAME <RESOURCENAME>, <APPROVED>

This IE contains the name of the reference webpage. The first parameter <RESOURCENAME> describes the reference page. The second parameter <APPROVED> is set to 0 for a resource not approved by ETSI and set to 1 for an ETSI approved resource.

VERSION <VERSIONNUMBER>

This IE contains a unique version number of this specific resource.

TSIZE <TBYTES>

This IE contains the accumulated size of all objects of the complete resource in bytes.

OBJECTS <NUMBER>

This IE contains the accumulated number of objects belonging to this specific page.

OBJECT <OID>,<OFNAME>,<ONAME>,<OSIZE>, <OREQUIRED>

For each object of the resource (identified by a separate file) an OBJECT tag should be available. Each OBJECT tag has 5 mandatory parameters:

<OID> contains a unique identifier per object on this page, starting the count from 1 for the main index-file that includes the content description.

<OFNAME> contains the filename of the referenced object.

<ONAME> contains a description of the object.

<OSIZE> contains the original size of the object in bytes.

<OREQUIRED> is set to 1 if the object relevant for user perception. If the object may be removed while in transit, the value should be set to 0 (e.g., for advertisements).

INFO <TEXT>

This IE contains additional information about the resource.

VII.2 FTP

Content integrity checking of an object transferred by FTP should be done by the use of message-digest algorithm 5 (MD5) checksums (16 bytes long). Two methods are supported:

Method 1:

In the same directory of the FTP server where the reference file is located lays a second file containing the MD5 checksum of the reference file (identified by the same name, but a file name extension.MD5). By downloading both files, the test client can determine the content integrity of the reference file by calculating its MD5 checksum and comparing it to the value contained in the checksum file.

To identify that method 1 is used, the filename of the reference file starts with the fragment "CIC_M1_" followed by the name of the file plus extension.

Example 1: DEMO.TXT becomes CIC_M1_DEMO.TXT and CIC_M1_DEMO.MD5.

Method 2:

The MD5 checksum of the file is appended to the original reference file, increasing its size by 16 bytes. For content integrity check, the test client cuts the last 16 bytes of the downloaded file and calculates the MD5 checksum of the remaining fragment. This checksum can be compared to the checksum received with the last 16 bytes of the file.

To identify that method 2 is used the filename of the reference file starts with the fragment "CIC_M2_" followed by the name of the file plus extension.

Example 2: DEMO.TXT becomes CIC_M2_DEMO.TXT.

VII.3 MMS

For further study.

Appendix VIII

Transfer times versus used data rate and content size

(This appendix does not form an integral part of this Recommendation.)

Table VIII.1 provides transfer times versus used data rate and content size.

Table VIII.1 – Transfer times versus used data rate and content size

		Size						
		100 kbytes (102 400 bytes)	**200 kbytes (204 800 bytes)**	**500 kbytes (512 000 bytes)**	**1 Mbytes (1 048 576 bytes)**	**2 Mbytes (2 097 152 bytes)**	**5 Mbytes (5 242 880 bytes)**	**10 Mbytes (10 485 760 bytes)**
Data rate	10 kbit/s	81.92 s	163.84 s	409.6 s	838.9 s	1 677.7 s	4 194.3 s	8 388.6 s
	20 kbit/s	40.96 s	81.92 s	204.8 s	419.4 s	838.9 s	2 097.1 s	4 194.3 s
	40 kbit/s	20.48 s	40.96 s	102.4 s	209.7 s	419.4 s	1 048.6 s	2 097.2 s
	80 kbit/s	10.24 s	20.48 s	51.2 s	104.9 s	209.7 s	524.3 s	1 048.6 s
	160 kbit/s	5.12 s	10.24 s	25.6 s	52.4 s	104.9 s	262.1 s	524.3 s
	320 kbit/s	2.56 s	5.12 s	12.8 s	26.2 s	52.4 s	131.1 s	262.1 s
	640 kbit/s	1.28 s	2.56 s	6.4 s	13.1 s	26.2 s	65.5 s	131.1 s
	1 280 kbit/s	0.64 s	1.28 s	3.2 s	6.6 s	13.1 s	32.8 s	65.5 s
	2 560 kbit/s	0.32 s	0.64 s	1.6 s	3.3 s	6.6 s	16.4 s	32.8 s
	5 120 kbit/s	0.16 s	0.32 s	0.8 s	1.6 s	3.3 s	8.2 s	16.4 s
	10 240 kbit/s	0.08 s	0.16 s	0.4 s	0.8 s	1.6 s	4.1 s	8.2 s
	20 480 kbit/s	0.04 s	0.08 s	0.2 s	0.4 s	0.8 s	2.1 s	4.1 s

Appendix IX

Examples of statistical calculations

(This appendix does not form an integral part of this Recommendation.)

In the following, some example computations are given for different topics. All computations are done step by step as well as by applying statistical software tools. The statistical software mainly applied here is the open source language and environment R, based on the S language which is widely spread in the statistics area. For further information and download, the reader is referred to:

- http://www.r-project.org

R is a piece of freely distributed software. Its installation is straightforward and is commonly done within five minutes. For further applications and more sophisticated statistical methods, a number of libraries are available from the website. For creating graphics and first steps in programming see also [b-Venables]. For reliable results, the use of R is highly recommended.

As Excel is a standard software used for calculations, there are some commands which are given for Excel-users. Nevertheless, it must be said that Excel does not have a uniform definition for the computation of different expressions or operators, e.g., quantiles. Most of the mathematical functions are defined only with regard to specific desired tests. The user should therefore be warned to use any of Excels mathematical procedures without a deeper understanding of the functionality differences between these procedures.

IX.1 Confidence intervals for binomial distribution

This example tries to clarify the usage of the Pearson-Clopper formula which is related to the binomial distribution and may be used for measurements with a small amount of measurement data.

Example: During a one hour manual test of service X, the service access attempts lead to the following results ("0" represents an unsuccessful attempt, "1" a successful attempt).

No. 1-10	1	0	1	0	1	1	1	1	0	0
No. 11-20	0	1	1	1	1	0	1	1	0	1
No. 21-30	1	1	0	1	1	1	1	1	1	1
No. 31-40	0	1	0	0	1	1	1	1	1	1

Within the $n = 40$ attempts, $m = 29$ have been successful. The point estimation leads to

$$p = \frac{m}{n} = \frac{29}{40} = 0.725$$

IX.1.1 Step by step computation

Since the results show a binary outcome, the binomial distribution can be applied in any case. At first, the allowance of the easier to handle Normal distribution has to be checked via the following expression:

$$n \times p \times (1 - p) = 7.975 < 9$$

Therefore, the normal distribution should not be used for this measurement. Furthermore, the following computations are directly based on the binomial distribution.

If the required confidence level is defined as $1-\alpha=0.95$, the resulting α value is $\alpha=0.05$. According to this, the Pearson-Clopper formulas now read:

$$p_1=\frac{m\cdot F_{2m,2(n-m+1);\frac{\alpha}{2}}}{n-m+1+m\cdot F_{2m,2(n-m+1);\frac{\alpha}{2}}}=\frac{29\cdot F_{58,24;0,025}}{12+29\cdot F_{58,24;0,025}}$$

$$p_2=\frac{(m+1)\cdot F_{2(m+1),2(n-m);1-\frac{\alpha}{2}}}{n-m+(m+1)\cdot F_{2(m+1),2(n-m);1-\frac{\alpha}{2}}}=\frac{30\cdot F_{60,22;0,975}}{11+30\cdot F_{60,22;0,975}}$$

Eventually, four steps have to be executed to get the relevant confidence interval:

1) Lookup if the needed quantile values of the F distribution are tabulated.

2) If the quantiles are not tabulated, try the relation $F_{n_1,n_2;1-\gamma}=\frac{1}{F_{n_2,n_1;\gamma}}$ to get the required information.

3) If both approaches do not succeed, try the approximation $F_{n_1,n_2;\gamma}\cong\exp(u\cdot a-b)$ for γ in the range 0.5< γ <1.

4) Determine the confidence interval by using the quantile values.

Now, the quantiles $F_{58,24;0,025}$ and $F_{60,22;0,975}$ have to be retrieved before the Pearson-Clopper values are computable.

1) Looking up some tabulated quantile values may lead to the following results:

$$F_{60,22;0,975}=2{,}145$$

If the quantile $F_{58,24;0,025}$ cannot be found, the following steps may be appropriate.

2) If $F_{58,24;0,025}$ is missing in the tables, perhaps the quantile

$$F_{24,58;0,975}=\frac{1}{F_{58,24;0.025}}$$

is available. If this is also not the case, a first sight approximation is given by a neighbouring quantile value:

$$F_{24,60;0,975}=\frac{1}{F_{60,24;0,025}}=1{,}882$$

$$\Rightarrow F_{60,24;0,025}=\frac{1}{F_{24,60;0,975}}=0{,}5313$$

3) Since the quantile variable $\gamma=0.975$ lies in the range of 0.5< γ <1, the approximation

$$F_{n_1,n_2;\gamma}\cong\exp(u\cdot a-b)$$

can be applied. Therefore, the following computational steps have to be executed to determine $F_{24,58;0.975}$ in a more precise way:

– At first, the parameter d is done:

$$d=\frac{1}{n_1-1}+\frac{1}{n_2-1}=\frac{1}{24-1}+\frac{1}{58-1}\approx 0{,}06102$$

– Before computing $c=\frac{(u_\gamma)^2-3}{6}$, the 0.975-quantile of the standard normal distribution $N(0,1)$ has to be retrieved from a table:

$$\gamma=0{,}975 \Rightarrow \quad u_{0{,}975}=1{,}96$$

So c reads

$$c=\frac{(1{,}96)^2-3}{6}=0{,}1402\overline{6}$$

– As a result, b is given by:

$$b=2\cdot\left(\frac{1}{n_1-1}-\frac{1}{n_2-1}\right)\cdot\left(c+\frac{5}{6}-\frac{d}{3}\right)=2\cdot\left(\frac{1}{24-1}-\frac{1}{58-1}\right)\cdot\left(0{,}14026+\frac{5}{6}-\frac{0{,}06102}{3}\right)=0{,}04944$$

– With these results, a leads to:

$$a=\sqrt{2d+cd^2}=\sqrt{2\cdot 0{,}06102+0{,}14026\cdot 0.06102^2}=0{,}35$$

– Finally, the approximation for the quantile value reads:

$$F_{58,24;0{,}975}\cong\exp(1.96\cdot 0.35-0.04944)=1.8899$$

The originally searched quantile value $F_{58,24;0{,}025}=\frac{1}{F_{24,58;0.975}}$ results then in:

$$F_{58,24;0{,}025}=\frac{1}{1{,}8899}=0{,}5291$$

4) After the quantiles of the F distribution are known, in the last step the Pearson-Clopper values itself can be determined:

$$p_1=\frac{m\cdot F_{2m,2(n-m+1);\frac{\alpha}{2}}}{n-m+1+m\cdot F_{2m,2(n-m+1);\frac{\alpha}{2}}}=\frac{29\cdot F_{58,24;0{,}025}}{12+29\cdot F_{58,24;0{,}025}}=\frac{29\cdot 0{,}5291}{12+29\cdot 0{,}5291}=0{,}5611$$

$$p_2=\frac{(m+1)\cdot F_{2(m+1),2(n-m);1-\frac{\alpha}{2}}}{n-m+(m+1)\cdot F_{2(m+1),2(n-m);1-\frac{\alpha}{2}}}=\frac{30\cdot F_{60,22;0{,}975}}{11+30\cdot F_{60,22;0{,}975}}=\frac{30\cdot 2{,}145}{11+30\cdot 2{,}145}=0{,}854$$

With these values, the confidence interval for the given measurement can be described as:

$$[p_1;p_2]=[0.5611;0.854]$$

IX.1.2 Computation using statistical software

The different calculations can be executed by R. To enable a user-oriented simplicity, the according expressions are given in the next clauses.

IX.1.2.1 Computation in R

Required quantiles of the F-distribution may also be obtained in R. Here no approximation is carried out. Commands (marked by ">") and results (marked by "[1..]") are given by:

```
> qf(0.025, 24, 58)
[1] 0.4797205

> qf(0.975, 60, 22)
[1] 2.144594
```

Alternatively, a function can be defined for computing the Pearson-Clopper confidence interval directly. This function takes the following input variables:

- n: Number of trials (total);
- m: Number of (un-)successful interesting trials, either successful ones or non-successful ones;
- alpha: desired confidence level (default: 5% 1- alpha = 95%, means alpha = 5% = 0,05).

The output consists of:

- estimator: The estimated value of the according rate parameter;
- confidence interval for the estimator (lower and upper bound).

