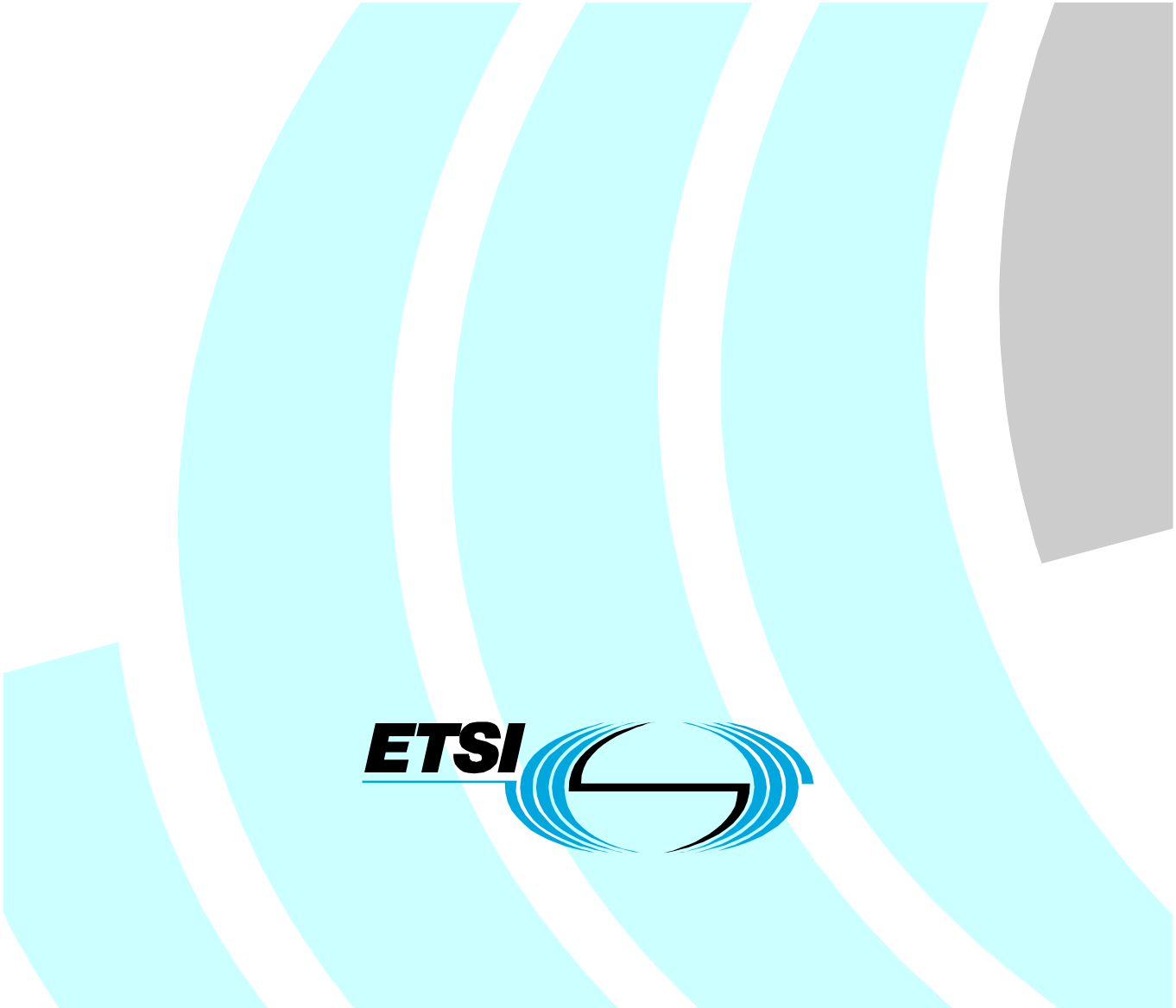
ETSI TS 102 250-2 V1.3.1 (2005-07)

Technical Specification

**Speech Processing, Transmission and Quality Aspects (STQ);
QoS aspects for popular services in GSM and 3G networks;
Part 2: Definition of Quality of Service parameters
and their computation**



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Contents

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Speech Processing, Transmission and Quality Aspects (STQ).

The present document is part 2 of a multi-part deliverable covering the QoS aspects for popular services in GSM and 3G networks, as identified below:

Part 1: "Identification of Quality of Service aspects";

Part 2: "Definition of Quality of Service parameters and their computation";

Part 3: "Typical procedures for Quality of Service measurement equipment";

Part 4: "Requirements for Quality of Service measurement equipment";

Part 5: "Definition of typical measurement profiles";

Part 6: "Post processing and statistical methods";

Part 7: "Sampling methodology".

Part 1 identifies QoS aspects for popular services in GSM and 3G networks. For each service chosen QoS indicators are listed. They are considered to be suitable for the quantitatively characterization of the dominant technical QoS aspects as experienced from the end-customer perspective.

Part 2 defines QoS parameters and their computation for popular services in GSM and 3G networks. The technical QoS indicators, listed in part 1, are the basis for the parameter set chosen. The parameter definition is split into two parts: the abstract definition and the generic description of the measurement method with the respective trigger points. Only measurement methods not dependent on any infrastructure provided are described in the present document. The harmonized definitions given in the present document are considered as the prerequisites for comparison of QoS measurements and measurement results.

Part 3 describes typical procedures used for QoS measurements over GSM, along with settings and parameters for such measurements.

Part 4 defines the minimum requirements of QoS measurement equipment for GSM and 3G networks in the way that the values and trigger-points needed to compute the QoS parameter as defined in part 2 can be measured following the procedures defined in part 3. Test-equipment fulfilling the specified minimum requirements, will allow to perform the proposed measurements in a reliable and reproducible way.

Part 5 specifies test profiles which are required to enable benchmarking of different GSM or 3G networks both within and outside national boundaries. It is necessary to have these profiles so that when a specific set of tests are carried out then customers are comparing "like for like" performance.

Part 6 describes procedures to be used for statistical calculations in the field of QoS measurement of GSM and 3G networks using probing systems.

Part 7 describes the field measurement method procedures used for QoS measurements over GSM where the results are obtained applying inferential statistics.

Introduction

All the defined quality of service parameters and their computations are based on field measurements. That indicates that the measurements were made from customers point of view (full end-to-end perspective, taking into account the needs of testing).

It is assumed that the end customer can handle his mobile and the services he wants to use (operability is not evaluated at this time). For the purpose of measurement it is assumed:

- that the service is available and not barred for any reason;
- routing is defined correctly without errors; and
- the target subscriber equipment is ready to answer the call.

Speech quality values from completed speech quality samples measured should only be employed by calls ended successfully for statistical analysis if the parameter speech quality per call is reported.

However, measured values from calls ended unsuccessfully (e.g. dropped) should be available for additional evaluations (e.g. with the speech quality per sample parameter) and therefore, must be stored.

Further preconditions may apply when reasonable.

1 Scope

The present document defines QoS parameters and their computation for popular services in GSM and 3G networks.

The technical QoS indicators, listed in TS 102 250-1 [4], are the basis for the parameter set chosen. The parameter definition is split into two parts: the abstract definition and the generic description of the measurement method with the respective trigger points. Only measurement methods not dependent on any infrastructure provided are described in the present document.

NOTE: Computation of certain parameters may depend in the vary cellular system, i.e. GSM or 3GPP specified 3G system. In this case respective notification is provided.

The harmonized definitions given in the present document are considered as the prerequisites for comparison of QoS measurements and measurement results.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

[1] ITU-T Recommendation P.862: "Perceptual evaluation of speech quality (PESQ), an objective method for end-to-end speech quality assessment of narrowband telephone networks and speech codecs".

[2] WAP-206-MMSCTR-20020115-a: "Wireless Application Protocol; Multimedia Messaging Service; Client Transactions".

[3] PRD IR.43: "Typical procedures for QoS measurement equipment".

[4] ETSI TS 102 250-1: "Speech Processing, Transmission and Quality Aspects (STQ); QoS aspects for popular services in GSM and 3G networks; Part 1: Identification of Quality of Service aspects".

[5] ETSI TS 102 250-3: "Speech processing, Transmission and Quality Aspects (STQ); QoS aspects for popular services in GSM and 3G networks; Part 3: Typical procedures for Quality of Service measurement equipment".

[6] ITU-R Recommendation BS.1387-1: "Method for objective measurements of perceived audio quality".

[7] IETF RFC 3550 (2003): "RTP: A Transport Protocol for Real-Time Applications".

[8] IETF RFC 2326 (1998): "Real Time Streaming Protocol (RTSP)".

[9] ITU-T Recommendation P.862.1: "Mapping function for transforming P.862 raw result scores to MOS-LQO".

[10] ETSI TS 124 008: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Mobile radio interface Layer 3 specification; Core network protocols; Stage 3 (3GPP TS 24.008 Release 5)".

[11] ETSI TS 145 008: "Digital cellular telecommunications system (Phase 2+); Radio subsystem link control (3GPP TS 45.008 Release 5)".

[12] ETSI TS 129 002: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Mobile Application Part (MAP) specification (3GPP TS 29.002)".

3 Definitions and abbreviations

3.1 Definitions

For all QoS parameter definitions within the present document, the second column of the trigger point table - "Trigger Points" (from customer's point of view) - is mandatory (if present) for all QoS parameter definitions. In the case that the measurement system is capable of tracking details presented in the third column - "Technical Description" - the specific points indicated are also mandatory.