The function is given by:

```
pearson.clopper <- function(n, m, alpha = 0.05) {
    # computation of F-quantiles
    f1 <- qf(alpha/2,2*m,2*(n-m+1))
    f2 <- qf(1-alpha/2,2*(m+1),2*(n-m))
    # computation of confidence limits
    p1 <- m * f1/(n-m+1+m*f1)
    p2 <- (m+1)*f2/(n-m+(m+1)*f2)
    out <- list(estimator = m/n, confidence.interval = c(p1, p2))
    return(out)
}
```

The function is applied by calling it with the required arguments. The result for the above example is given by:

```
> pearson.clopper(40, 29)
$estimator
[1] 0.725

$confidence.interval
[1] 0.5611171 0.8539910
```

IX.1.2.2 Computation in Excel

In Excel, quantiles of the F-distribution are derived by applying the functions:

FINV (p-value;df1;df2)

Related to the use of Excel, it is very important to have a very clear understanding what is done by a certain expression. For example, the calculation of FINV depends on the parameter p-value which is NOT the same as the parameter "alpha" in the R section!

IX.2 Transition from binomial to normal distribution

To use of the transition from a binomial distribution to a normal one, the condition:

$$n \times p \times q = n \times p \times (1 - p) \geq 9$$

has to be fulfilled.

Example 1: If n = 30 samples are gathered which lead to an estimated rate of p = 0.8, the condition reads:

$$n \times p \times q = n \times p \times (1 - p) = 30 \times 0.8 \times 0.2 = 4.8 < 9$$

This means, the approximation is not allowed and confidence intervals have to be calculated with the Pearson-Clopper formula.

Example 2: Now, the same rate p = 0.8 is estimated on a basis of n = 300 samples. The according relation reads

$$n \times p \times q = n \times p \times (1-p) = 300 \times 0.8 \times 0.2 = 48 > 9$$

In this case, the approximation of the binomial distribution by a normal distribution is allowed. The confidence intervals can be calculated with the according expressions of the normal distribution.

IX.3 Definitions of EG 201 769

The following clause presents another definition of confidence intervals related to the normal distribution. It can be found in [b-ETSI EG 201 769]:

- Relationship between the accuracy of estimator of the unsuccessful call ratio and the number of calls to be observed.

If k unsuccessful calls are observed out of N call attempts, then the true value of the unsuccessful call ratio lies between $\frac{k}{N}-\Delta$ and $\frac{k}{N}+\Delta$ with a confidence level $1-\alpha$, Δ being approximated (for large value of N) by:

$$\Delta \approx \sigma(\alpha) \times \sqrt{\frac{p \times (1-p)}{N}}$$

where p is the expected unsuccessful call ratio and $\sigma(\alpha)$ is the $(1-\frac{\alpha}{2}) \times 100$ percentile of the normal distribution with mean 0 and standard deviation 1 (N(0,1)). I.e., the number of call attempts to be observed should be:

$$N = \frac{\sigma(\alpha)^2 \times p(1-p)}{\Delta^2}$$

- If the confidence level is $1-\alpha = 0.95$ then $\sigma\ (\alpha) = 1.96 \approx 2.$
- If the required accuracy for $p \leq 0.01$ is $\Delta = 0.001$, then the number of call attempts to be observed should be $N = 4 \times 10^6 \times p \times (1-p)$ for a confidence level of 95 per cent.
- If the required accuracy for $p > 0.01$ is,

$$\frac{\Delta}{p} = 0.1$$

then the number of call attempts to be observed should be $N = 400 \times ((1-p)/p)$ for a confidence level of 95 per cent.

- For example, if the expected unsuccessful call ratio is 1 per cent, the number of call attempts to be observed should be

  $$N = 4 \times 10^6 \times 0.01 \times (1-0.01) = 39\,600$$

  for an accuracy of $\Delta = 0.001$ with a confidence level of 95 per cent.
- If the unsuccessful call ratio is expected to be 3 per cent, then the number of call attempts should be:

  $$N = 400 \times ((1-0.03) / 0.03) \times 13\,000$$

  for a relative accuracy of $\frac{\Delta}{p} = 0{,}1$ and with a confidence level of 95 per cent.

IX.4 Calculation of confidence intervals

This clause provides more information about the calculation of confidence intervals. Due to the possibility that also small numbers may occur, for example if service probing is done manually, the calculation of confidence intervals is based on the relations given by the Pearson-Clopper expressions.

The structure of this clause is as follows:

- Starting with some visualized use cases of the Pearson-Clopper formulas, an impression of the relationship between the estimated rate value and the corresponding confidence interval is given.

IX.4.1 Estimated rate of 5 per cent

The confidence interval gets smaller with an increasing number of available samples. The less data is available, the higher the uncertainty is. Another effect which is covered by the Pearson-Clopper approach is the asymmetric behaviour of the upper and lower limits of the confidence interval. Additionally, this asymmetric behaviour depends on the estimated rate values (see Figures IX.4.1 to IX.4.3).

Some further remarks might be helpful:

- The confidence interval can be calculated for rather small sample sizes.
- An overestimation which would have appeared by applying the normal (Gaussian) approximation is not recognizable.
- If a rate value is equal to 0 per cent, this is also the value of the lower limit of the confidence interval. The calculation of quantiles of the F-distribution is not valid in this case.
- If a rate value is equal to 100 per cent, this is also the value of the upper limit of the confidence interval. The calculation of quantiles of the F-distribution is not valid in this case.

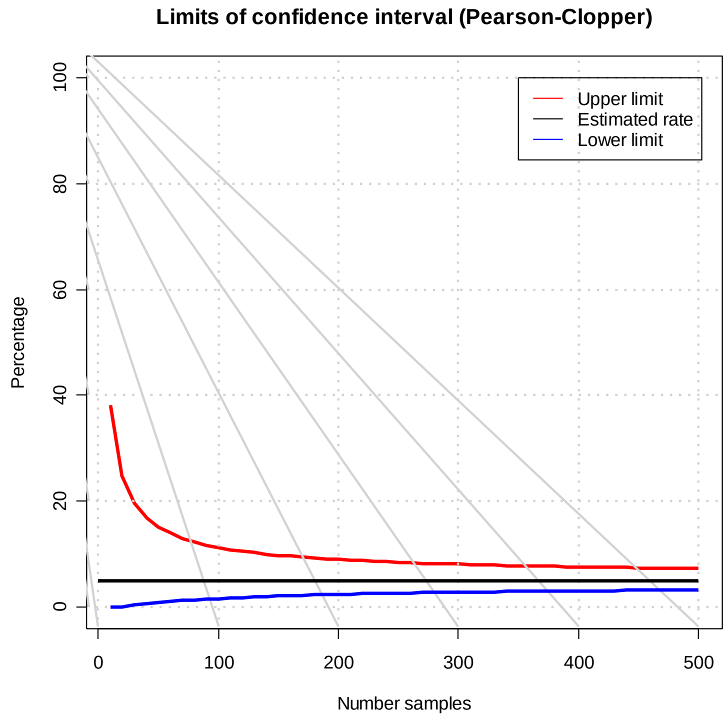


Figure IX.4.1 – Confidence interval for estimated rate of 5 per cent

The depicted limit curves can be found in the columns of the Tables IX.4.1 to IX.4.3 (estimated rate is constant, number of measurements varies).

IX.4.2 Estimated rate of 50 per cent

In Figure IX.4.2 the confidence interval for an estimated rate of 50 per cent is depicted. In this case the confidence interval has a symmetric behaviour.

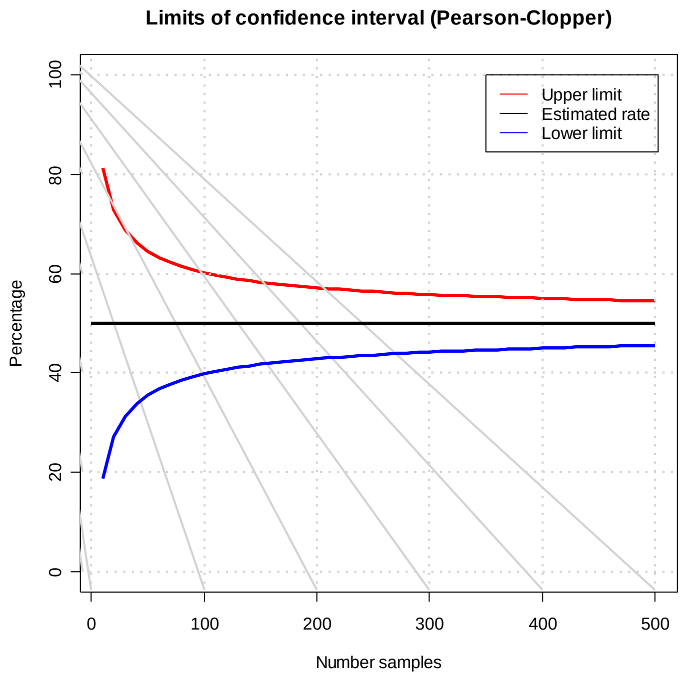


Figure IX.4.2 – Confidence interval for estimated rate of 50 per cent

IX.4.3 Estimated rate of 95 per cent

Figure IX.4.3 describes the situation according to a 95 per cent rate. The situation is comparable with the graph of the 5 per cent rate.

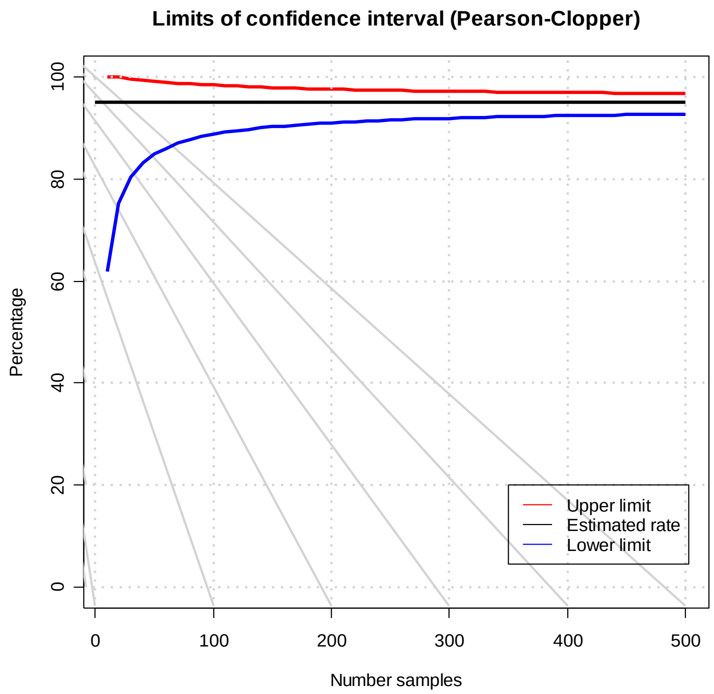


Figure IX.4.3 – Confidence interval for estimated rate of 95 per cent

IX.4.4 Lower limit of confidence intervals according to Pearson-Clopper formula

Table IX.1 contains values which specify the lower limit of the confidence interval. The lower limit depends on the number of samples and the according rate value. In Figures IX.4.1 to IX.4.3 this information can be found at the blue lines.

Table IX.1 – Lower limit of confidence intervals according to Pearson-Clopper formula

	Rate																				
NrMeas	1%	5%	10%	15%	20%	25%	30%	35%	40%	45%	50%	55%	60%	65%	70%	75%	80%	85%	90%	95%	100%
100	0.03%	1.64%	4.90%	8.65%	12.67%	16.88%	21.24%	25.73%	30.33%	35.00%	39.80%	44.70%	49.72%	54.82%	60.02%	65.29%	70.74%	76.37%	82.24%	88.48%	95.15%
200	0.12%	2.42%	6.22%	10.35%	14.69%	19.16%	23.74%	28.41%	33.15%	37.96%	42.86%	47.82%	52.85%	57.95%	63.14%	68.38%	73.75%	79.25%	84.93%	90.91%	97.53%
300	0.21%	2.83%	6.85%	11.16%	15.62%	20.20%	24.87%	29.61%	34.41%	39.28%	44.19%	49.17%	54.21%	59.31%	64.46%	69.69%	75.00%	80.43%	86.01%	91.84%	98.34%
400	0.27%	3.08%	7.24%	11.65%	16.19%	20.83%	25.55%	30.33%	35.16%	40.05%	44.99%	49.98%	55.01%	60.10%	65.25%	70.45%	75.73%	81.11%	86.61%	92.35%	98.75%
500	0.33%	3.26%	7.51%	11.98%	16.58%	21.26%	26.01%	30.82%	35.67%	40.58%	45.53%	50.52%	55.56%	60.64%	65.77%	70.96%	76.21%	81.55%	87.02%	92.69%	99.00%
600	0.37%	3.40%	7.72%	12.24%	16.87%	21.58%	26.36%	31.18%	36.05%	40.97%	45.92%	50.92%	55.96%	61.03%	66.15%	71.33%	76.57%	81.88%	87.31%	92.92%	99.17%
700	0.40%	3.51%	7.88%	12.44%	17.10%	21.83%	26.62%	31.46%	36.35%	41.27%	46.23%	51.23%	56.26%	61.34%	66.45%	71.62%	76.84%	82.13%	87.53%	93.10%	99.29%
800	0.43%	3.60%	8.01%	12.60%	17.28%	22.03%	26.84%	31.69%	36.58%	41.51%	46.48%	51.48%	56.51%	61.58%	66.69%	71.85%	77.05%	82.33%	87.70%	93.24%	99.37%
900	0.46%	3.67%	8.12%	12.73%	17.43%	22.20%	27.02%	31.88%	36.78%	41.71%	46.68%	51.68%	56.72%	61.78%	66.89%	72.03%	77.23%	82.49%	87.85%	93.36%	99.44%
1 000	0.48%	3.73%	8.21%	12.84%	17.56%	22.34%	27.17%	32.04%	36.95%	41.88%	46.85%	51.85%	56.89%	61.95%	67.05%	72.19%	77.38%	82.63%	87.97%	93.45%	99.50%
1 100	0.50%	3.79%	8.29%	12.94%	17.67%	22.47%	27.30%	32.18%	37.09%	42.03%	47.00%	52.00%	57.04%	62.10%	67.19%	72.33%	77.51%	82.75%	88.07%	93.54%	99.54%
1 200	0.52%	3.84%	8.36%	13.03%	17.77%	22.57%	27.42%	32.30%	37.21%	42.16%	47.13%	52.13%	57.16%	62.23%	67.32%	72.45%	77.62%	82.85%	88.16%	93.61%	99.58%
1 300	0.53%	3.88%	8.42%	13.10%	17.86%	22.67%	27.52%	32.40%	37.32%	42.27%	47.25%	52.25%	57.28%	62.34%	67.43%	72.55%	77.72%	82.94%	88.24%	93.67%	99.61%
1 400	0.55%	3.92%	8.48%	13.17%	17.93%	22.75%	27.61%	32.50%	37.42%	42.37%	47.35%	52.35%	57.38%	62.44%	67.52%	72.64%	77.81%	83.02%	88.31%	93.72%	99.64%
1 500	0.56%	3.95%	8.53%	13.23%	18.00%	22.83%	27.69%	32.58%	37.51%	42.46%	47.44%	52.44%	57.47%	62.53%	67.61%	72.73%	77.88%	83.09%	88.37%	93.77%	99.67%
1 600	0.57%	3.98%	8.57%	13.28%	18.07%	22.89%	27.76%	32.66%	37.59%	42.54%	47.52%	52.52%	57.55%	62.61%	67.69%	72.80%	77.95%	83.15%	88.42%	93.81%	99.69%
1 700	0.58%	4.01%	8.61%	13.33%	18.12%	22.96%	27.83%	32.73%	37.66%	42.62%	47.60%	52.60%	57.63%	62.68%	67.76%	72.87%	78.02%	83.21%	88.47%	93.85%	99.71%
1 800	0.59%	4.04%	8.65%	13.38%	18.17%	23.01%	27.89%	32.79%	37.73%	42.68%	47.66%	52.67%	57.69%	62.75%	67.82%	72.93%	78.08%	83.26%	88.52%	93.89%	99.72%
1 900	0.60%	4.06%	8.69%	13.42%	18.22%	23.07%	27.95%	32.85%	37.79%	42.75%	47.73%	52.73%	57.76%	62.81%	67.88%	72.99%	78.13%	83.31%	88.56%	93.92%	99.74%
2 000	0.61%	4.09%	8.72%	13.46%	18.27%	23.12%	28.00%	32.91%	37.84%	42.80%	47.78%	52.79%	57.81%	62.86%	67.94%	73.04%	78.18%	83.36%	88.60%	93.95%	99.75%
2 100	0.62%	4.11%	8.75%	13.50%	18.31%	23.16%	28.05%	32.96%	37.90%	42.86%	47.84%	52.84%	57.87%	62.92%	67.99%	73.09%	78.22%	83.40%	88.64%	93.98%	99.76%
2 200	0.63%	4.13%	8.78%	13.53%	18.35%	23.20%	28.09%	33.01%	37.94%	42.91%	47.89%	52.89%	57.92%	62.97%	68.04%	73.13%	78.27%	83.44%	88.67%	94.00%	99.77%
2 300	0.63%	4.15%	8.80%	13.56%	18.38%	23.24%	28.13%	33.05%	37.99%	42.95%	47.94%	52.94%	57.96%	63.01%	68.08%	73.18%	78.31%	83.47%	88.70%	94.03%	99.78%
2 400	0.64%	4.16%	8.83%	13.59%	18.42%	23.28%	28.17%	33.09%	38.03%	43.00%	47.98%	52.98%	58.01%	63.05%	68.12%	73.22%	78.34%	83.51%	88.73%	94.05%	99.79%
2 500	0.65%	4.18%	8.85%	13.62%	18.45%	23.31%	28.21%	33.13%	38.07%	43.04%	48.02%	53.02%	58.05%	63.09%	68.16%	73.25%	78.38%	83.54%	88.76%	94.07%	99.80%
2 600	0.65%	4.19%	8.87%	13.65%	18.48%	23.35%	28.24%	33.16%	38.11%	43.07%	48.06%	53.06%	58.09%	63.13%	68.20%	73.29%	78.41%	83.57%	88.78%	94.09%	99.81%
2 700	0.66%	4.21%	8.89%	13.67%	18.51%	23.38%	28.28%	33.20%	38.15%	43.11%	48.10%	53.10%	58.12%	63.17%	68.23%	73.32%	78.44%	83.60%	88.81%	94.11%	99.81%
2 800	0.67%	4.22%	8.91%	13.70%	18.53%	23.41%	28.31%	33.23%	38.18%	43.15%	48.13%	53.13%	58.16%	63.20%	68.26%	73.35%	78.47%	83.62%	88.83%	94.13%	99.82%

Table IX.1 – Lower limit of confidence intervals according to Pearson-Clopper formula

	Rate																				
2 900	0.67%	4.24%	8.93%	13.72%	18.56%	23.43%	28.34%	33.26%	38.21%	43.18%	48.16%	53.17%	58.19%	63.23%	68.29%	73.38%	78.50%	83.65%	88.85%	94.14%	99.83%
3 000	0.68%	4.25%	8.95%	13.74%	18.58%	23.46%	28.36%	33.29%	38.24%	43.21%	48.19%	53.20%	58.22%	63.26%	68.32%	73.41%	78.52%	83.67%	88.87%	94.16%	99.83%
3 100	0.68%	4.26%	8.97%	13.76%	18.60%	23.48%	28.39%	33.32%	38.27%	43.24%	48.22%	53.23%	58.25%	63.29%	68.35%	73.44%	78.55%	83.69%	88.89%	94.17%	99.84%
3 200	0.68%	4.27%	8.98%	13.78%	18.63%	23.51%	28.42%	33.35%	38.30%	43.27%	48.25%	53.26%	58.28%	63.32%	68.38%	73.46%	78.57%	83.71%	88.91%	94.19%	99.84%
3 300	0.69%	4.28%	9.00%	13.80%	18.65%	23.53%	28.44%	33.37%	38.32%	43.29%	48.28%	53.28%	58.31%	63.34%	68.40%	73.49%	78.59%	83.74%	88.92%	94.20%	99.85%
3 400	0.69%	4.29%	9.01%	13.82%	18.67%	23.55%	28.46%	33.40%	38.35%	43.32%	48.31%	53.31%	58.33%	63.37%	68.43%	73.51%	78.61%	83.75%	88.94%	94.21%	99.85%
3 500	0.70%	4.30%	9.03%	13.83%	18.69%	23.57%	28.48%	33.42%	38.37%	43.34%	48.33%	53.33%	58.35%	63.39%	68.45%	73.53%	78.64%	83.77%	88.96%	94.22%	99.86%
3 600	0.70%	4.31%	9.04%	13.85%	18.70%	23.59%	28.51%	33.44%	38.39%	43.37%	48.35%	53.36%	58.38%	63.42%	68.47%	73.55%	78.65%	83.79%	88.97%	94.24%	99.86%
3 700	0.71%	4.32%	9.05%	13.86%	18.72%	23.61%	28.53%	33.46%	38.42%	43.39%	48.38%	53.38%	58.40%	63.44%	68.49%	73.57%	78.67%	83.81%	88.99%	94.25%	99.86%
3 800	0.71%	4.33%	9.06%	13.88%	18.74%	23.63%	28.55%	33.48%	38.44%	43.41%	48.40%	53.40%	58.42%	63.46%	68.51%	73.59%	78.69%	83.82%	89.00%	94.26%	99.87%
3 900	0.71%	4.34%	9.08%	13.89%	18.75%	23.65%	28.56%	33.50%	38.46%	43.43%	48.42%	53.42%	58.44%	63.48%	68.53%	73.61%	78.71%	83.84%	89.01%	94.27%	99.87%
4 000	0.72%	4.35%	9.09%	13.91%	18.77%	23.66%	28.58%	33.52%	38.48%	43.45%	48.44%	53.44%	58.46%	63.50%	68.55%	73.63%	78.73%	83.86%	89.03%	94.28%	99.87%
4 100	0.72%	4.35%	9.10%	13.92%	18.79%	23.68%	28.60%	33.54%	38.50%	43.47%	48.46%	53.46%	58.48%	63.52%	68.57%	73.64%	78.74%	83.87%	89.04%	94.29%	99.88%
4 200	0.72%	4.36%	9.11%	13.93%	18.80%	23.70%	28.62%	33.56%	38.51%	43.49%	48.48%	53.48%	58.50%	63.54%	68.59%	73.66%	78.76%	83.88%	89.05%	94.30%	99.88%
4 300	0.72%	4.37%	9.12%	13.95%	18.81%	23.71%	28.63%	33.57%	38.53%	43.51%	48.49%	53.50%	58.52%	63.55%	68.61%	73.68%	78.77%	83.90%	89.06%	94.30%	99.88%
4 400	0.73%	4.37%	9.13%	13.96%	18.83%	23.73%	28.65%	33.59%	38.55%	43.52%	48.51%	53.52%	58.53%	63.57%	68.62%	73.69%	78.79%	83.91%	89.08%	94.31%	99.89%
4 500	0.73%	4.38%	9.14%	13.97%	18.84%	23.74%	28.66%	33.61%	38.56%	43.54%	48.53%	53.53%	58.55%	63.59%	68.64%	73.71%	78.80%	83.92%	89.09%	94.32%	99.89%
4 600	0.73%	4.39%	9.15%	13.98%	18.85%	23.75%	28.68%	33.62%	38.58%	43.56%	48.54%	53.55%	58.57%	63.60%	68.65%	73.72%	78.81%	83.93%	89.10%	94.33%	99.89%
4 700	0.74%	4.39%	9.16%	13.99%	18.86%	23.77%	28.69%	33.64%	38.60%	43.57%	48.56%	53.56%	58.58%	63.62%	68.67%	73.74%	78.83%	83.95%	89.11%	94.34%	99.89%
4 800	0.74%	4.40%	9.17%	14.00%	18.88%	23.78%	28.71%	33.65%	38.61%	43.59%	48.58%	53.58%	58.60%	63.63%	68.68%	73.75%	78.84%	83.96%	89.12%	94.34%	99.90%
4 900	0.74%	4.41%	9.17%	14.01%	18.89%	23.79%	28.72%	33.66%	38.62%	43.60%	48.59%	53.59%	58.61%	63.65%	68.69%	73.76%	78.85%	83.97%	89.13%	94.35%	99.90%
5 000	0.74%	4.41%	9.18%	14.02%	18.90%	23.80%	28.73%	33.68%	38.64%	43.61%	48.60%	53.61%	58.63%	63.66%	68.71%	73.78%	78.86%	83.98%	89.13%	94.36%	99.90%

IX.4.5 Upper limit of confidence intervals according to Pearson-Clopper formula

Table IX.2 contains values which specify the upper limit of the confidence interval. The upper limit depends on the number of samples and the according rate value. In Figures IX.4.1 to IX.4.3 this information can be found at the red lines.