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

3G	3rd Generation
3GPP	Third Generation Partnership Project
AAL2	Asynchronous Transfer Mode Adaptation Layer type 2
ALCAP	Access Link Control Application Protocol
AM	Acknowledged Mode
ANS	Answer Message
APN	Access Point Name
AT Command	ATtention Command
ATD	ATtention Dial
ATDT	ATtention Dial Tone
CCCH	Common Control CHannel
CRLF	Carriage Return Line Feed
CS	Circuit Switched
CSD	Circuit Switched Data
DCCH	Dedicated Control CHannel
DCE	Data Circuit-terminating Equipment
DCH	Data CHannel
DCH-FP	Data CHannel Frame Protocol
DLDT	DownLink Direct Transfer
DP	Detection Point
DQ	Data Quality
DT	Direct Transfer
DTE	Data Terminal Equipment
FACH	Forward Access CHannel
FTP	File Transfer Protocol
GGSN	Gateway GPRS Support Node
GMSC	Gateway Mobile Switching Centre
GPRS	General Packet Radio Service
GSM	Global System for Mobile communications
HLR	Home Location Register
HTTP	HyperText Transfer Protocol
IAM	Initial Address Message
ICMP	Internet Control Message Protocol
IMEI	International Mobile Equipment Identification
IP	Internet Protocol
ISUP	ISDN User Part
IWMSC	Inter Working Mobile Switching Centre
KPI	Key Performance Indicator

L1	Layer 1
L3	Layer 3
MGW	Media GateWay
MMS	Multimedia Messaging Service
MMSC	Multimedia Messaging Service Centre
MO	Mobile Originated
MOS	Mean Opinion Score
MOS-LQO	Mean Opinion Score - Listening speech Quality Objective
MS	Mobile Station
MSC	Mobile Switching Centre
MT	Mobile Terminated
NBAP	Node B Application Part
PDP	Packet Data Protocol
PLMN	Public Land Mobile Network
POP3	Post Office Protocol version 3
PS	Packet Switched
PSD	Packet Switched Data
QoS	Quality of Service
RAB	Radio Access Bearer
RACH	Random Access CHannel
RANAP	Radio Access Network Application Protocol
RAS	Remote Access Service
REL	Release Message
RNC	Radio Network Controller
RRC	Radio Resource Control
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
RTSP	Real Time Streaming Protocol
RX	Reception
SCCP	Signalling Connection Control Part
SDCCH	Stand-alone Dedicated Control CHannel
SGSN	Serving GPRS Support Node
SMS	Short Message Service
SMSC	Short Message Service Centre
SMTP	Simple Mail Transfer Protocol
SpQ	Speech Quality
SYN	TCP synchronize flag
TBF	Temporary Block Flow
TCP	Transmission Control Protocol
TCP-HS	Transmission Control Protocol Handshake
TX	Transmission
UE	User Equipment
ULDT	UpLink Direct Transfer
UM	Unacknowledged Mode
UMTS	Universal Mobile Telecommunications System
VLR	Visitor Location Register
VT	Video Telephony
WAP	Wireless Application Protocol
WGR	WAP Get Request
WSP	Wireless Session Protocol
WTP	Wireless Transport Protocol

4 QoS Parameter

4.1 Overview

Figure 1 shows a model for quality of service parameters. This model has three layers.

The first layer is the Network Access, the basic requirement for all the other QoS aspects and QoS parameters. The outcome of this layer is the QoS parameter Network Accessibility.

The second layer contains the other three QoS aspects Service Access, Service Integrity and Service Retainability.

The different services are located in the third layer. Their outcome are the QoS parameters.

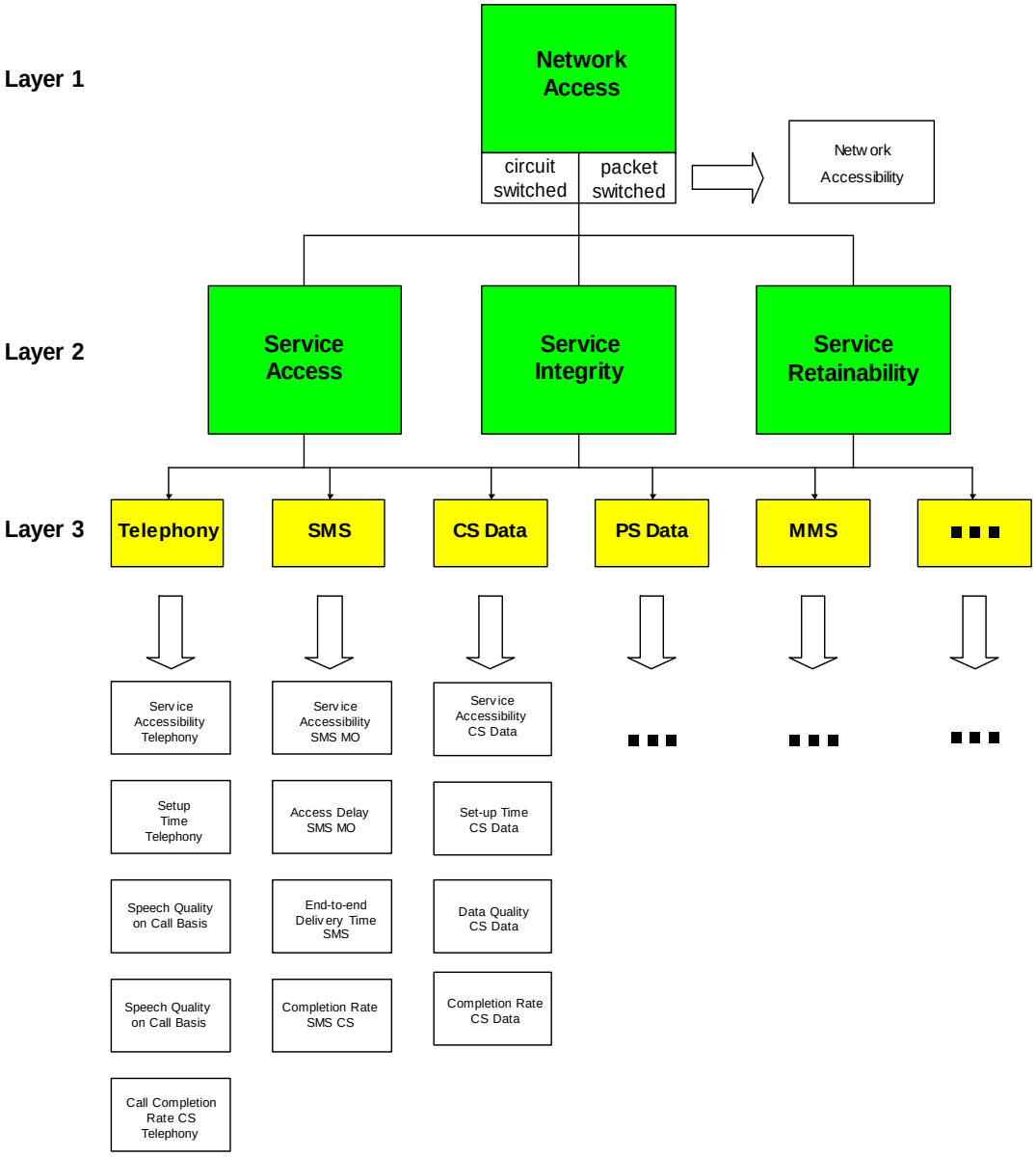


Figure 1: QoS aspects and the corresponding QoS parameters

4.2 Service independent

4.2.1 Radio Network Availability (RNAv) [%]

4.2.1.1 Abstract definition

Probability that the mobile services are offered to a user.

4.2.1.2 Computation (GSM/GPRS)

Abstract equation:

$$\text{Radio Network Availability}\,[\%] = \frac{\text{No. of probing attempts with mobile services available}}{\text{No. of all probing attempts}} \times 100\ \%$$

Trigger points:

Radio Network Availability is given if the following conditions are met:

- C1-Criteria > 0. Any emergency camping on any other than the target networks is considered as no network.

NOTE: For information on how the C1-Criteria is defined please refer to [11].

- GPRS: Specific signalling contained in System Information 3 exists on cell selection.

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

4.2.1.3 Computation (UMTS)

To be specified.

4.2.2 Network Accessibility (NAc) [%]

4.2.2.1 Abstract definition

Probability that the user can perform a successful registration on the PLMN.

4.2.2.2 Computation

Abstract equation:

$$\text{Network Accessibility}\,[\%] = \frac{\text{No. of successful registrations on the PLMN}}{\text{No. of all registration attempts}} \times 100\ \%$$

Trigger points:

Event	Trigger points from customer's point of view	Technical description (AT command)
Successful registration	Start: User turns mobile on.	Start: -
	Stop: Operator/PS logo appears in the display of the UE.	Stop: 'at+creg?' returns <stat> = 1 (CS), 'at+cgreg?' returns <stat> = 1 (PS).
Unsuccessful registration	Stop trigger point not reached.	

Remarks:

1) The AT command 'at+creg?' will return <stat> = 1 if the UE is registered on the home network, both for GSM and UMTS, i.e. it cannot differentiate if the UE is registered to GSM or UMTS (see 3GPP TS 27.007, AT command set for UE). Conform to this behaviour 'at+cgreg?' returns <stat> = 1 if the UE is registered either to GPRS or UMTS.

   The access technology selected by the UE can be verified with 'at+cops?'. This command will return <AcT> = 0 for GSM and <AcT> = 2 for UMTS.

The Network Accessibility is checked once at the start of a probing cycle (e.g. with the AT command 'at+creg?;+cops?').

4.3 Telephony

4.3.1 Service Accessibility Telephony (SA-T) [%]

4.3.1.1 Abstract definition

Probability that the end-customer can access the Mobile Telephony Service when requested if it is offered by display of the network indicator on the Mobile Equipment.

4.3.1.2 Computation

There are two possibilities for a successful call attempt:

- the customer hears the alerting;
- B-party is busy.

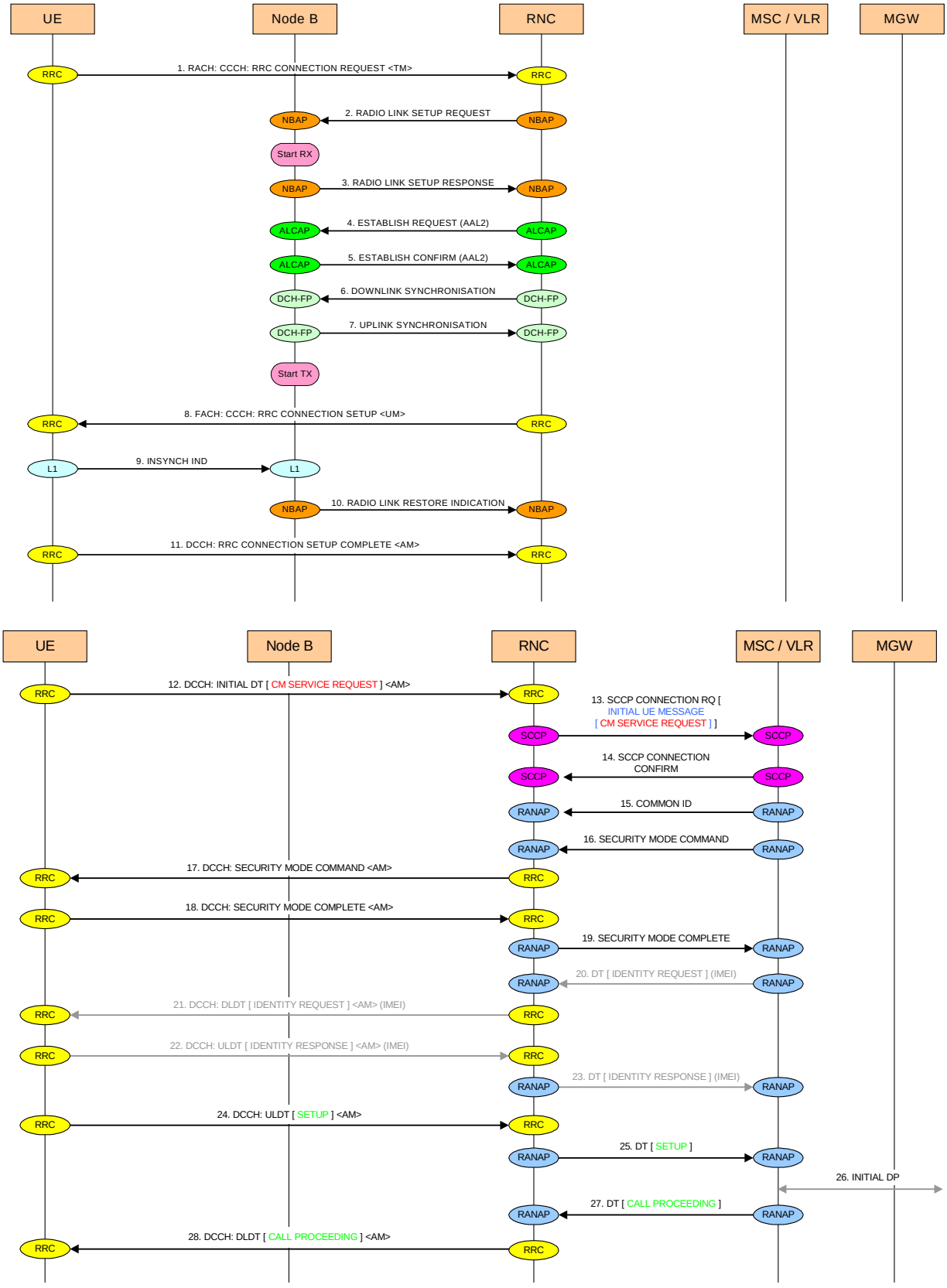
It is assumed that the routing to the destination is successful (without any failures).

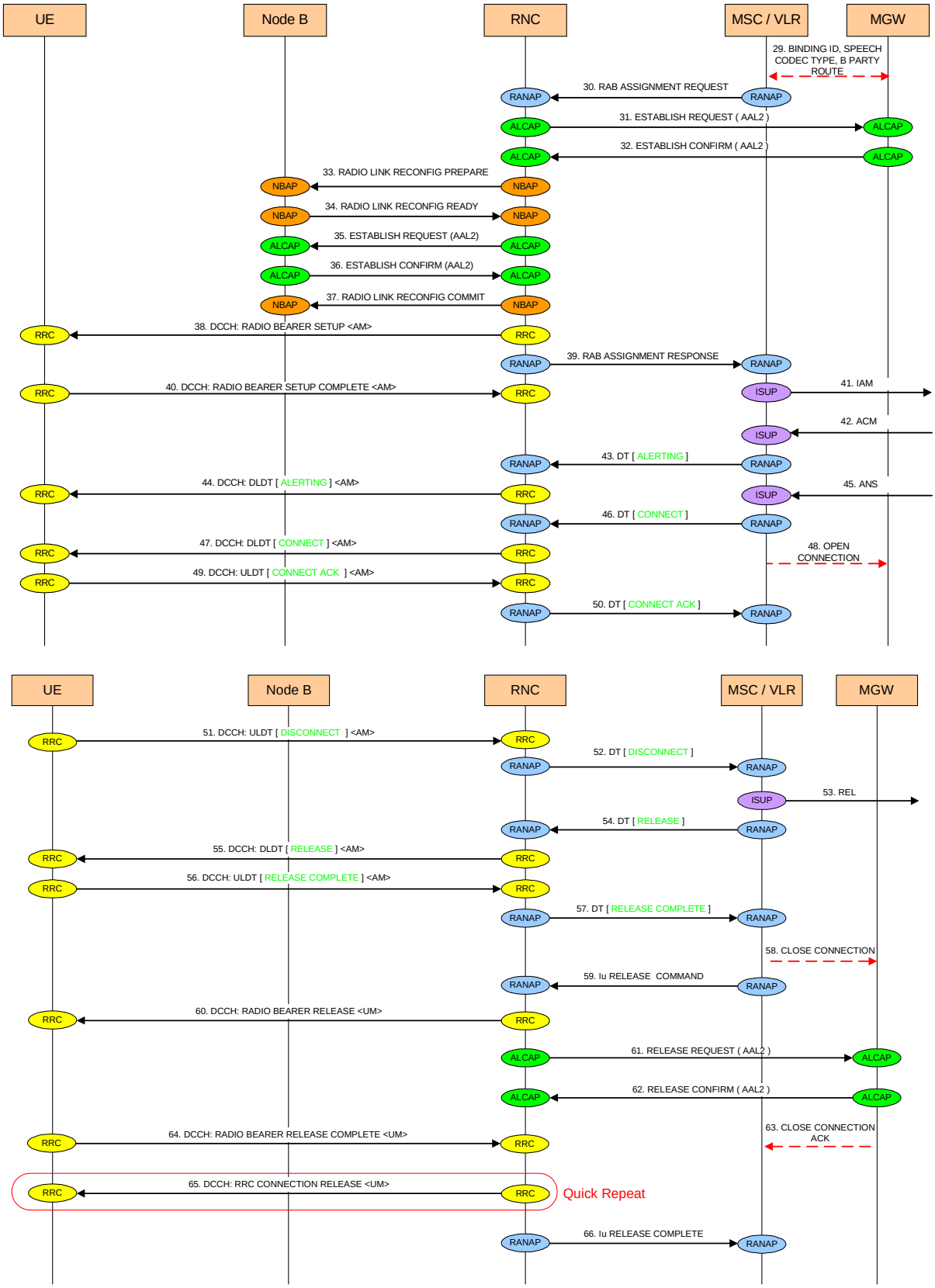
Abstract formula:

$$\text{Service Accessibility Telephony}[\%] = \frac{\text{Number of successful call attempts}}{\text{Number of call attempts}} \times 100\,\%$$

Trigger points:

Event (from equation)	Trigger Point (from customer's point of view)	Technical description/protocol part over 3G
Call Attempt	Push Send button (it is important to check, if coverage has been given when send button is pressed, otherwise this Call Attempt counts to Network Non Accessibility (NNA)).	The RRC CONNECTION REQUEST message carried on the CCCH logical channel and mapped to the RACH transport channel is sent. (Figure 2; signalling point number 1). Comment: It can be more than one RRC CONNECTION REQUEST message per Call Attempt, only the first RRC CONNECTION REQUEST should be taken into account for the calculation.
Successful call attempt	Alerting or busy tone heard by the A-party coming from B-party	The CONNECT message on the DCCH logical channel is not passed from the MSC to the UE to indicate that the connection has been established. (Figure 2; signalling point number 47). NOTE: With automatic tools there is not a significant difference between consider the alerting or the connect message, as the answer machine should always answer immediately.


UE
Node B
RNC
MSC / VLR
MGW
RRC
1. RACH: CCCH: RRC CONNECTION REQUEST <TM>
NBAP
2. RADIO LINK SETUP REQUEST
Start RX
3. RADIO LINK SETUP RESPONSE
ALCAP
4. ESTABLISH REQUEST (AAL2)
5. ESTABLISH CONFIRM (AAL2)
DCH-FP
6. DOWNLINK SYNCHRONISATION
7. UPLINK SYNCHRONISATION
Start TX
8. FACH: CCCH: RRC CONNECTION SETUP <UM>
L1
9. INSYNCH IND
10. RADIO LINK RESTORE INDICATION
11. DCCH: RRC CONNECTION SETUP COMPLETE <AM>
12. DCCH: INITIAL DT [CM SERVICE REQUEST] <AM>
13. SCCP CONNECTION RQ [INITIAL UE MESSAGE [CM SERVICE REQUEST]]
SCCP
14. SCCP CONNECTION CONFIRM
RANAP
15. COMMON ID
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>


UE
Node B
RNC
MSC / VLR
MGW
29. BINDING ID, SPEECH CODEC TYPE, B PARTY ROUTE
30. RAB ASSIGNMENT REQUEST
31. ESTABLISH REQUEST (AAL2)
32. ESTABLISH CONFIRM (AAL2)
33. RADIO LINK RECONFIG PREPARE
34. RADIO LINK RECONFIG READY
35. ESTABLISH REQUEST (AAL2)
36. ESTABLISH CONFIRM (AAL2)
37. RADIO LINK RECONFIG COMMIT
38. DCCH: RADIO BEARER SETUP <AM>
39. RAB ASSIGNMENT RESPONSE
40. DCCH: RADIO BEARER SETUP COMPLETE <AM>
41. IAM
42. ACM
43. DT [ALERTING]
44. DCCH: DLDT [ALERTING] <AM>
45. ANS
46. DT [CONNECT]
47. DCCH: DLDT [CONNECT] <AM>
48. OPEN CONNECTION
49. DCCH: ULDT [CONNECT ACK] <AM>
50. DT [CONNECT ACK]
51. DCCH: ULDT [DISCONNECT] <AM>
52. DT [DISCONNECT]
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>
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