Table IX.2 – Upper limit of confidence intervals according to Pearson-Clopper formula

	Rate																				
NrMeas	1%	5%	10%	15%	20%	25%	30%	35%	40%	45%	50%	55%	60%	65%	70%	75%	80%	85%	90%	95%	100%
100	5.45%	11.28%	17.62%	23.53%	29.18%	34.66%	39.98%	45.18%	50.28%	55.30%	60.19%	64.98%	69.67%	74.27%	78.76%	83.18%	87.41%	91.44%	95.20%	98.48%	100.00%
200	3.57%	9.00%	15.02%	20.72%	26.22%	31.60%	36.87%	42.05%	47.15%	52.18%	57.14%	62.03%	66.85%	71.59%	76.26%	80.86%	85.34%	89.68%	93.82%	97.63%	100.00%
300	2.89%	8.11%	13.97%	19.55%	24.98%	30.30%	35.53%	40.69%	45.79%	50.82%	55.80%	60.73%	65.59%	70.39%	75.14%	79.81%	84.39%	88.86%	93.17%	97.21%	100.00%
400	2.54%	7.62%	13.37%	18.88%	24.26%	29.54%	34.75%	39.90%	44.99%	50.03%	55.01%	59.95%	64.84%	69.67%	74.45%	79.18%	83.82%	88.37%	92.77%	96.94%	100.00%
500	2.32%	7.29%	12.97%	18.44%	23.78%	29.04%	34.23%	39.36%	44.45%	49.48%	54.47%	59.42%	64.32%	69.18%	73.99%	78.74%	83.43%	88.02%	92.50%	96.75%	100.00%
600	2.16%	7.06%	12.68%	18.11%	23.43%	28.67%	33.84%	38.97%	44.05%	49.08%	54.08%	59.03%	63.95%	68.82%	73.65%	78.42%	83.13%	87.77%	92.29%	96.61%	100.00%
700	2.05%	6.89%	12.47%	17.86%	23.16%	28.38%	33.55%	38.66%	43.74%	48.77%	53.77%	58.73%	63.65%	68.54%	73.38%	78.17%	82.91%	87.57%	92.13%	96.50%	100.00%
800	1.96%	6.75%	12.29%	17.67%	22.94%	28.15%	33.31%	38.42%	43.49%	48.52%	53.52%	58.49%	63.42%	68.31%	73.16%	77.97%	82.72%	87.41%	92.00%	96.41%	100.00%
900	1.89%	6.63%	12.15%	17.50%	22.77%	27.96%	33.11%	38.22%	43.29%	48.32%	53.32%	58.29%	63.22%	68.12%	72.98%	77.80%	82.57%	87.27%	91.89%	96.34%	100.00%
1 000	1.83%	6.54%	12.03%	17.37%	22.62%	27.81%	32.95%	38.05%	43.11%	48.15%	53.15%	58.12%	63.05%	67.96%	72.83%	77.66%	82.44%	87.16%	91.79%	96.27%	100.00%
1 100	1.78%	6.46%	11.93%	17.25%	22.49%	27.67%	32.80%	37.90%	42.97%	48.00%	53.00%	57.97%	62.91%	67.82%	72.70%	77.54%	82.33%	87.06%	91.71%	96.22%	100.00%
1 200	1.74%	6.39%	11.84%	17.15%	22.38%	27.55%	32.68%	37.77%	42.84%	47.87%	52.87%	57.84%	62.79%	67.70%	72.58%	77.43%	82.23%	86.98%	91.64%	96.17%	100.00%
1 300	1.70%	6.33%	11.76%	17.06%	22.28%	27.45%	32.57%	37.66%	42.72%	47.75%	52.75%	57.73%	62.68%	67.59%	72.48%	77.33%	82.14%	86.90%	91.58%	96.12%	100.00%
1 400	1.67%	6.28%	11.69%	16.98%	22.19%	27.35%	32.48%	37.56%	42.62%	47.65%	52.65%	57.63%	62.58%	67.50%	72.39%	77.25%	82.07%	86.83%	91.52%	96.09%	100.00%
1 500	1.64%	6.23%	11.63%	16.91%	22.12%	27.27%	32.39%	37.47%	42.53%	47.56%	52.56%	57.54%	62.49%	67.42%	72.31%	77.18%	82.00%	86.77%	91.47%	96.05%	100.00%
1 600	1.62%	6.18%	11.58%	16.84%	22.05%	27.20%	32.31%	37.39%	42.45%	47.48%	52.48%	57.46%	62.41%	67.34%	72.24%	77.11%	81.94%	86.72%	91.43%	96.02%	100.00%
1 700	1.60%	6.15%	11.53%	16.79%	21.98%	27.13%	32.24%	37.32%	42.37%	47.40%	52.40%	57.38%	62.34%	67.27%	72.17%	77.04%	81.88%	86.67%	91.39%	95.99%	100.00%
1 800	1.58%	6.11%	11.48%	16.73%	21.92%	27.07%	32.18%	37.25%	42.31%	47.33%	52.34%	57.32%	62.27%	67.21%	72.11%	76.99%	81.83%	86.62%	91.35%	95.96%	100.00%
1 900	1.56%	6.08%	11.44%	16.69%	21.87%	27.01%	32.12%	37.19%	42.24%	47.27%	52.27%	57.25%	62.21%	67.15%	72.06%	76.93%	81.78%	86.58%	91.31%	95.94%	100.00%
2 000	1.54%	6.05%	11.40%	16.64%	21.82%	26.96%	32.06%	37.14%	42.19%	47.21%	52.22%	57.20%	62.16%	67.09%	72.00%	76.89%	81.73%	86.54%	91.28%	95.92%	100.00%
2 100	1.52%	6.02%	11.36%	16.60%	21.78%	26.91%	32.01%	37.08%	42.13%	47.16%	52.16%	57.14%	62.10%	67.04%	71.96%	76.84%	81.69%	86.50%	91.25%	95.89%	100.00%
2 200	1.51%	6.00%	11.33%	16.56%	21.73%	26.86%	31.96%	37.03%	42.08%	47.11%	52.11%	57.09%	62.06%	66.99%	71.91%	76.80%	81.65%	86.47%	91.22%	95.87%	100.00%
2 300	1.50%	5.97%	11.30%	16.53%	21.69%	26.82%	31.92%	36.99%	42.04%	47.06%	52.06%	57.05%	62.01%	66.95%	71.87%	76.76%	81.62%	86.44%	91.20%	95.86%	100.00%
2 400	1.48%	5.95%	11.27%	16.49%	21.66%	26.78%	31.88%	36.95%	41.99%	47.02%	52.02%	57.00%	61.97%	66.91%	71.83%	76.72%	81.58%	86.41%	91.17%	95.84%	100.00%
2 500	1.47%	5.93%	11.24%	16.46%	21.62%	26.75%	31.84%	36.91%	41.95%	46.98%	51.98%	56.96%	61.93%	66.87%	71.79%	76.69%	81.55%	86.38%	91.15%	95.82%	100.00%
2 600	1.46%	5.91%	11.22%	16.43%	21.59%	26.71%	31.80%	36.87%	41.91%	46.94%	51.94%	56.93%	61.89%	66.83%	71.76%	76.66%	81.52%	86.35%	91.13%	95.81%	100.00%
2 700	1.45%	5.89%	11.19%	16.40%	21.56%	26.68%	31.77%	36.83%	41.88%	46.90%	51.90%	56.89%	61.85%	66.80%	71.72%	76.62%	81.49%	86.33%	91.11%	95.79%	100.00%
2 800	1.44%	5.87%	11.17%	16.38%	21.53%	26.65%	31.74%	36.80%	41.84%	46.87%	51.87%	56.85%	61.82%	66.77%	71.69%	76.60%	81.47%	86.30%	91.09%	95.78%	100.00%
2 900	1.43%	5.86%	11.15%	16.35%	21.50%	26.62%	31.70%	36.77%	41.81%	46.83%	51.84%	56.82%	61.79%	66.74%	71.66%	76.57%	81.44%	86.28%	91.07%	95.77%	100.00%
3 000	1.42%	5.84%	11.13%	16.33%	21.48%	26.59%	31.68%	36.74%	41.78%	46.80%	51.81%	56.79%	61.76%	66.71%	71.64%	76.54%	81.42%	86.26%	91.05%	95.75%	100.00%
3 100	1.42%	5.83%	11.11%	16.31%	21.45%	26.56%	31.65%	36.71%	41.75%	46.77%	51.78%	56.76%	61.73%	66.68%	71.61%	76.52%	81.40%	86.24%	91.03%	95.74%	100.00%
3 200	1.41%	5.81%	11.09%	16.28%	21.43%	26.54%	31.62%	36.68%	41.72%	46.74%	51.75%	56.73%	61.70%	66.65%	71.58%	76.49%	81.37%	86.22%	91.02%	95.73%	100.00%
3 300	1.40%	5.80%	11.07%	16.26%	21.41%	26.51%	31.60%	36.66%	41.69%	46.72%	51.72%	56.71%	61.68%	66.63%	71.56%	76.47%	81.35%	86.20%	91.00%	95.72%	100.00%

Table IX.2 – Upper limit of confidence intervals according to Pearson-Clopper formula

	Rate																				
3 400	1.39%	5.79%	11.06%	16.24%	21.38%	26.49%	31.57%	36.63%	41.67%	46.69%	51.69%	56.68%	61.65%	66.60%	71.54%	76.45%	81.33%	86.18%	90.99%	95.71%	100.00%
3 500	1.39%	5.77%	11.04%	16.23%	21.36%	26.47%	31.55%	36.61%	41.65%	46.67%	51.67%	56.66%	61.63%	66.58%	71.52%	76.43%	81.31%	86.17%	90.98%	95.70%	100.00%
3 600	1.38%	5.76%	11.03%	16.21%	21.34%	26.45%	31.53%	36.58%	41.62%	46.64%	51.65%	56.63%	61.61%	66.56%	71.49%	76.41%	81.30%	86.15%	90.96%	95.69%	100.00%
3 700	1.38%	5.75%	11.01%	16.19%	21.33%	26.43%	31.51%	36.56%	41.60%	46.62%	51.62%	56.61%	61.58%	66.54%	71.47%	76.39%	81.28%	86.14%	90.95%	95.68%	100.00%
3 800	1.37%	5.74%	11.00%	16.18%	21.31%	26.41%	31.49%	36.54%	41.58%	46.60%	51.60%	56.59%	61.56%	66.52%	71.45%	76.37%	81.26%	86.12%	90.94%	95.67%	100.00%
3 900	1.36%	5.73%	10.98%	16.16%	21.29%	26.39%	31.47%	36.52%	41.56%	46.58%	51.58%	56.57%	61.54%	66.50%	71.44%	76.35%	81.25%	86.11%	90.92%	95.66%	100.00%
4 000	1.36%	5.72%	10.97%	16.14%	21.27%	26.37%	31.45%	36.50%	41.54%	46.56%	51.56%	56.55%	61.52%	66.48%	71.42%	76.34%	81.23%	86.09%	90.91%	95.66%	100.00%
4 100	1.35%	5.71%	10.96%	16.13%	21.26%	26.36%	31.43%	36.48%	41.52%	46.54%	51.54%	56.53%	61.50%	66.46%	71.40%	76.32%	81.22%	86.08%	90.90%	95.65%	100.00%
4 200	1.35%	5.70%	10.95%	16.12%	21.24%	26.34%	31.41%	36.46%	41.50%	46.52%	51.52%	56.51%	61.49%	66.44%	71.38%	76.30%	81.20%	86.07%	90.89%	95.64%	100.00%
4 300	1.34%	5.69%	10.94%	16.10%	21.23%	26.32%	31.39%	36.45%	41.48%	46.50%	51.51%	56.49%	61.47%	66.43%	71.37%	76.29%	81.19%	86.06%	90.88%	95.63%	100.00%
4 400	1.34%	5.69%	10.92%	16.09%	21.21%	26.31%	31.38%	36.43%	41.47%	46.48%	51.49%	56.48%	61.45%	66.41%	71.35%	76.27%	81.17%	86.04%	90.87%	95.63%	100.00%
4 500	1.34%	5.68%	10.91%	16.08%	21.20%	26.29%	31.36%	36.41%	41.45%	46.47%	51.47%	56.46%	61.44%	66.39%	71.34%	76.26%	81.16%	86.03%	90.86%	95.62%	100.00%
4 600	1.33%	5.67%	10.90%	16.06%	21.19%	26.28%	31.35%	36.40%	41.43%	46.45%	51.46%	56.45%	61.42%	66.38%	71.32%	76.25%	81.15%	86.02%	90.85%	95.61%	100.00%
4 700	1.33%	5.66%	10.89%	16.05%	21.17%	26.26%	31.33%	36.38%	41.42%	46.44%	51.44%	56.43%	61.40%	66.36%	71.31%	76.23%	81.14%	86.01%	90.84%	95.61%	100.00%
4 800	1.32%	5.65%	10.88%	16.04%	21.16%	26.25%	31.32%	36.37%	41.40%	46.42%	51.42%	56.41%	61.39%	66.35%	71.29%	76.22%	81.12%	86.00%	90.83%	95.60%	100.00%
4 900	1.32%	5.65%	10.87%	16.03%	21.15%	26.24%	31.30%	36.35%	41.39%	46.41%	51.41%	56.40%	61.38%	66.34%	71.28%	76.21%	81.11%	85.99%	90.83%	95.59%	100.00%
5 000	1.32%	5.64%	10.87%	16.02%	21.14%	26.22%	31.29%	36.34%	41.37%	46.39%	51.40%	56.39%	61.36%	66.32%	71.27%	76.20%	81.10%	85.98%	90.82%	95.59%	100.00%

IX.4.6 Span of confidence intervals according to Pearson-Clopper formula

Table IX.3 contains values which specify the difference ("span") between the upper and the lower limit of the confidence interval. The span depends on the number of samples and the according rate value. In Figures IX.4.1 to IX.4.3 this information can be found as the vertical distance between the red and the blue lines.