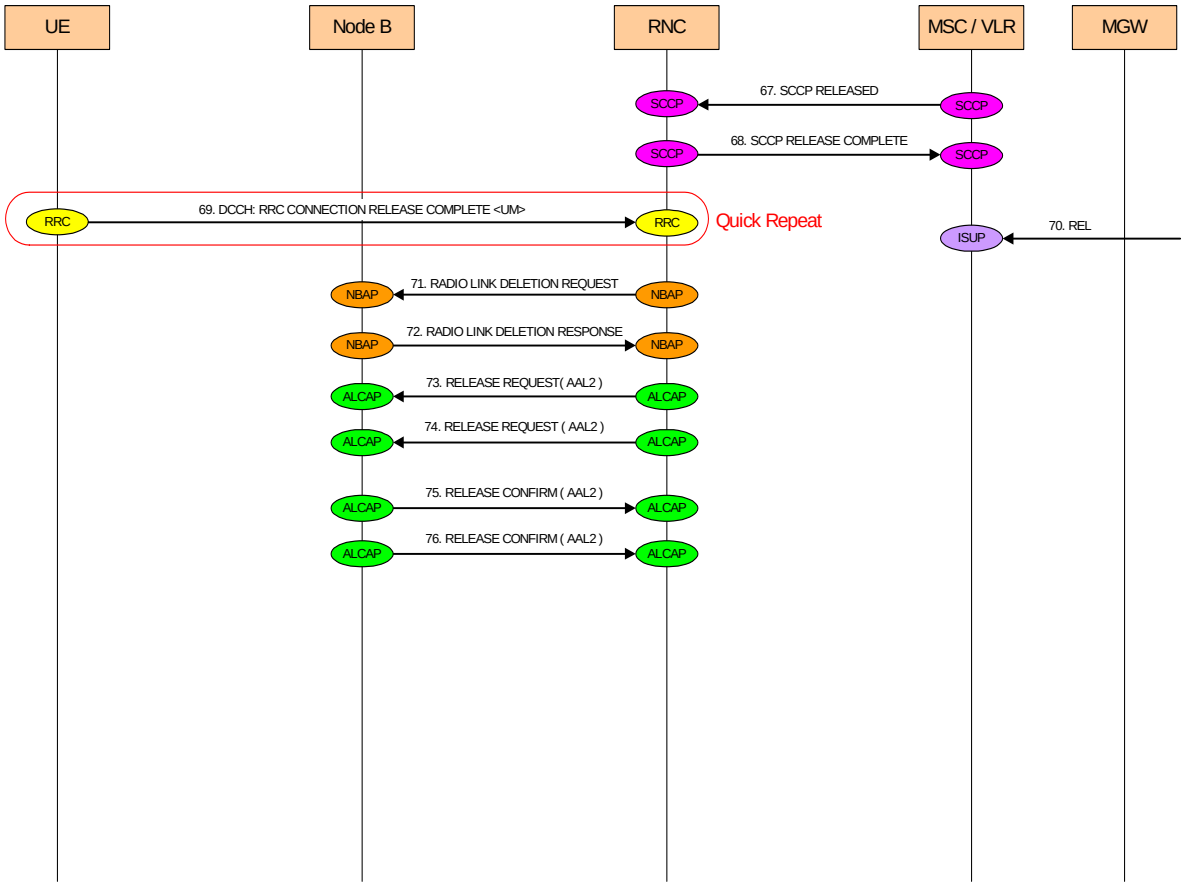


Figure 2: 3G Voice Signalling Flow Chart: Mobile Originated Call Establishment Procedure

4.3.2 Setup Time Telephony (ST-T) [s]

4.3.2.1 Abstract definition

Time between sending of complete address information and receipt of call set-up notification.

4.3.2.2 Computation

Abstract formula:

$$\text{Setup Time Telephony } [s] = t_2 - t_1$$

t_2: point of time where connect is established (e.g. alerting or subscriber busy is detected by test equipment), see note.

t_1: point of time where the customer presses the send button on mobile equipment.

NOTE: If you do not establish an end-to-end connection afterwards you must ignore this measurement. It is assumed that early traffic channel assignment is used.

Trigger points:

Event (from equation)	Trigger Point (from customer's point of view)	Technical description/protocol part over 3G
Call Attempt	Push Send button	The RRC CONNECTION REQUEST message carried on the CCCH logical channel and mapped to the RACH transport channel is sent. (Figure 2; signalling point number 1). Comment: It can be more than one RRC CONNECTION REQUEST message per Telephony Call Attempt, the first RRC CONNECTION REQUEST should be taken into account for the time calculation.
Connection established (Successful call attempt)	Alerting or busy tone heard by the A-party coming from B-party	The CONNECT message on the DCCH logical channel is not passed from the MSC to the UE to indicate that the connection has been established. (Figure 2; signalling point number 47). NOTE: With automatic tools there is not a significant difference between consider the alerting or the connect message, as the answer machine should always answer immediately.

4.3.3 Speech Quality on Call Basis (SpQ-C)

4.3.3.1 Abstract definition

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.

4.3.3.2 Computation

The validation of the end-to-end quality is made using the $MOS_{\_LQO}$ scale. This scale describes the opinion of customers with voice transmission and its troubles (noise, robot voice, echo, dropouts etc). The speech quality measurement is taken per call. An aggregation should be made on one value for speech quality per call.

Reference: ITU-T Recommendation P.862 [1] in conjunction with ITU-T Recommendation P.862.1 [9].

Abstract formula:

$$\text{Speech Quality on Call Basis (received A - side)} = f(MOS_{\_LQO})$$
$$\text{Speech Quality on Call Basis (received B - side)} = f(MOS_{\_LQO})$$

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).

Trigger points:

NOTE: The acoustic behaviour of terminals is not part of this speech quality measurement.

Event (from equation)	Trigger Point (from customer's point of view)	Technical description/protocol part over 3G
	Interchange speech samples between a-party and b-party	A 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 2; signalling point number 47).
	Release of connection	A DISCONNECT message on the DCCH logical channel is sent from the UE (message sent when the user ends the call). (Figure 2; signalling point number 51).

4.3.4 Speech Quality on Sample Basis (SpQ-S)

4.3.4.1 Abstract definition

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.

4.3.4.2 Computation

The validation of the end-to-end quality is made using the MOS scale. This scale describes the opinion of customers with voice transmission and its troubles (noise, robot voice, echo, dropouts etc). The speech quality measurement is taken per sample. An aggregation for measurement campaigns or parts of it should be made on speech sample basis.

Reference: ITU-T Recommendation P.862 [1] in conjunction with ITU-T Recommendation P.862.1 [9].

Abstract formula:

$$\text{Speech Quality on Sample Basis (received A - side)} = f(MOS\_LQO)$$
$$\text{Speech Quality on Sample Basis (received B - side)} = f(MOS\_LQO)$$

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).

Trigger points: The same as for Speech Quality on call basis (see clause 4.3.3.2).

NOTE: The acoustic behaviour of terminals is not part of this speech quality measurement.

4.3.5 Call Completion Rate Circuit Switched Telephony (CCR-CS-T) [%]

4.3.5.1 Abstract definition

Probability that a successful call attempt is maintained for a predetermined time until it is released intentionally by A- or B-party.

4.3.5.2 Computation

Abstract formula:

$$\text{Call Completion Rate CS Telephony} = \frac{\text{Number of intentionally terminated telephony calls}}{\text{Number of successful telephony call attempts}} \times 100\%$$

Trigger points:

Event (from equation)	Trigger Point (from customer's point of view)	Technical description/protocol part over 3G
Successful Telephony Call Attempt	Alerting or busy tone heard by the A-party coming from B-party;	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 2; signalling point number 47). NOTE: With automatic tools there is not a significant difference between consider the alerting or the connect message, as the answer machine should always answer immediately.
Intentionally Terminated Telephony Call	Release of connection directly by A- or B-party	A DISCONNECT message on the DCCH logical channel is sent from the UE (message sent when the user ends the call). (Figure 2; signalling point number 51).

4.4 Short Message Service (SMS)

4.4.1 Service Accessibility SMS MO (SA-SMS-MO) [%]

4.4.1.1 Abstract definition

Probability that the end-customer can access the Short Message Service when requested while it is offered by display of the network indicator on the Mobile Equipment. In this case the customer wants to send a Short Message.

4.4.1.2 Computation

NOTE: For the trigger point explained here, the connection over the air interface must be measured (e.g. Layer-3) and the answers of the SMSC must be counted statistically. The protocol for every connection shows the deviation from the successful service access.

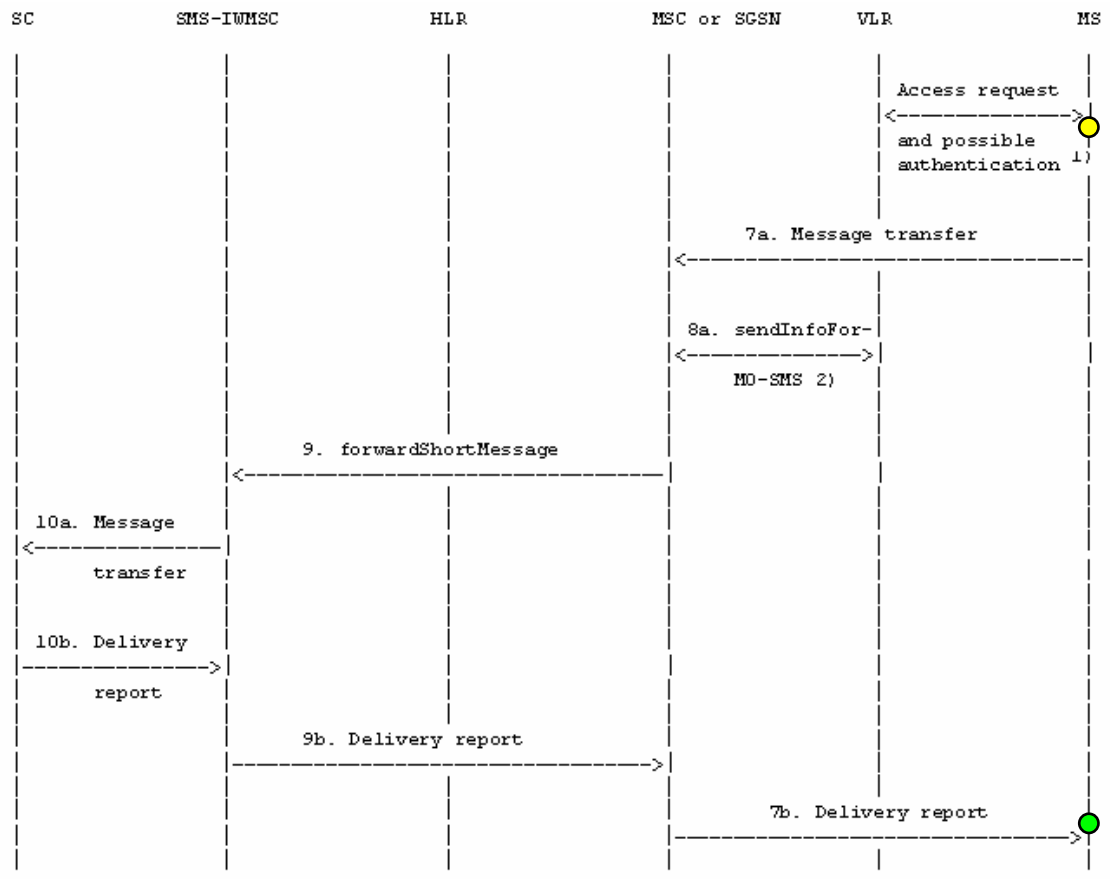
Only the first try should be measured. If the Short Message is established with the second try this should not be counted.

Abstract formula:

$$\text{Service Accessibility SMS MO}[\%] = \frac{\text{Number of successful SMS service attempts}}{\text{Number of all SMS service attempts}} \times 100\%$$

Trigger points:

Event (from equation)	Trigger Point (from customer's point of view)	Technical description/protocol part over 3G
SMS Service Attempt	Push send button (Initiate sending a SMS)	The 'Access request' is sent by the MS (MO). (Yellow point in figure 6). Detailed : CM Service Request is sent from MO.
Successful SMS Service Attempt	Receive the acknowledgement from the SMSC in the MO-party	'Delivery Report' is received in the MS (MO). (Green point in figure 6). Detailed : CP_DATA (RP_ACK) is received by MO.



NOTE 1: Described in TS 124 008 [10] and TS 129 002 [12].
NOTE 2: This operation is not used by the SGSN.

Figure 3: SMS Transaction flow - MO

4.4.2 Access Delay SMS MO (AD SMS-MO) [s]

4.4.2.1 Abstract definition

Time between sending a Short Message to a Short Message Centre (SMSC) and receiving the notification from the Short Message Centre.

4.4.2.2 Computation

Abstract formula:

$$\text{Access Delay SMS MO}[s] = t_{receive} - t_{send\,SMS}$$

$t_{receive}$: point of time the mobile equipment receives the confirmation from the SMS Centre.

$t_{send\,SMS}$: point of time the customer sends his SMS to the SMS Centre.

Trigger points:

Event (from equation)	Trigger Point (from customer's point of view)	Technical description/protocol part over 3G
$t_{send\ SMS}$	Push send button (Initiate sending a SMS)	The "Access request" is sent by the MS (MO). (Yellow point in figure 3). Detailed : CM Service Request is sent from MO.
$t_{receive}$	Acknowledgement from the SMSC is received in MO-party	"Delivery Report" is received in the MS (MO). (Green point in figure 3). Detailed : CP_-DATA (RP_ACK) is received by MO.

4.4.3 End-to-end Delivery Time SMS (DT-SMS) [s]

4.4.3.1 Abstract definition

Time between sending a Short Message to a Short Message Centre and receiving the very same Short Message on another mobile equipment.

4.4.3.2 Computation

Abstract formula:

$$\text{End - to - end Delivery Time SMS}[s] = t_{receive\ SMS} - t_{send\ SMS}$$

$t_{receive\ SMS}$: point of time the mobile equipment 2 receives the Short Message from mobile equipment 1.

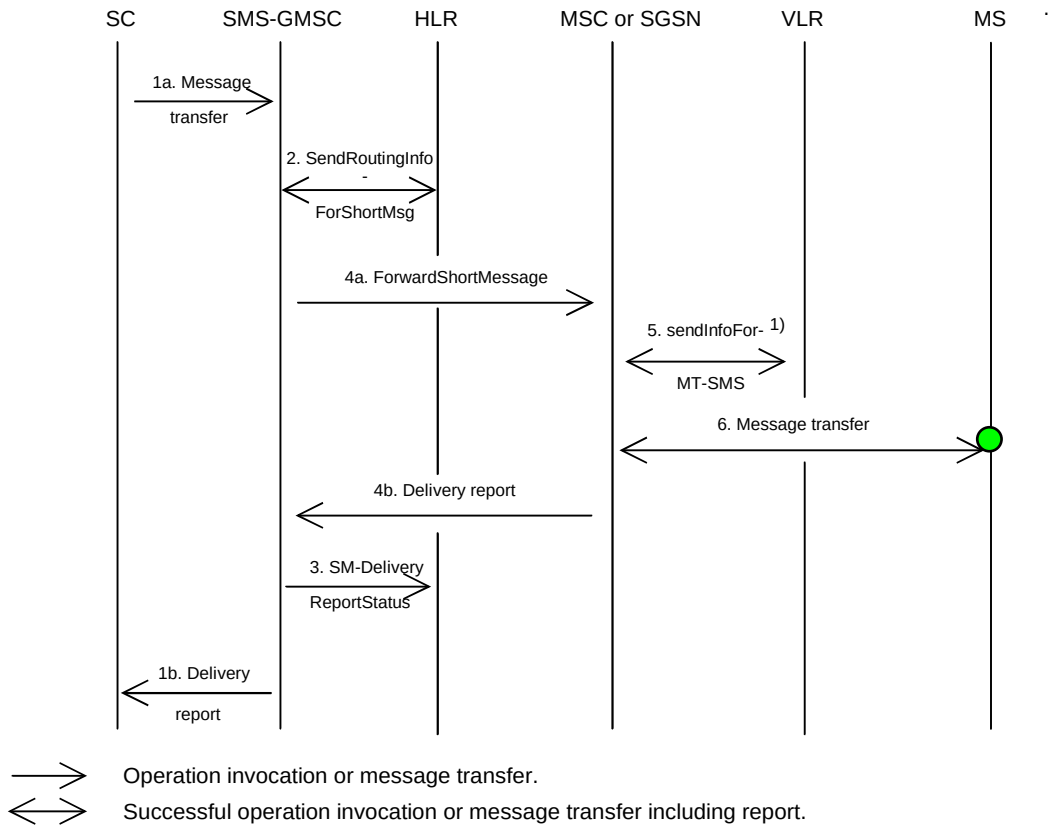
$t_{send\ SMS}$: point of time the customer sends his Short Message to the SMS Centre.

Trigger points:

Start SMS service attempt: Initiate sending a SMS.

End SMS service attempt: Receiving SMS on Mobile Equipment 2.

Event (from equation)	Trigger Point (from customer's point of view)	Technical description/protocol part over 3G
$t_{send\ SMS}$	Push send button (Initiate sending a SMS)	The 'Access request' is sent by the MS (MO). (Yellow point in figure 3). Detailed : CM Service Request is sent from MO.
$t_{receive\ SMS}$	The Short Message is received by MT-party's mobile	'Message Transfer' is received in the MS (MT). (Green point in figure 3). Detailed : CP_DATA (RP_ACK) is received by MT.



NOTE 1: This operation is not used by the SGSN.

Figure 4: SMS Transaction flow - MT

4.4.4 Completion Rate SMS Circuit Switched (CR-SMS-CS) [%]

4.4.4.1 Abstract definition

Ratio of received and send Test SMS from one mobile to another mobile part, excluding duplicate received and corrupted Test SMS.

A corrupted Test SMS is a SMS with at least one bit error.

For test and measurement purposes a message is considered valid if it is delivered successfully within a time window defined (see PRD IR.43 [3]).

4.4.4.2 Computation

Abstract formula:

$$\text{Completion Rate SMS CS } [\%] = \frac{\text{successful received Test SMS - duplicate received Test SMS - corrupted Test SMS}}{\text{Number of all send Test SMS}} \times 100\%$$

Trigger points:

Event (from equation)	Trigger Point (from customer's point of view)	Technical description/protocol part over 3G
SMS Service Attempt	Push send button (Initiate sending a SMS)	The 'Access request' is sent by the MS (MO). (Yellow point in figure 3). Detailed : CM Service Request is sent from MO.
Successful Received Test SMS	The Short Message is received by MT-party's mobile	'Message Transfer' is received in the MS (MT). (Green point in figure 4). Detailed : CP_DATA (RP_ACK) is received by MT.

4.5 Circuit Switched Data (CSD) Service

4.5.1 Service Accessibility Circuit Switched Data (SA - CSD) [%]

4.5.1.1 Abstract definition

Probability that the end-customer's DTE can access the Mobile Data Service when requested. This will be indicated by the DTE receiving the valid connect message from the distant DTE.

There are 2 layers of accessibility for CSD:

- access to the target network DCE;
- access to the required data service provided by a data server.

To a customer, these 2 events would be seamless and therefore the calculation for the service access should be a composite of these 2 activities. The field test system therefore must automate and combine the two layers to provide a single Service Accessibility CSD metric.

To combine the 2 layers should involve calculation of the success of the following actions:

- ATDT command including target number;
- receive Connect from target network DCE;
- send relevant command to target Data Server;
- receive valid response from Data Server.

The specific commands and responses from data servers are detailed in TS 102 250-3 [5].

4.5.1.2 Computation

A successful call attempt is when the A-party DTE receives valid response from test server. This can either be a dedicated data test server or a data server accessed when testing functionality via the public internet.

Abstract formula:

$$\text{Service Accessibility Circuit Switched Data}\,[\%] = \frac{\text{Number of successful call attempts}}{\text{Number of call attempts}} \times 100\%$$

Trigger points:

Beginning of the call attempt: ATDT command with dialled number sent by A-party DTE.

Successful call attempt: Valid response received from Data Server.

4.5.2 Setup Time Circuit Switched Data (ST - CSD) [s]

4.5.2.1 Abstract definition

Time between sending of complete address information in ATDT command by A-party and receipt of valid response from data server.

4.5.2.2 Computation

Abstract formula:

$$\text{Setup Time Circuit Switched Data [s]} = t_2 - t_1$$

t_1: point of time where A-party DTE sends ATDT command.

t_2: point of time where connect is established (valid response received by A-party from data server).

Trigger points:

Beginning of the Set-up time measurement: Sending of ATDT command by A-party.