Table IX.3 – Span of confidence intervals according to Pearson-Clopper formula

	Rate																				
NrMeas	1%	5%	10%	15%	20%	25%	30%	35%	40%	45%	50%	55%	60%	65%	70%	75%	80%	85%	90%	95%	100%
100	5.42%	9.64%	12.72%	14.89%	16.52%	17.78%	18.74%	19.46%	19.95%	20.30%	20.38%	20.28%	19.95%	19.46%	18.74%	17.90%	16.67%	15.07%	12.96%	10.00%	4.85%
200	3.44%	6.58%	8.81%	10.36%	11.53%	12.44%	13.13%	13.64%	13.99%	14.22%	14.29%	14.21%	13.99%	13.64%	13.13%	12.48%	11.59%	10.43%	8.90%	6.71%	2.47%
300	2.69%	5.29%	7.12%	8.40%	9.36%	10.10%	10.66%	11.08%	11.39%	11.54%	11.61%	11.55%	11.37%	11.08%	10.68%	10.12%	9.39%	8.43%	7.17%	5.36%	1.66%
400	2.27%	4.54%	6.13%	7.24%	8.07%	8.71%	9.21%	9.57%	9.82%	9.98%	10.02%	9.97%	9.82%	9.57%	9.21%	8.73%	8.09%	7.26%	6.16%	4.59%	1.25%
500	1.99%	4.03%	5.46%	6.45%	7.20%	7.77%	8.21%	8.54%	8.77%	8.90%	8.95%	8.90%	8.77%	8.54%	8.22%	7.79%	7.21%	6.47%	5.48%	4.07%	1.00%
600	1.80%	3.66%	4.97%	5.87%	6.56%	7.08%	7.49%	7.78%	8.00%	8.11%	8.15%	8.11%	7.99%	7.78%	7.49%	7.09%	6.57%	5.89%	4.98%	3.69%	0.83%
700	1.65%	3.38%	4.59%	5.43%	6.06%	6.55%	6.92%	7.20%	7.39%	7.50%	7.54%	7.50%	7.39%	7.20%	6.93%	6.56%	6.07%	5.44%	4.60%	3.40%	0.71%

Table IX.3 – Span of confidence intervals according to Pearson-Clopper formula

	Rate																				
800	1.53%	3.15%	4.28%	5.07%	5.66%	6.12%	6.47%	6.73%	6.90%	7.01%	7.05%	7.01%	6.90%	6.73%	6.47%	6.12%	5.67%	5.08%	4.29%	3.17%	0.63%
900	1.43%	2.96%	4.03%	4.77%	5.33%	5.76%	6.09%	6.34%	6.51%	6.60%	6.64%	6.60%	6.50%	6.34%	6.10%	5.77%	5.34%	4.78%	4.04%	2.98%	0.56%
1 000	1.35%	2.81%	3.82%	4.52%	5.05%	5.46%	5.77%	6.01%	6.17%	6.26%	6.29%	6.26%	6.17%	6.01%	5.78%	5.47%	5.06%	4.53%	3.83%	2.82%	0.50%
1 100	1.28%	2.67%	3.64%	4.31%	4.81%	5.20%	5.50%	5.72%	5.88%	5.97%	6.00%	5.97%	5.87%	5.72%	5.50%	5.21%	4.82%	4.31%	3.64%	2.68%	0.46%
1 200	1.22%	2.55%	3.48%	4.12%	4.61%	4.98%	5.26%	5.48%	5.62%	5.71%	5.74%	5.71%	5.62%	5.48%	5.27%	4.98%	4.61%	4.13%	3.48%	2.56%	0.42%
1 300	1.17%	2.45%	3.34%	3.96%	4.42%	4.78%	5.05%	5.26%	5.40%	5.48%	5.51%	5.48%	5.40%	5.26%	5.06%	4.78%	4.43%	3.96%	3.34%	2.46%	0.39%
1 400	1.12%	2.36%	3.21%	3.81%	4.26%	4.60%	4.87%	5.07%	5.20%	5.28%	5.31%	5.28%	5.20%	5.06%	4.87%	4.61%	4.26%	3.81%	3.22%	2.36%	0.36%
1 500	1.08%	2.27%	3.10%	3.68%	4.11%	4.45%	4.70%	4.89%	5.02%	5.10%	5.12%	5.10%	5.02%	4.89%	4.70%	4.45%	4.12%	3.68%	3.11%	2.28%	0.33%
1 600	1.05%	2.20%	3.00%	3.56%	3.98%	4.30%	4.55%	4.73%	4.86%	4.93%	4.96%	4.93%	4.86%	4.73%	4.55%	4.31%	3.98%	3.56%	3.01%	2.21%	0.31%
1 700	1.01%	2.13%	2.91%	3.45%	3.86%	4.17%	4.41%	4.59%	4.71%	4.79%	4.81%	4.79%	4.71%	4.59%	4.41%	4.17%	3.86%	3.46%	2.91%	2.14%	0.29%
1 800	0.98%	2.07%	2.83%	3.35%	3.75%	4.05%	4.29%	4.46%	4.58%	4.65%	4.67%	4.65%	4.58%	4.46%	4.29%	4.06%	3.75%	3.36%	2.83%	2.08%	0.28%
1 900	0.95%	2.01%	2.75%	3.26%	3.65%	3.94%	4.17%	4.34%	4.46%	4.52%	4.55%	4.52%	4.46%	4.34%	4.17%	3.95%	3.65%	3.27%	2.75%	2.02%	0.26%
2 000	0.93%	1.96%	2.68%	3.18%	3.55%	3.84%	4.06%	4.23%	4.34%	4.41%	4.43%	4.41%	4.34%	4.23%	4.07%	3.84%	3.56%	3.18%	2.68%	1.97%	0.25%
2 100	0.90%	1.91%	2.61%	3.10%	3.47%	3.75%	3.97%	4.13%	4.24%	4.30%	4.32%	4.30%	4.24%	4.13%	3.97%	3.75%	3.47%	3.10%	2.62%	1.92%	0.24%
2 200	0.88%	1.87%	2.55%	3.03%	3.39%	3.66%	3.87%	4.03%	4.14%	4.20%	4.22%	4.20%	4.14%	4.03%	3.87%	3.66%	3.39%	3.03%	2.56%	1.87%	0.23%
2 300	0.86%	1.83%	2.50%	2.96%	3.31%	3.58%	3.79%	3.94%	4.05%	4.11%	4.13%	4.11%	4.05%	3.94%	3.79%	3.58%	3.31%	2.96%	2.50%	1.83%	0.22%
2 400	0.84%	1.79%	2.44%	2.90%	3.24%	3.50%	3.71%	3.86%	3.96%	4.02%	4.04%	4.02%	3.96%	3.86%	3.71%	3.51%	3.24%	2.90%	2.44%	1.79%	0.21%
2 500	0.82%	1.75%	2.39%	2.84%	3.17%	3.43%	3.63%	3.78%	3.88%	3.94%	3.96%	3.94%	3.88%	3.78%	3.63%	3.43%	3.18%	2.84%	2.39%	1.75%	0.20%
2 600	0.81%	1.71%	2.34%	2.78%	3.11%	3.37%	3.56%	3.70%	3.80%	3.86%	3.88%	3.86%	3.80%	3.70%	3.56%	3.37%	3.11%	2.78%	2.35%	1.72%	0.19%
2 700	0.79%	1.68%	2.30%	2.73%	3.05%	3.30%	3.49%	3.63%	3.73%	3.79%	3.81%	3.79%	3.73%	3.63%	3.49%	3.30%	3.05%	2.73%	2.30%	1.68%	0.19%
2 800	0.78%	1.65%	2.26%	2.68%	3.00%	3.24%	3.43%	3.57%	3.66%	3.72%	3.74%	3.72%	3.66%	3.57%	3.43%	3.24%	3.00%	2.68%	2.26%	1.65%	0.18%
2 900	0.76%	1.62%	2.22%	2.63%	2.95%	3.19%	3.37%	3.51%	3.60%	3.65%	3.67%	3.65%	3.60%	3.50%	3.37%	3.19%	2.95%	2.63%	2.22%	1.62%	0.17%
3 000	0.75%	1.59%	2.18%	2.59%	2.90%	3.13%	3.31%	3.45%	3.54%	3.59%	3.61%	3.59%	3.54%	3.45%	3.31%	3.13%	2.90%	2.59%	2.18%	1.60%	0.17%
3 100	0.74%	1.57%	2.14%	2.55%	2.85%	3.08%	3.26%	3.39%	3.48%	3.53%	3.55%	3.53%	3.48%	3.39%	3.26%	3.08%	2.85%	2.55%	2.15%	1.57%	0.16%
3 200	0.72%	1.54%	2.11%	2.51%	2.80%	3.03%	3.21%	3.34%	3.42%	3.48%	3.49%	3.48%	3.42%	3.34%	3.21%	3.03%	2.80%	2.51%	2.11%	1.54%	0.16%
3 300	0.71%	1.52%	2.08%	2.47%	2.76%	2.98%	3.16%	3.28%	3.37%	3.42%	3.44%	3.42%	3.37%	3.28%	3.16%	2.98%	2.76%	2.47%	2.08%	1.52%	0.15%
3 400	0.70%	1.50%	2.05%	2.43%	2.72%	2.94%	3.11%	3.23%	3.32%	3.37%	3.39%	3.37%	3.32%	3.23%	3.11%	2.94%	2.72%	2.43%	2.05%	1.50%	0.15%
3 500	0.69%	1.47%	2.02%	2.39%	2.68%	2.90%	3.06%	3.19%	3.27%	3.32%	3.34%	3.32%	3.27%	3.19%	3.06%	2.90%	2.68%	2.40%	2.02%	1.48%	0.14%
3 600	0.68%	1.45%	1.99%	2.36%	2.64%	2.86%	3.02%	3.14%	3.23%	3.28%	3.29%	3.28%	3.23%	3.14%	3.02%	2.86%	2.64%	2.36%	1.99%	1.45%	0.14%
3 700	0.67%	1.43%	1.96%	2.33%	2.60%	2.82%	2.98%	3.10%	3.18%	3.23%	3.25%	3.23%	3.18%	3.10%	2.98%	2.82%	2.60%	2.33%	1.96%	1.43%	0.14%
3 800	0.66%	1.41%	1.93%	2.30%	2.57%	2.78%	2.94%	3.06%	3.14%	3.19%	3.20%	3.19%	3.14%	3.06%	2.94%	2.78%	2.57%	2.30%	1.94%	1.41%	0.13%
3 900	0.65%	1.39%	1.91%	2.27%	2.54%	2.74%	2.90%	3.02%	3.10%	3.15%	3.16%	3.15%	3.10%	3.02%	2.90%	2.74%	2.54%	2.27%	1.91%	1.40%	0.13%
4 000	0.64%	1.38%	1.88%	2.24%	2.50%	2.71%	2.86%	2.98%	3.06%	3.11%	3.12%	3.11%	3.06%	2.98%	2.86%	2.71%	2.50%	2.24%	1.89%	1.38%	0.13%
4 100	0.64%	1.36%	1.86%	2.21%	2.47%	2.67%	2.83%	2.94%	3.02%	3.07%	3.08%	3.07%	3.02%	2.94%	2.83%	2.68%	2.47%	2.21%	1.86%	1.36%	0.12%
4 200	0.63%	1.34%	1.84%	2.18%	2.44%	2.64%	2.79%	2.91%	2.99%	3.03%	3.05%	3.03%	2.99%	2.91%	2.80%	2.64%	2.44%	2.18%	1.84%	1.34%	0.12%