Successful connection: Valid response received from Data Server.

4.5.3 Data Quality Circuit Switched Data (DQ-CSD)

To be defined.

4.5.4 Completion Rate Circuit Switched Data (CR-CSD) [%]

4.5.4.1 Abstract definition

Probability that a successful call attempt is not released except when intended by any of the parties involved in the call.

4.5.4.2 Computation

Abstract formula:

$$\text{Call completion Ratio Circuit Switched Data [\%]} = \frac{\text{Number of calls terminated by end users}}{\text{Number of successful data call attempts}} \times 100\%$$

Trigger points:

Successful call attempt: Valid response received by A-party DTE.

Completed call: DTE "ready" only when call ended by either party intentionally.

4.6 Packet Switched Data Services

The main QoS indicators defined for packet switched data services are:

- Service Accessibility Ratio (SA-PSD);
- Setup Time (ST-PSD);
- IP-Service Access Ratio (IPSA-PSD);
- IP-Service Setup Time (IPST-PSD);
- Completed Session Ratio (CoSeR-PSD);

- Session Time (SeT-PSD);
- Mean Data Rate (MDR-PSD);
- Data Transfer Cut-off Ratio (DTCoR-PSD);
- Round Trip Time (RTT-PSD).

Currently two main views about the best way to reflect the user's experience are in place: One preferring the payload throughput philosophy and the other preferring the transaction throughput philosophy:

- Method A, specified in clause 4.6.1, defines trigger points which are as independent as possible from the service used, therefore representing a more generic view (payload throughput).
- Method B, specified in clause 4.6.2, 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 figure 5 and figure 6: The start trigger point for the Mean Data Transfer 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 the present document shall measure both sets (Method A and B) of QoS indicators using commercial UEs.

In addition, a set of technical QoS indicators is defined, which is given in clause 4.6.3. Field test systems shall be able to measure these QoS indicators.

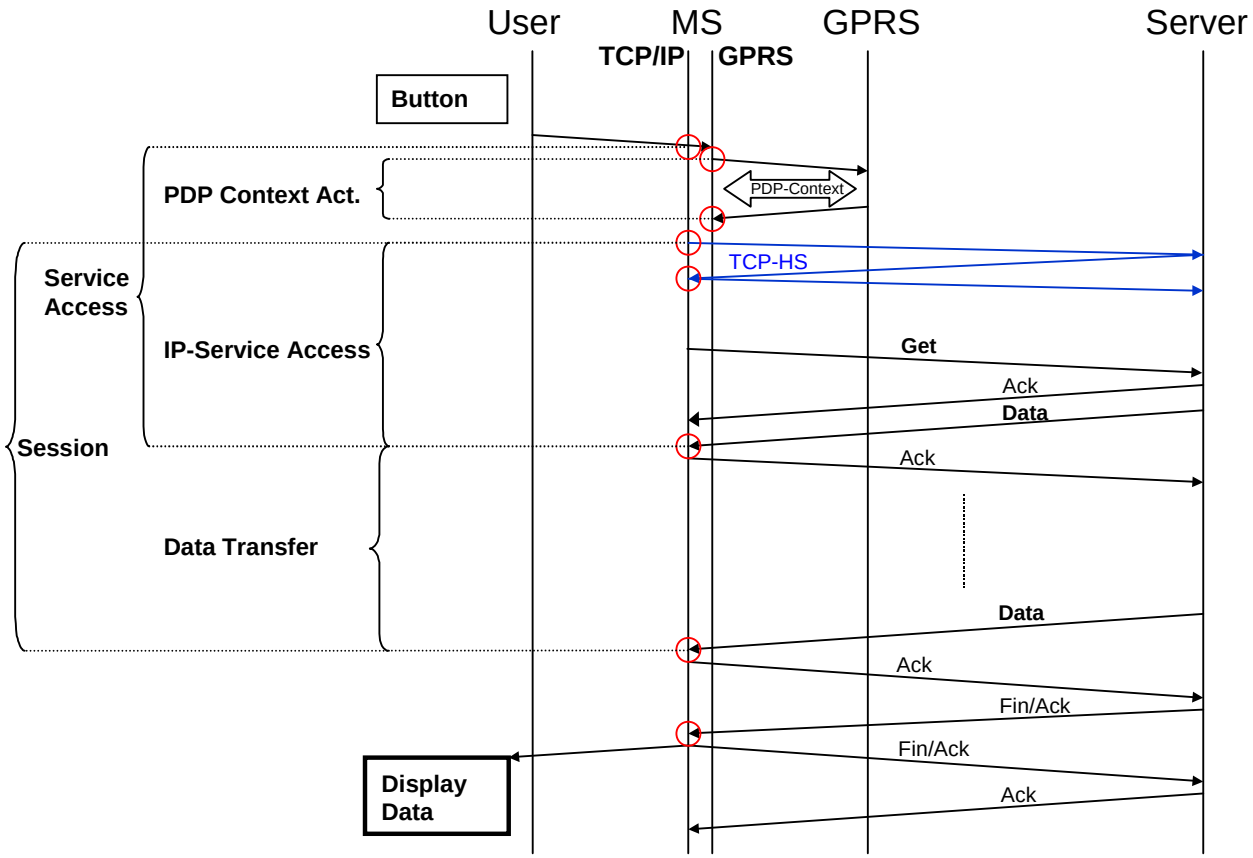


Figure 5: Key Performance Indicators Version A (Example: HTTP via GPRS)

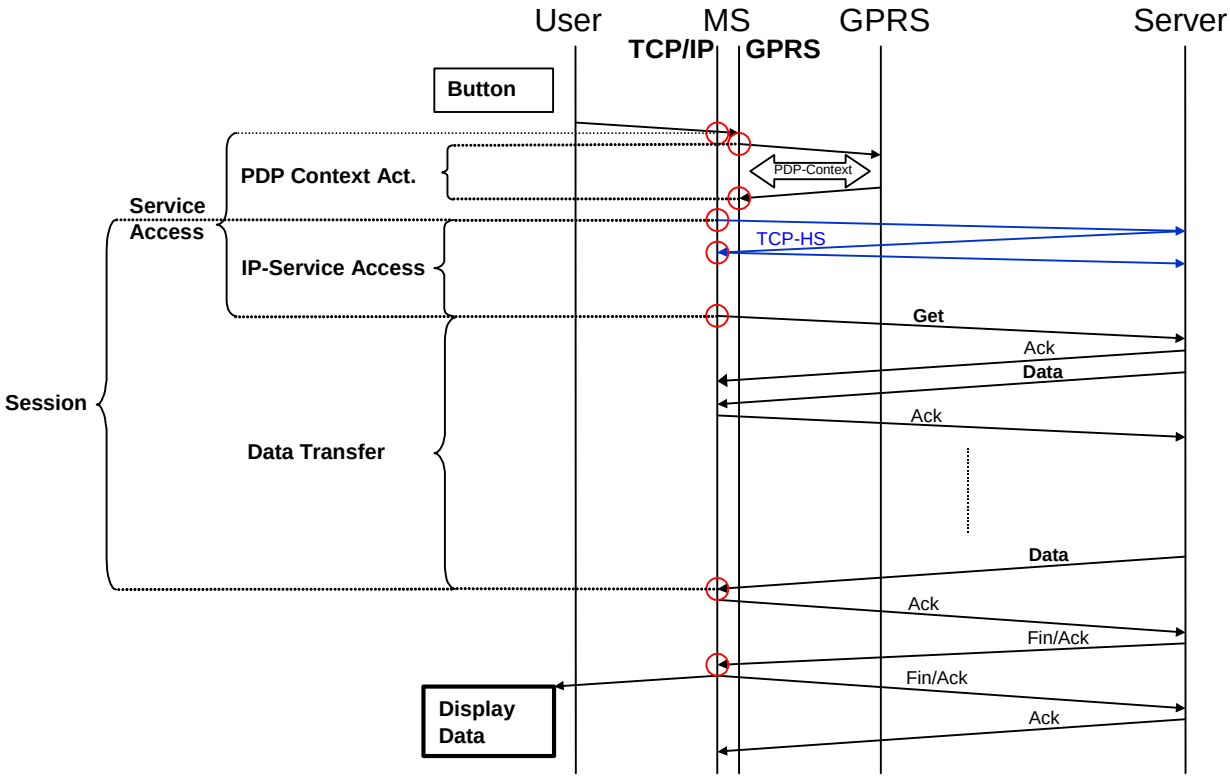


Figure 6: Key Performance Indicators Version B (Example: HTTP via GPRS)

4.6.1 Key Performance Indicators Method A

4.6.1.1 {Service} Service Accessibility Ratio (SA-PSD) [%]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.1.1.1 Abstract definition

The service accessibility ratio denotes the probability that a subscriber can establish a PDP context and access the service successfully.

4.6.1.1.2 Computation

Abstract equation:

$$\text{Service Accessibility Ratio}[\%] = \frac{\text{No. of successfull attempts to reach the point when content is sent or received}}{\text{No. of all attempts to reach the point when content is sent or received}} \times 100\,\%$$

Trigger points:

FTP (download), E-Mail POP3 (receiving), HTTP

- Start: ATD command from the mobile to the network.
- Stop: Reception of the first data packet containing content.

NOTE: The term "content" has a different meaning depending on the service that is accessed. In case of a FTP session content is a file, in case of a HTTP session it is a web page and the content of an E-Mail session is the text of the E-Mail.

FTP (upload), E-Mail SMTP (sending)

- Start: ATD command from the mobile to the network.
- Stop: Sending of the first data packet containing content.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio).

4.6.1.2 {Service} Setup Time (ST-PSD) [s]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.1.2.1 Abstract definition

The setup time describes 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.

Abstract equation:

$$\text{Setup Time [s]} = t_{\text{Content sent or received}} - t_{\text{Dial-up connection initiated}}$$

Trigger points:

FTP (download), E-Mail POP3 (receiving), HTTP

- Start: ATD command from the mobile to the network.
- Stop: Reception of the first data packet containing content.

FTP (upload), E-Mail SMTP (sending)

- Start: ATD command from the mobile to the network.
- Stop: Sending of the first data packet containing content.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio).

4.6.1.3 {Service} IP-Service Access Ratio (IPSA-PSD) [%]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.1.3.1 Abstract definition

The IP-service access ratio denotes the probability that a subscriber can establish an TCP/IP connection to the server of a service successfully.

4.6.1.3.2 Computation

Abstract equation:

$$\text{IP - Service Access Ratio}[\%] = \frac{\text{No. of successfull attempts to a establish an IP connection to the server}}{\text{No. of all attempts to establish an IP connection to the server}} \times 100\,\%$$

Trigger points:

FTP (download), E-Mail POP3 (receiving), HTTP

- Start: First [SYN] sent.
- Stop: Reception of the first data packet containing content.

FTP (upload), E-Mail SMTP (sending)

- Start: First [SYN] sent.
- Stop: Sending of the first data packet containing content.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio) as well as the respective PDP context has to be activated (cf. PDP Context Activation Failure Ratio).

4.6.1.4 {Service} IP-Service Setup Time (IPST-PSD) [s]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.1.4.1 Abstract definition

The IP-service setup time is the time period needed to establish an 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.

4.6.1.4.2 Computation

Abstract equation:

$$\text{IP - Service Setup Time}[s] = t_{\text{Content sent or received}} - t_{\text{Query sent}}$$

Trigger points:

FTP (download), E-Mail POP3 (receiving), HTTP

- Start: First [SYN] sent.
- Stop: Reception of the first data packet containing content.

FTP (upload), E-Mail SMTP (sending)

- Start: First [SYN] sent.
- Stop: Sending of the first data packet containing content.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio) as well as the respective PDP context has to be activated (cf. PDP Context Activation Failure Ratio).

4.6.1.5 {Service} Completed Session Ratio (CoSeR-PSD) [%]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.1.5.1 Abstract definition

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

4.6.1.5.2 Computation

Abstract equation:

$$\text{Completed Session Ratio}[\%] = \frac{\text{Number of completed sessions}}{\text{Number of successfully started sessions}} \times 100\,\%$$

Trigger points:

FTP (download), E-Mail POP3 (receiving), HTTP

- Start: First [SYN] sent.
- Stop: Reception of the last data packet containing content.

FTP (upload), E-Mail SMTP (sending)

- Start: First [SYN] sent.
- Stop: Reception of the [FIN, ACK] for the last data packet containing content.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio) as well as the respective PDP context has to be activated (cf. PDP Context Activation Failure Ratio).

4.6.1.6 {Service} Session Time (SeT-PSD) [s]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.1.6.1 Abstract definition

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

4.6.1.6.2 Computation

Abstract equation:

$$\text{Session Time}[s] = t_{\text{Session end}} - t_{\text{Session start}}$$

Trigger points:

FTP (download), E-Mail POP3 (receiving), HTTP

- Start: First [SYN] sent.
- Stop: Reception of the last data packet containing content.

FTP (upload), E-Mail SMTP (sending)

- Start: First [SYN] sent.
- Stop: Reception of the [FIN, ACK] for the last data packet containing content.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio) as well as the respective PDP context has to be activated (cf. PDP Context Activation Failure Ratio).

4.6.1.7 {Service} Mean Data Rate (MDR-PSD) [kbit/s]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.1.7.1 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.

4.6.1.7.2 Computation

Abstract equation:

$$\text{Mean Data Rate}[\text{kbit/s}] = \frac{\text{User data transferred}[\text{kbit}]}{(\text{End of data transfer} - \text{Start of data transfer})[\text{s}]} \times 100\,\%$$

Trigger points: The average throughput is measured from opening the data connection to the end of the successful transfer of the content (file, e-mail or web page).

FTP (download), E-Mail POP3 (receiving), HTTP

- Start: Reception of the first data packet containing content.
- Stop: Reception of the last data packet containing content.

FTP (upload), E-Mail SMTP (sending)

- Start: Sending of the first data packet containing content.
- Stop: Reception of the [FIN, ACK] for the last data packet containing content.

Remark(s): The mobile station is already attached (cf. Attach Failure Ratio), a PDP context is activated (cf. PDP Context Activation Failure Ratio) and a service was accessed successfully (cf. Service Non-Accessibility).

4.6.1.8 {Service} Data Transfer Cut-off Ratio (DTCoR-PSD) [%]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.1.8.1 Abstract definition

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

4.6.1.8.2 Computation

Abstract equation:

$$\text{Data Transfer Cut} - \text{off Ratio}\,[\%] = \frac{\text{Number of incomplete data transfers}}{\text{Number of successfully started data transfers}} \times 100\%$$

Trigger points:

FTP (download), E-Mail POP3 (receiving), HTTP

- Start: Reception of the first data packet containing content.
- Stop: Reception of the last data packet containing content.

FTP (upload), E-Mail SMTP (sending)

- Start: Sending of the first data packet containing content.
- Stop: Reception of the [FIN, ACK] of the last data packet containing content.

Remark(s): The mobile station is already attached (cf. Attach Failure Ratio), a PDP context is activated (cf. PDP Context Activation Failure Ratio) and a service was accessed successfully (cf. Service Non-Accessibility).

4.6.2 Key Performance Indicators Method B

4.6.2.1 {Service} Service Accessibility Ratio (SA-PSD) [%]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.2.1.1 Abstract definition

The service accessibility ratio denotes the probability that a subscriber can establish a PDP context and access the service successfully.

4.6.2.1.2 Computation

Abstract equation:

$$\text{Service Accessibility Ratio}[\%] = \frac{\text{No. of successfull attempts to reach the point when content is sent or received}}{\text{No. of all attempts to reach the point when content is sent or received}} \times 100\%$$

Trigger points:

FTP (download), FTP (upload)

- Start: ATD command from the mobile to the network.
- Stop: Reception of the [ACK] from the [SYN, ACK].

E-Mail POP3 (receiving)

- Start: ATD command from the mobile to the network.
- Stop: Send RETR command.

E-Mail SMTP (sending)

- Start: ATD command from the mobile to the network.
- Stop: Reception of the positive acknowledgement (250) for the HELO command which was sent from the client before. This definition applies to the "none login procedure", for other login procedures the trigger point has to be defined.

HTTP

- Start: ATD command from the mobile to the network.
- Stop: Sending of the first GET command.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio).

4.6.2.2 {Service} Setup Time (ST-PSD) [seconds]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.2.2.1 Abstract definition

The setup time describes 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.

Abstract equation:

$$\text{Setup Time [s]} = t_{\text{Content sent or received}} - t_{\text{Dial-up connection initiated}}$$

Trigger points:

FTP (download), FTP (upload)

- Start: ATD command from the mobile to the network.
- Stop: Reception of the [ACK] from the [SYN, ACK].

E-Mail POP3 (receiving)

- Start: ATD command from the mobile to the network.
- Stop: Send RETR command.

E-Mail SMTP (sending)

- Start: ATD command from the mobile to the network.
- Stop: Reception of the positive acknowledgement (250) for the HELO command which was sent from the client before. This definition applies to the "none login procedure", for other login procedures the trigger point has to be defined.

HTTP

- Start: ATD command from the mobile to the network.
- Stop: Sending of the first GET command.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio).

4.6.2.3 {Service} IP-Service Access Ratio (IPSA-PSD)

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.2.3.1 Abstract definition

The IP-service access ratio denotes the probability that a subscriber can establish an TCP/IP connection to the server of a service successfully.

4.6.2.3.2 Computation

Abstract equation:

$$\text{IP - Service Access Ratio}[\%] = \frac{\text{No. of successfull attempts to establish an IP connection to the server}}{\text{No. of all attempts toestablish an IP connection the server}} \times 100\%$$

Trigger points:

FTP (download), FTP (upload)

- Start: First [SYN] sent.
- Stop: Reception of the [ACK] from the [SYN, ACK].

E-Mail POP3 (receiving)

- Start: First [SYN] sent.
- Stop: Send RETR command.

E-Mail SMTP (sending)

- Start: First [SYN] sent.
- Stop: Reception of the positive acknowledgement (250) for the HELO command which was sent from the client before. This definition applies to the "none login procedure", for other login procedures the trigger point has to be defined.

HTTP

- Start: First [SYN] sent.
- Stop: Sending of the first GET command.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio) as well as the respective PDP context has to be activated (cf. PDP Context Activation Failure Ratio).

4.6.2.4 {Service} IP-Service Setup Time (IPST-PSD)

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.2.4.1 Abstract definition

The IP-service setup time is the time period needed to establish an 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.

4.6.2.4.2 Computation

Abstract equation:

$$\text{IP - Service Setup Time}[s] = t_{\text{Content sent or received}} - t_{\text{Query sent}}$$

Trigger points:

FTP (download), FTP (upload)

- Start: First [SYN] sent.
- Stop: Reception of the [ACK] from the [SYN, ACK].

E-Mail POP3 (receiving)

- Start: First [SYN] sent.
- Stop: Send RETR command.

E-Mail SMTP (sending)

- Start: First [SYN] sent.
- Stop: Reception of the positive acknowledgement (250) for the HELO command which was sent from the client before. This definition applies to the "none login procedure", for other login procedures the trigger point has to be defined.

HTTP

- Start: First [SYN] sent.
- Stop: Sending of the first GET command.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio) as well as the respective PDP context has to be activated (cf. PDP Context Activation Failure Ratio).

4.6.2.5 {Service} Completed Session Ratio (CoSeR-PSD) [%]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.2.5.1 Abstract definition

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

4.6.2.5.2 Computation

Abstract equation:

$$\text{Completed Session Ratio}[\%] = \frac{\text{Number of completed sessions}}{\text{Number of successfully started sessions}} \times 100\%$$

Trigger points:

FTP (download), HTTP

- Start: First [SYN] sent.
- Stop: Reception of the last data packet containing content.

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

FTP (upload)

- Start: First [SYN] sent.
- Stop: Reception of the [FIN, ACK] sent from the server.

E-Mail POP3 (receiving)

- Start: First [SYN] sent.
- Stop: Reception of the data packet containing the finish sequence (CRLF.CRLF).

E-Mail SMTP (sending)

- Start: First [SYN] sent.
- Stop: Reception of the positive acknowledgement (250) for the EOM command.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio) as well as the respective PDP context has to be activated (cf. PDP Context Activation Failure Ratio).