Table IX.3 – Span of confidence intervals according to Pearson-Clopper formula

	Rate																				
4 300	0.62%	1.33%	1.82%	2.16%	2.41%	2.61%	2.76%	2.87%	2.95%	3.00%	3.01%	3.00%	2.95%	2.87%	2.76%	2.61%	2.41%	2.16%	1.82%	1.33%	0.12%
4 400	0.61%	1.31%	1.80%	2.13%	2.39%	2.58%	2.73%	2.84%	2.92%	2.96%	2.98%	2.96%	2.92%	2.84%	2.73%	2.58%	2.39%	2.13%	1.80%	1.31%	0.11%
4 500	0.61%	1.30%	1.78%	2.11%	2.36%	2.55%	2.70%	2.81%	2.88%	2.93%	2.94%	2.93%	2.88%	2.81%	2.70%	2.55%	2.36%	2.11%	1.78%	1.30%	0.11%
4 600	0.60%	1.28%	1.76%	2.09%	2.33%	2.52%	2.67%	2.78%	2.85%	2.90%	2.91%	2.90%	2.85%	2.78%	2.67%	2.52%	2.33%	2.09%	1.76%	1.28%	0.11%
4 700	0.59%	1.27%	1.74%	2.06%	2.31%	2.50%	2.64%	2.75%	2.82%	2.87%	2.88%	2.87%	2.82%	2.75%	2.64%	2.50%	2.31%	2.06%	1.74%	1.27%	0.11%
4 800	0.59%	1.25%	1.72%	2.04%	2.28%	2.47%	2.61%	2.72%	2.79%	2.83%	2.85%	2.84%	2.79%	2.72%	2.61%	2.47%	2.28%	2.04%	1.72%	1.26%	0.10%
4 900	0.58%	1.24%	1.70%	2.02%	2.26%	2.44%	2.59%	2.69%	2.76%	2.81%	2.82%	2.81%	2.76%	2.69%	2.59%	2.45%	2.26%	2.02%	1.70%	1.24%	0.10%
5 000	0.57%	1.23%	1.68%	2.00%	2.24%	2.42%	2.56%	2.66%	2.74%	2.78%	2.79%	2.78%	2.74%	2.66%	2.56%	2.42%	2.24%	2.00%	1.68%	1.23%	0.10%

IX.5 Different sample sizes

The following examples show the effect of different sample sizes in a measurement campaign. It is also based on the Pearson-Clopper formulas for the calculation of confidence intervals. Therefore, the examples are valid in a generic way and even for small sample sizes. For higher sample numbers, the calculation of confidence intervals based on the approximation of a normal distribution can be applied.

Figures IX.5.1 to IX.5.3 show the width of the confidence interval for different sample sizes in the following ranges:

- between 100 and 1 100 samples;
- between 1 100 and 2 100 samples; and
- between 1 000 and 11 000 samples.

There is correspondence between curves in Figure IX.5.1 to IX.5.3 and information provided in Tables IX.4.1 to IX.4.3 (number of measurements is constant, estimated rate varies).

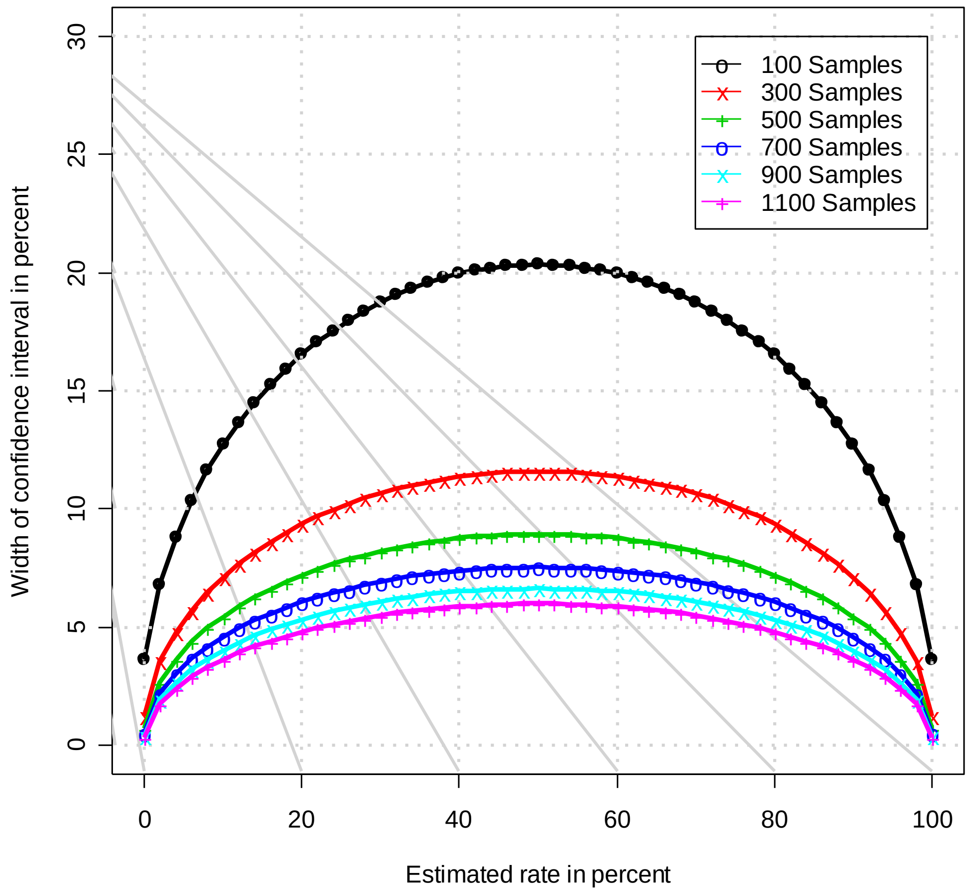


Figure IX.5.1 – Width of confidence interval for different sample sizes

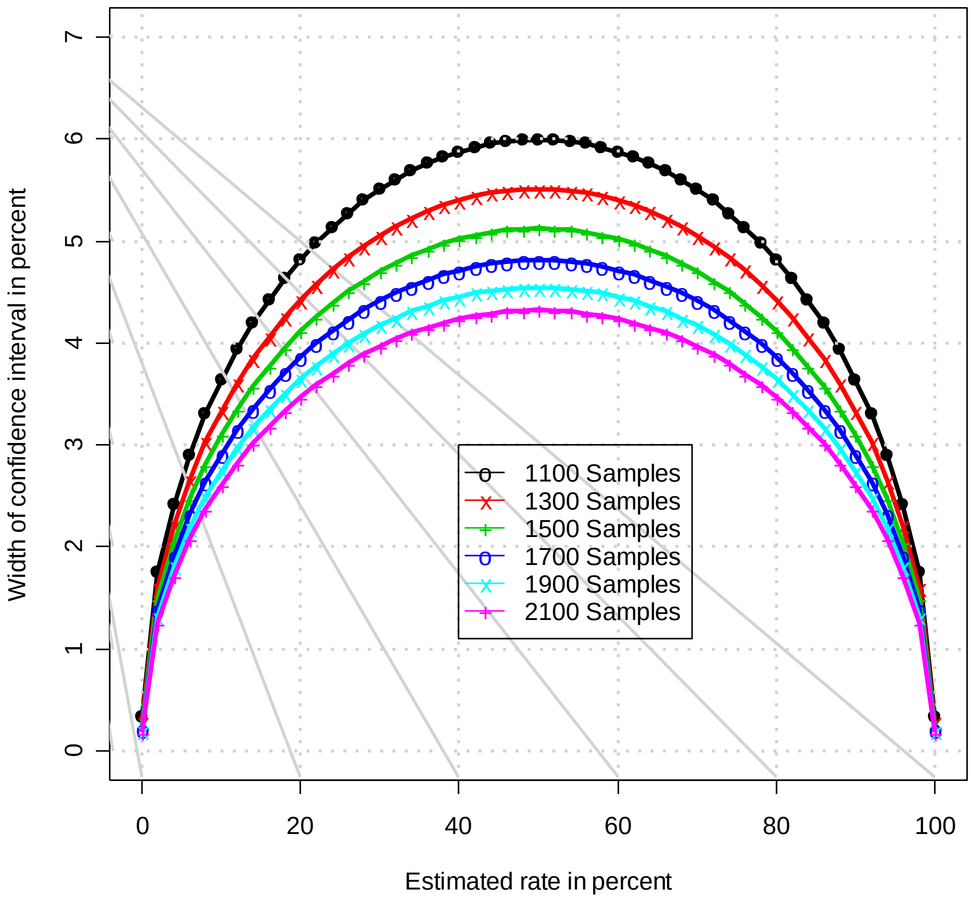


Figure IX.5.2 – Width of confidence interval for different sample sizes

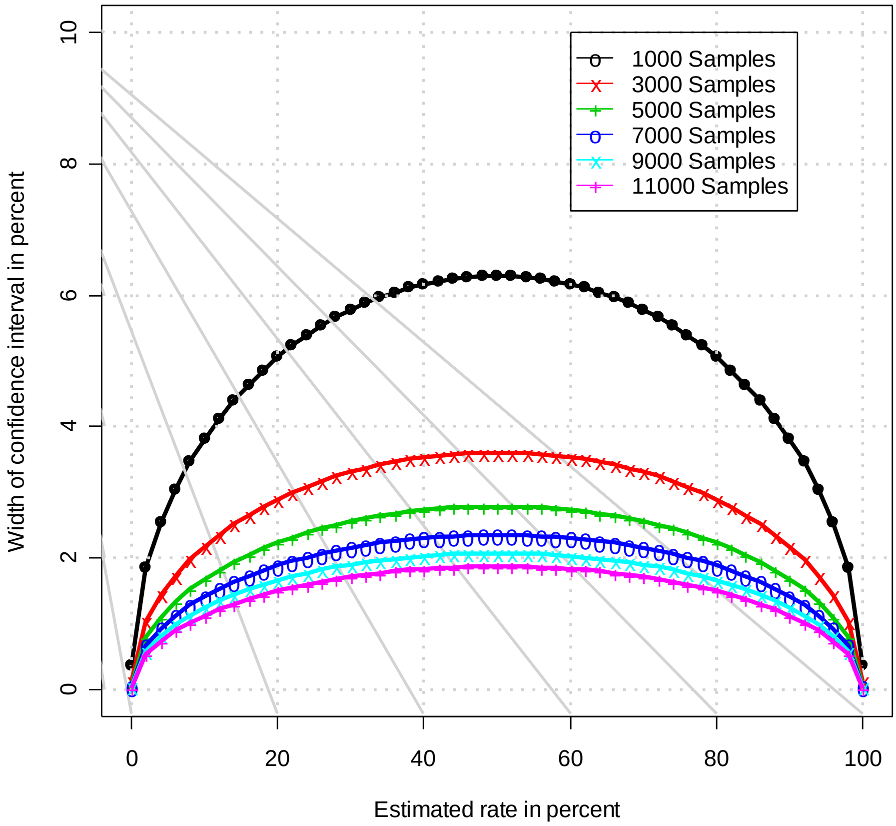


Figure IX.5.3 – Width of confidence interval for different sample sizes

IX.6 Calculation methods

This clause depicts some examples of how to calculate statistical values out of measurement data.

IX.6.1 Calculation of quantiles

In this clause the different basic steps to calculate quantile values related to measurement samples, shown in Figure IX.6.1, are described.

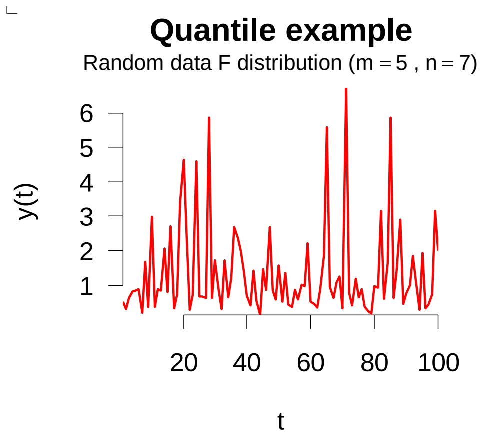


Figure IX.6.1 – Example of measured data as a time series

Assuming that measurement data corresponding to the Figure IX.6.1 have been collected, the following steps can be executed:

- Determine the number N of available measurements.

- Sorting of data: The samples are sorted in an ascending order.
- Define the p-quantile value that should be retrieved. In this example, the 95% quantile (Q95) is requested, so p = 95% = 0.95.
- Start counting the sorted samples, until reaching the p-percentage of all available samples. In this example, this means 95 per cent of the samples have to be counted.
- The sample where the corresponding percentage is reached is taken. The appropriate ordinate value represents the searched p-, in this case the 95% quantile.

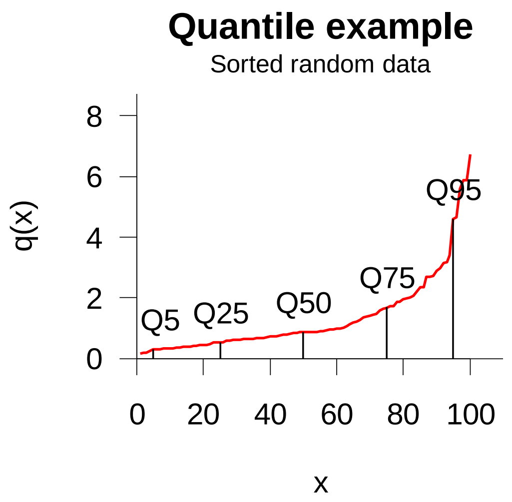


Figure IX.6.2 – Determination of quantiles on sorted data

The different steps are visualized in Figure IX.6.2. Additional examples for other p-quantiles are:

p percentage	5%	25%	50%	75%	95%
p-quantile	0.2959737	0.5370118	0.8579087	1.6867595	4.5992459

If for example the 95 per cent value is not covered by a sample, an interpolation between the left-hand and the right-hand neighbour may be appropriate. This interpolation may have different grades, e.g., linear or quadratic interpolation.

Another possibility to determine quantile values is given by analysis of the cumulative distribution function (CDF). The steps to create a CDF out of measurement results are generally the same as described above.

IX.7 Reporting of results

This clause describes which pieces of information should be provided when generating a test report. The categories of different data types are related to the definitions in clause 3.

IX.7.1 Methods to use

The variables *x*, *y* and *z* in the Table IX.4 must be accordingly replaced by the estimated data.

When quantile values are used, it should be kept in mind that the computation of quantiles separates a low percentage of outlier data from the remaining data. This means:

- If lower values represent a better outcome from the user's perspective, a small percentage containing the highest values could be separated by calculating a 95%-quantile or a 90%-quantile. This is the case for example for duration values.

- If higher values represent a better outcome from the user's perspective, a small percentage containing the lowest values could be separated by calculating a 5%-quantile or a 10%-quantile. This is the case for example for throughput values.
- Related to content quality, the appropriate quantile computation orientates itself on the scale of the determined test results. In practice, some algorithms define a value of 0 on a scale from 0 to 5 as the best quality whereas others define the value of 5 as the highest possible quality. Table IX.4 provides some hints on how to use the quantile computation in these cases.

Table IX.4 – Proposed methods for preparation of result statements on quantile computation

Category of data	Type of information	Method to use	Reporting statement	Additional information	Related clauses
Binary values (Success rates, error rates, ...)	Estimated rate plus confidence interval	Pearson-Clopper	$x\%^{+y_1\%}_{-y_2\%}$	Always valid, borders of confidence interval are asymmetric (except for x = 50)	11.2.7.2.1
		Gaussian approximation	$x\% \pm y\%$	Applicable if $n \cdot p \cdot q \geq 9$, symmetric borders of confidence interval	11.2.7.2.2
Duration values (End-to-end delay, establishment delay, ...)	Mean delay plus standard deviation	Empirical mean plus empirical standard deviation	$x\ s \pm y\ s\ (N = z)$	Always valid N: number of samples taken into account	11.2.4and 11.2.5
	α -Quantile plus number of samples	Quantile computation	$q_\alpha = x\ s\ (N = z)$	N: number of samples taken into account α : Desired quantile level, mostly $\alpha = 95\%$ or $\alpha = 90\%$	11.2.4and 11.2.5

Table IX.4 – Proposed methods for preparation of result statements on quantile computation

Category of data	Type of information	Method to use	Reporting statement	Additional information	Related clauses
Throughput values (Data rates)	Mean data rate plus standard deviation	Empirical mean plus empirical standard deviation	$x\ kbit/s \pm y\ kbit/s\ (N = z)$	Always valid N: number of samples taken into account	11.2.4and 11.2.5
	α -Quantile plus number of samples	Quantile computation	$q_{\alpha} = x\ kbit/s\ (N = z)$	N: number of samples taken into account α : Desired quantile level, mostly $\alpha = 5\%$ or $\alpha = 10\%$	11.2.4and 11.2.5
Content quality values (Audio quality, video quality)	Mean score plus standard deviation	Empirical mean plus empirical standard deviation	$x\ MOS \pm y\ MOS\ (N = z)$	Always valid N: number of samples taken into account	11.2.4and 11.2.5
	α -Quantile plus number of samples	Quantile computation	$q_{\alpha} = x\ MOS\ (N = z)$	N: number of samples taken into account α : Desired quantile level, mostly $\alpha = 95\%$ or $\alpha = 90\%$ if lower values represent better quality, $\alpha = 5\%$ or $\alpha = 10\%$ if higher values represent better quality	11.2.4and 11.2.5

IX.7.2 Number of significant decimals

When representing final results, the number of reported significant decimals should be orientated on the precision of the evaluation method used (e.g., calculation of standard deviation, confidence interval, ...).

IX.7.3 Rounding of the end results

During the execution of consecutive calculation steps, no rounding of decimals should be applied. Only the final results may be rounded. At least three significant decimals should still remain after applying the rounding of decimals whenever possible.

Appendix X

The concept of QoE reporting

(This appendix does not form an integral part of this Recommendation.)

A QoE reporting mechanism has been standardized in 3GPP for packet-switched streaming service (PSS) [b-ETSI TS 126 234], multimedia broadcast/multicast service (MBMS) [b-ETSI TS 126 346] and multimedia telephony service for IMS (MTSI) [b-ETSI TS 126 114]. The standards make it possible for the operator to activate QoE feedback report from the mobiles whenever PSS, MBMS or MTSI services are used. This makes it possible to closely follow relevant parameters from an end-user service quality perspective.

The specific implementation differs slightly between the three cases, but has a common conceptual structure:

- The operator can activate QoE reporting for a specific terminal, or for a statistical subset of the terminals. For instance, it might be enough to monitor only 10 per cent of the terminals to get a statistically good result, and thus saving some cell capacity due to lower amount of QoE reporting.
- The operator can specify which parameters the terminal should report, and how often each parameter should be calculated. For instance, the terminal can be ordered to measure the amount of packet loss and rebuffering for a streaming service, or the frame loss and speech codec usage for an MTSI speech service. A typical parameter calculation length could be in the order of 8 to 12 seconds, which corresponds to commonly used sequence lengths for subjective tests.
- The operator can specify how often the measurements calculated by the terminal should be reported back to the system, for instance every fifth minute. This allows a more efficient feedback transmission, as several measurements calculated and buffered by the terminal are packed together before sending. Except for the periodic reports, a final report is always sent after the service has ended with the remaining buffered measurements.

NOTE – While the QoE feedback reporting feature can be very useful for the operator, it is currently specified as "optional" in the 3GPP standards. This means that some terminals have implemented the feature, while others have not. Some terminals might have implemented the feature in the terminal platform, but only include it in specific terminal models if the operator asks for it.

Appendix XI

Examples of network based QoS measurements

(This appendix does not form an integral part of this Recommendation.)

XI.1 Accessibility and retainability parameters

Figure XI.1 shows a subset of the protocol messages sent for a mobile-terminated call. The full sequence is described in clause 7 under "Telephony service non-accessibility".

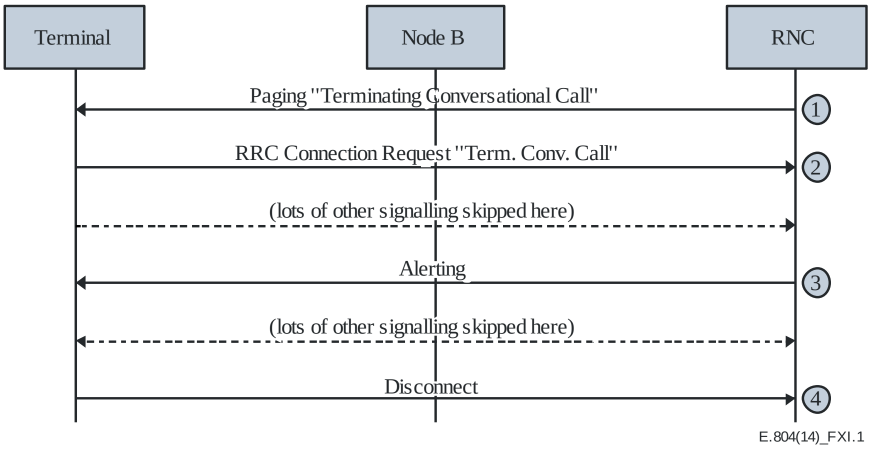


Figure XI.1 – Call set-up and disconnect sequence for a mobile-terminated speech call

This example measures the telephony service non-accessibility for a circuit-switched call. The measurements are done on the Iub interface close to the RNC. The starting trigger for the measurement can be either point 1 or point 2. If point 1 is chosen the measurements may include some terminals which are temporarily out of coverage, but it will include all call attempts. If point 2 is chosen, the measurements will only include terminals which have coverage, but might miss the call set-up for some terminals which have basic signal strength coverage (i.e., they seem to have radio network availability) but not enough to be able to send the RRC connection request.

The end trigger for a successful call set-up is point 3, where the Alerting message is sent to the terminal. If the end trigger is not reached the call attempt is counted as unsuccessful. This can for instance indicate that the terminal has sufficient signal strength to be able to use the signalling channel, but not enough to be able to set up the dedicated speech radio bearer.

The time to make a successful call set-up (telephony set-up time) can else be measured using the same example as above, where the time difference between the end trigger (point 3) and the start trigger (point 1 or 2) is measured.

The retainability (telephony cut-off call ratio) can be measured by observing the messages at point 3 and 4. A call is counted as being cut-off if the start trigger (point 3) is seen and the call is ended when the network (call control entity – the MSC) sends a DISCONNECT message to the terminal with a cause different to normal call ending. See [ETSI TS 124 008] for details about signalling for "Call Clearing" in an UMTS network.

XI.2 Media quality parameters

In this example the media quality is estimated for the 3GPP packet-switched streaming service (PSS) [b-ETSI TS 126 234], where the parameters are measured in the terminal (PCO) and sent to the network (POR) using the 3GPP standardized QoE reporting. The steps involved in measuring the media quality are:

- Initial service access initiated from the terminal (see Figure XI.2, point 1).
- The media server sets up QoE reporting from terminals using the SDP and the RTSP protocols (point 2). The QoE set-up is performed when the user wants to see a video or hear audio where the PSS service is used. The terminal acknowledges the QoE configuration (point 3).
- The terminal measures the QoE parameters and sends periodic QoE reports to the media server (POR) during the PSS session (point 4). The QoE reports are sent using the RTSP protocol.
- The QoE reports are used together with additional information from the server to calculate the perceived media quality. The media quality MOS is estimated using an objective parametric media quality model.

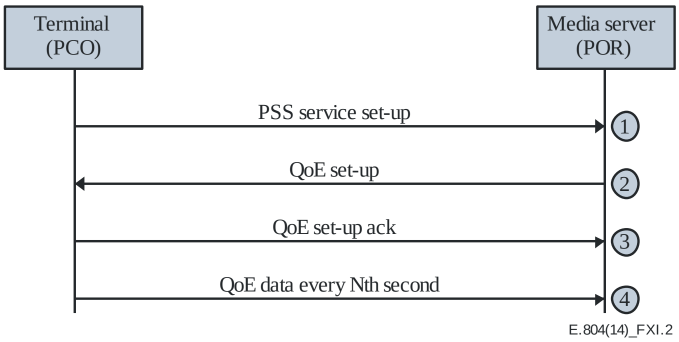


Figure XI.2 – Media quality measurement for the 3GPP PSS service

XI.3 Response time parameters

In this example the HTTP IP-service set-up time QoS parameter is measured at the SGSN node interface for a HTTP download of a single file from a server. The steps involved in the measurement are:

- The HTTP session is set up between the terminal (client) and the HTTP server.
- At a standardized interface at the SGSN the data packets for the HTTP session are inspected and the HTTP and TCP protocol headers are analysed.
- The time when the "HTTP get" request from the HTTP client to the HTTP server leaves the SGSN on the Gn interface (the PCO and the POR) is measured. This is represented with the upper circle (point 1) in the sequence diagram for the SGSN node in Figure XI.3.
- When the first data packet containing content is received at the SGSN interface the time of arrival is recorded. This is represented by the lower circle (point 2).
- The response time is calculated by calculating the time difference between receiving the "HTTP get" command and receiving the first data packet containing content.

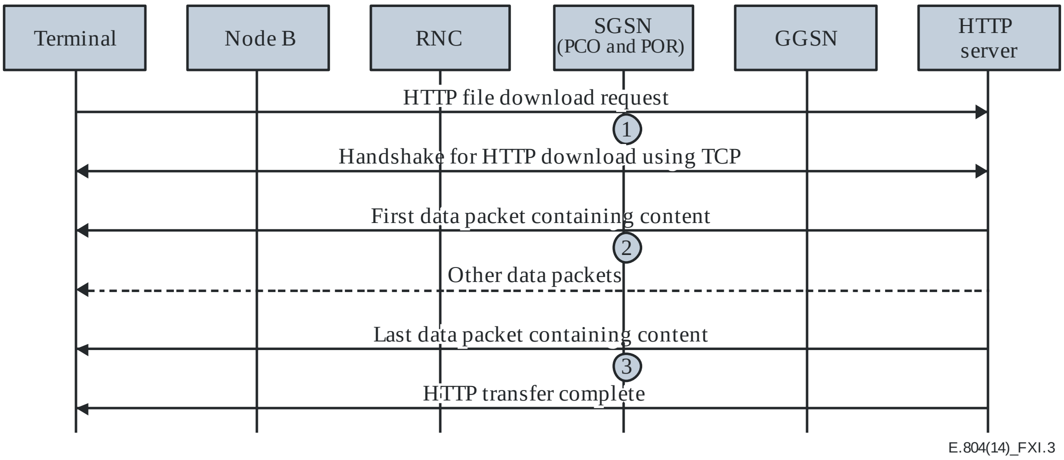


Figure XI.3 – Measuring HTTP download mean set-up time at standardized interface at the SGSN node

Note that the measured time above does not include the time for the transmission of the "HTTP get" command from the client to the SGSN and the time for the first data packet to be transferred from the SGSN to the client. These times can, for example, be estimated as follows:

- When the first TCP data packet is sent from the SGSN towards the terminal (point 2), note the time and TCP sequence number for this and the following TCP packets.
- When a TCP ACK is received from the terminal (not shown in figure), note the time of arrival of this packet.
- Calculate the time between the reception of the TCP ACK packet and the time of sending of the last TCP packet which was acknowledged in the TCP ACK.

By adding the two times (SGSN to server and SGSN to terminal) an estimate of the total response time can be calculated.

Note that if packets are lost the calculated ACK time can be larger than the shortest possible time. This can, for instance, be handled by making several ACK time measurements and using the shortest or the average time.

XI.4 Data rate parameters

In this example the HTTP mean data rate QoS parameter is measured in the SGSN node interface for a HTTP download of a single file from an HTTP server. The steps involved in the measurement are:

- The HTTP session is set up between the terminal (client) and the HTTP server.
- At a standardized interface at the SGSN, the data packets for the HTTP session are inspected and the HTTP and TCP protocol headers are analysed.
- When the first data packet containing content is received at the SGSN interface (the PCO and POR) the time of arrival is recorded. This is represented by point 2 in the sequence diagram for the SGSN node in Figure XI.3.
- For the whole file transfer from the server to the client the total amount of user data transferred is measured. When the file has been completely transferred the information about the total file size is known at the SGSN interface.
- When the last data packet containing content is received at the SGSN interface the time of arrival for that packet is recorded (point 3).
- The mean data rate QoS parameter can now be calculated using the amount of user data transferred, and the time between arrival of the first and last data packet containing content. Note that throughput can be calculated taking or not taking TCP retransmissions into account.

Throughput can also be calculated taking only the payload into account or including both payload and transport protocol headers.

Appendix XII

3GPP SA5 "UE management"

(This appendix does not form an integral part of this Recommendation.)

The 3GPP SA5 standardization group is discussing opening a new work item for "UE management". One of the proposed purposes of this work item is:

"Defining a candidate set of UE measurements to enhance multi-technology radio network planning as well as self auto-optimization where possible, including the consideration of the impact on UE performance and air interface. The UE measurements do not imply to be the end-to-end, e.g., end-to-end service related measurements."

This potential 3GPP SA5 work item might be related to the QoE reporting concept described in Appendix XI.

Bibliography

[b-ITU-T E.419] Recommendation ITU-T E.419 (2006), *Business oriented Key Performance Indicators for management of networks and services.*

[b-ITU-T E.801] Recommendation ITU-T E.801 (1996), *Framework for Service Quality Agreement.*

[b-ITU-T G.1010] Recommendation ITU-T G.1010 (2001), *End-user multimedia QoS categories.*

[b-ITU-T X.290] Recommendation ITU-T X.290 (1995), *OSI conformance testing methodology and framework for protocol Recommendations for ITU-T applications – General concepts.*

[b-ETSI EG 201 769] ETSI EG 201 769 V1.1.2 (2000), *Speech Processing, Transmission and Quality Aspects (STQ); QoS parameter definitions and measurements; Parameters for voice telephony service required under the ONP Voice Telephony Directive 98/10/EC.*

[b-ETSI EG 202 057-1] ETSI EG 202 057-1 V1.3.1 (2008), *Speech and multimedia Transmission Quality (STQ); User related QoS parameter definitions and measurements; Part 1: General.*

[b-ETSI EG 202 057-2] ETSI EG 202 057-2 V1.3.2 (2011), *Speech and multimedia Transmission Quality (STQ); User related QoS parameter definitions and measurements; Part 2: Voice telephony, Group 3 fax, modem data services and SMS.*

[b-ETSI EG 202 057-3] ETSI EG 202 057-3 V1.1.1 (2005), *Speech and multimedia Transmission Quality (STQ); User related QoS parameter definitions and measurements;* Part 3*: QoS parameters specific to Public Land Mobile Networks (PLMN).*

[b-ETSI EG 202 057-4] ETSI EG 202 057-1 V1.2.1 (2008), *Speech and multimedia Transmission Quality (STQ); User related QoS parameter definitions and measurements; Part 1:* Internet access.

[b-ETSI ETS 300 607-1] ETSI ETS 300 607-1 ed. 15(2000), *Digital cellular telecommunications system (Phase 2) ; Mobile Station (MS) conformance specification; Part 1: Conformance specification; (GSM 11.10-1 Version 4.28.1).*

[b-ETSI EN 300 911] ETSI EN 300 911 V8.4.1 (2000), *Digital cellular telecommunications system (Phase 2+); Radio subsystem link control (GSM 05.08 version 8.4.1Release 1999).*

[b-ETSI EN 302 304] ETSI EN 302 304 V1.1.1(2004), *Digital Video Broadcasting (DVB); Transmission System for Handheld Terminals (DVB-H).*

[b-ETSI TS 102 250-1] ETSI TS 102 250-1 V2.2.1 (2011), *Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in mobile networks; Part 1: Assessment of Quality of Service.*

[b-ETSI TS 102 250-2] ETSI TS 102 250-2 V2.2.1 (2011), *Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in mobile networks; Part 2: Definition of Quality of Service parameters and their computation.*

[b-ETSI TS 102 250-3] ETSI TS 102 250-3 V2.3.1 (2013), *Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in mobile networks; Part 3: Typical procedures for Quality of Service measurement equipment.*

[b-ETSI TS 102 250-4] ETSI TS 102 250-4 V2.2.1 (2011), *Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in mobile networks; Part 4: Requirements for Quality of Service measurement equipment*.

[b-ETSI TS 102 250-5] ETSI TS 102 250-5 V2.4.1 (2013), *Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in mobile networks; Part 5: Definition of typical measurement profiles.*

[b-ETSI TS 102 250-6] ETSI TS 102 250-6 V1.2.1 (2004), *Speech Processing, Transmission and Quality Aspects (STQ); QoS aspects for popular services in GSM and 3G networks; Part 6: Post processing and statistical methods.*

[b-ETSI TS 102 250-7] ETS ETSI TS 102 250-7 V1.1.1 (2009), *Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in GSM and 3G networks; Part 7: Network based Quality of Service measurements*.

[b-ETSI TS 102 471] ETSI TS 102 471V1.4.1 (2010), *Digital Video Broadcasting (DVB);IP Datacast over DVB-H: Electronic Service Guide (ESG).*

[b-ETSI TR 102 473] ETSI TR 102 473 V1.1.1 (2006), *Digital Video Broadcasting (DVB);IP Datacast over DVB-H: Use Cases and Services.*

[b-ETSI TS 102 474] ETSI TS 102 474 V1.2.1 (2009), *Digital Video Broadcasting (DVB);IP Datacast over DVB-H: Service Purchase and Protection.*

[b-ETSI TS 123 107] ETSI TS 123 107 V5.13.0 (2004), *Universal Mobile Telecommunications System (UMTS); Quality of Service (QoS) concept and architecture (3GPP TS 23.107 version 5.13.0 Release 5).*

[b-ETSI TS 123 246] ETSI TS 123 246 V6.12.0 (2007), *Universal Mobile Telecommunications System (UMTS); LTE; Multimedia Broadcast/Multicast Service (MBMS); Architecture and functional description (3GPP TS 23.246 version 6.12.0 Release 6).*

[b-ETSI TS 125 101] ETSI TS 125 101 V9.13.0 (2013), *Universal Mobile Telecommunications System (UMTS); User Equipment (UE) radio transmission and reception (FDD) (3GPP TS 25.101 version 9.13.0 Release 9).*

[b-ETSI TS 126 114] ETSI TS 126 114 V7.17.0 (2012), *Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction (3GPP TS 26.114 version 7.17.0 Release 7).*

[b-ETSI TS 126 234] ETSI TS 126 234 V10.5.0 (2013), *Universal Mobile Telecommunications System (UMTS); LTE; Transparent end-to-end Packet-switched Streaming Service (PSS); Protocols and codecs (3GPP TS 26.234 version 10.5.0 Release 10).*

[b-ETSI TS 126 346] ETSI TS 126 346 V8.7.0 (2012), *Universal Mobile Telecommunications System (UMTS); LTE; Multimedia Broadcast/Multicast Service (MBMS); Protocols and codecs (3GPP TS 26.346 version 8.7.0 Release 8).*

[b-IETF RFC 2617] IETF RFC 2617 (1999), *HTTP Authentication: Basic and Digest Access Authentication.*

[b-IETF RFC 3261] IETF RFC 3261 (2002), *SIP: Session Initiation Protocol.*

[b-IETF RFC 3501] IETF RFC 3501 (2003), *Internet message access protocol – Version 4 rev1.*

[b-IETF RFC 3550] IETF RFC 3550 (2003), *RTP: A Transport Protocol for Real-Time Applications.*

[b-IETF RFC 3903] IETF RFC 3903 (2004), *Session Initiation Protocol (SIP) Extension for Event State Publication.*

[b-IETF RFC 5245] IETF RFC 5245 (2010), *Interactive Connectivity Establishment (ICE): A Protocol for Network Address Translator (NAT) Traversal for Offer/Answer Protocols.*

[b-IETF RFC 5322] IETF RFC 5322 (2008), *Internet Message Format.*

[b-IETF RFC 5389] IETF RFC 5389 (2008), *Session Traversal Utilities for NAT (STUN).*

[b-Nist 1] NIST/SEMATECH e-Handbook of Statistical Methods, Clause 2.2.2.1, http://www.itl.nist.gov/div898/handbook/mpc/section2/mpc221.htm, retrieved 5 August 2014.

[b-Nist 2] NIST/SEMATECH e-Handbook of Statistical Methods, Clause 6.3.2.3, http://www.itl.nist.gov/div898/handbook/pmc/section3/pmc323.htm, retrieved 5 August 2014.

[b-Nist 3] NIST/SEMATECH e-Handbook of Statistical Methods, Clause 6.3.2.4, http://www.itl.nist.gov/div898/handbook/pmc/section3/pmc324.htm, retrieved 5 August 2014.

[b-Nist 4] NIST/SEMATECH e-Handbook of Statistical Methods, Clause 6.3.2, http://www.itl.nist.gov/div898/handbook/pmc/section3/pmc32.htm, retrieved 5 August 2014.

[b-WAP Forum] WAP Forum B-WAP FORUM-MMSCTR-20020115-a (2002), *Wireless Application Protocol; Multimedia Messaging Service; Client TransactionsSpecification.* http://technical.openmobilealliance.org/tech/affiliates/wap/b-WAP Forum-mmsctr-20020115-a.pdf

[b-Schulzrinne] Schulzrinne, Henning (2001), *Internet Media on Demand: The Real Time Streaming Protocol*.

[b-Bates] Bates, D.M. and Chambers, J.M (1988), *Nonlinear regression Analysis and Applications, Wiley & Sons, 1988.*

[b-Bronstein] I.N. Bronstein, K.A. Semendjajew, G. Musiol, H. Mühlig (1999), *Handbook of Mathematics, Verlag Harri Deutsch; Frankfurt/Main.*

[b-Box] Box, Cox (1964), *An analysis of transformations, Journal of the Royal Statistical Society, Vol. B 26, p. 211-251.*

[b-Hardy] W.C. Hardy (2001), *QoS Measurement and Evaluation of Telecommunications Quality of Service, Wiley & Sons.*

[b-Hartung] J. Hartung (2002), *Lehr- und Handbuch der angewandten Statistik, Oldenbourg Wissenschaftsverlag, 13th meditation.*

[b-Law] A.M. Law, W.D. Kelton (2000), *Simulation modeling and analysis, McGraw-Hill, 3rd edition.*

[b-Mood] Mood, Graybill, Boes (1974), *Introduction to the theory of statistics, McCraw-Hill Statistics Series.*

[b-Venables] Venables, W.N. and Ripley, B.D. (1999), *Modern Applied Statistics with S-Plus, Springer Verlag.*

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