4.6.2.6 {Service} Session Time (SeT-PSD)

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.2.6.1 Abstract definition

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

4.6.2.6.2 Computation

Abstract equation:

$$\text{Session Time [s]} = t_{\text{Session end}} - t_{\text{Session start}}$$

Trigger points:

FTP (download), HTTP

- Start: First [SYN] sent.
- Stop: Reception of the last data packet containing content.

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

FTP (upload)

- Start: First [SYN] sent.
- Stop: Reception of the [FIN, ACK] sent from the server.

E-Mail POP3 (receiving)

- Start: First [SYN] sent.
- Stop: Reception of the data packet containing the finish sequence (CRLF.CRLF).

E-Mail SMTP (sending)

- Start: First [SYN] sent.
- Stop: Reception of the positive acknowledgement (250) for the EOM command.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio) as well as the respective PDP context has to be activated (cf. PDP Context Activation Failure Ratio).

4.6.2.7 {Service} Mean Data Rate (MDR-PSD) [kbit/s]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.2.7.1 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.

4.6.2.7.2 Computation

Abstract equation:

$$\text{Mean Data Rate}[\text{kbit/s}] = \frac{\text{User data transferred}\,[\text{kbit}]}{\text{End of data transfer}\,[\text{s}] - \text{Start of data transfer}\,[\text{s}]}$$

Trigger points: The average throughput is measured from opening the data connection to the end of the successful transfer of the content (file, e-mail or web page).

FTP (download)

- Start: Reception of the [ACK] from the [SYN, ACK].
- Stop: Reception of the last data packet containing content.

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

FTP (upload)

- Start: Reception of the [ACK] from the [SYN, ACK].
- Stop: Reception of the [FIN, ACK] sent from the server.

E-Mail POP3 (receiving)

- Start: Send RETR command.
- Stop: Reception of the data packet containing the finish sequence(CRLF.CRLF).

E-Mail SMTP (sending)

- Start: Reception of the positive acknowledgement (250) for the HELO command which was sent from the client before. This definition applies to the "none login procedure", for other login procedures the trigger point has to be defined.

- Stop: Reception of the positive acknowledgement (250) for the EOM command.

HTTP

- Start: Sending of the first GET command.
- Stop: Reception of the last data packet containing content.

Remark(s): The mobile station is already attached (cf. Attach Failure Ratio), a PDP context is activated (cf. PDP Context Activation Failure Ratio) and a service was accessed successfully (cf. Service Non-Accessibility).

4.6.2.8 {Service} Data Transfer Cut-off Ratio (DTCoR-PSD) [%]

Service(s) defined: FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.2.8.1 Abstract definition

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

4.6.2.8.2 Computation

Abstract equation:

$$\text{Data Transfer Cut - off Ratio}[\%] = \frac{\text{Number of incomplete data transfers}}{\text{Number of successfully started data transfers}} \times 100\%$$

Trigger points:

FTP (download)

- Start: Reception of the [ACK] from the [SYN, ACK].
- Stop: Reception of the last data packet containing content.

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

FTP (upload)

- Start: Reception of the [ACK] from the [SYN, ACK].
- Stop: Reception of the [FIN, ACK] sent from the server.

E-Mail POP3 (receiving)

- Start: Send RETR command.
- Stop: Reception of the data packet containing the finish sequence (CRLF.CRLF).

E-Mail SMTP (sending)

- Start: Reception of the positive acknowledgement (250) for the HELO command which was sent from the client before. This definition applies to the "none login procedure", for other login procedures the trigger point has to be defined.
- Stop: Reception of the positive acknowledgement (250) for the EOM command.

HTTP

- Start: Sending of the first GET command.
- Stop: Reception of the last data packet containing content.

Remark(s): The mobile station is already attached (cf. Attach Failure Ratio), a PDP context is activated (cf. PDP Context Activation Failure Ratio) and a service was accessed successfully (cf. Service Non-Accessibility).

4.6.3 Performance Indicators

4.6.3.1 Attach Failure Ratio [%]

4.6.3.1.1 Abstract definition

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

4.6.3.1.2 Computation

Abstract equation:

$$\text{Attach Failure Ratio}[\%] = \frac{\text{No. of unsuccessful attach attempts}}{\text{No. of all attach attempts}} \times 100\%$$

Connection to other parameters: Unavailability

Trigger points:

- Start: Mobile sends the attach request message.
- Stop: Mobile receives the attach accept message.

Remark(s):

1) 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 TBF request is initiated).

2) It might occur that the mobile station 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 TS 124 008 [10]). Therefore the timeout interval for the attach procedure is 75 seconds, i.e. if the attach procedure was not completed after 75 seconds it is considered as failure.

   These retries should not have impact on the attach failure ratio, since only one attach request message should be counted in the calculation.

3) The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability).

The above timeouts are considered to be "technical" timeouts and do not necessarily reflect actual user behaviour. Users may not be prepared to wait as long as the "technical timeout" values before considering the transaction as failed.

The "technical timeouts" should be used for gathering the measurements, and then potentially shorter "user behaviour timeouts" can be used in post-processing of the results to calculate the actual KPI values. In this way, results will not be discarded that only just exceed the "user behaviour timeouts". This could be useful when producing distribution tables/graphs of results.

4.6.3.2 Attach Setup Time [seconds]

4.6.3.2.1 Abstract definition

This attach setup time describes the time period needed to attach to the PS network.

4.6.3.2.2 Computation

Abstract equation:

$$\text{Attach Setup Time}[s] = t_{\text{attach complete}} - t_{\text{attach request}}$$

Connection to other parameters: Unavailability

Remark(s):

1) The difference between an attach of a known subscriber and an unknown subscriber will be reflected in the time period indicating the attach setup time. In case of an unknown subscriber (meaning that the SGSN has changed since the detach, or if it is the very first attach of the mobile to the network), the SGSN contacts the HLR in order to receive the subscriber data. The attach setup time of an unknown subscriber will be slightly longer than the one of a known subscriber.

2) While determining the average attach setup time only successful attach attempts are included in the calculations.

3) The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability).

Trigger points:

- Start: Point of time when the mobile sends the attach request message.
- Stop: Point of time when the mobile receives the attach accept message.

4.6.3.3 {Service} PDP Context Activation Failure Ratio [%]

Service(s) defined: All

4.6.3.3.1 Abstract definition

The PDP context activation failure ratio denotes 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.

4.6.3.3.2 Computation

Abstract equation:

$$\text{PDP Context Activation Failure Ratio}[\%] = \frac{\text{No. of unsuccessful PDP context activation attempts}}{\text{No. of all PDP context activation attempts}} \times 100\%$$

Connection to other parameters:

- Unavailability.
- Attach Failure Ratio.

Trigger points:

- Start: Mobile sends the PDP context activation request message.
- Stop: Mobile receives the PDP context activation accept message.

Remark(s):

1) It might occur that the mobile station 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, cf. TS 124 008 [10]). Therefore the timeout interval for the PDP context activation procedure is 150 seconds, i.e. if the PDP context activation procedure was not completed after 150 seconds it is considered as failure.

   These retries should not have impact on the activation failure ratio, since only one PDP context activation request message should be counted in the calculation.

2) The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio).

The above timeouts are considered to be "technical" timeouts and do not necessarily reflect actual user behaviour. Users may not be prepared to wait as long as the "technical timeout" values before considering the transaction as failed.

The "technical timeouts" should be used for gathering the measurements, and then potentially shorter "user behaviour timeouts" can be used in post-processing of the results to calculate the actual KPI values. In this way, results will not be discarded that only just exceed the "user behaviour timeouts". This could be useful when producing distribution tables/graphs of results.

4.6.3.4 {Service} PDP Context Activation Time [seconds]

Service(s) defined: All

4.6.3.4.1 Abstract definition

This parameter describes the time period needed for activating the PDP context.

4.6.3.4.2 Computation

Abstract equation:

$$\text{PDP Context Activation Time}[s] = t_{\text{PDP context activation accept}} - t_{\text{PDP context activation request}}$$

Connection to other parameters:

- Unavailability.
- Attach Failure Ratio.

Remark(s):

1) While determining the average PDP context activation time only successful activation attempts are included in the calculations.

2) 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 WAP.

3) The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached (cf. Attach Failure Ratio).

Trigger points:

- Start: Point of time when the mobile sends the PDP context activation request message.
- Stop: Point of time when the mobile receives the PDP context activation accept message.

4.6.3.5 {Service} PDP Context Cut-off Ratio [%]

Service(s) defined: All

4.6.3.5.1 Abstract definition

The PDP context cut-off ratio denotes the probability that a PDP context is deactivated without being initiated intentionally by the user.

4.6.3.5.2 Abstract equation

Abstract equation:

$$\text{PDP Context Cut}-\text{off Ratio}[\%] = \frac{\text{No. of PDP context losses not initiated by the user}}{\text{No. of all successfully activated PDP contexts}} \times 100\%$$

Trigger points: 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.

Remark(s): Precondition for measuring this parameter is that a PDP context was successfully established first.

4.6.3.6 {Service} Round Trip Time [milliseconds]

Service(s) defined: Ping
FTP (download/upload)
E-Mail POP3
E-Mail SMTP
HTTP

4.6.3.6.1 Abstract definition

The 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 already be established.

4.6.3.6.2 Computation

Abstract equation:

$$\text{Round Trip Time}[\text{ms}] = t_{\text{Packet received}} - t_{\text{Packet sent}}$$

Trigger points: Ping

- Start: Point of time when the ICMP echo request is sent ($t_{\text{ICMP echo request}}$).
- Stop: Point of time when the ICMP echo reply is received by the sender ($t_{\text{ICMP echo reply}}$).

FTP, E-Mail, HTTP

The measurement of the round trip time is done by evaluating the TCP handshake:

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

4.7 Multimedia Messaging Service (MMS)

NOTE: 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 RAS on the setup of PDP Context. (See TS 102 250-3 [5]).

4.7.1 MMS send failure ratio (MO) [%]

4.7.1.1 Abstract definition

The parameter MMS Send Failure Ratio (MO) describes the probability that a MMS-message can not be send by the subscriber, although he has requested to do so by pushing the "send button".

4.7.1.2 Computation

Trigger Points:

Event	Trigger Point	Technical description/protocol part
MMS Send Attempt (MO)	Pushing of 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).
Unsuccessful MMS Send Attempt (MO)	Do not see "Message sent"	The *m-send.conf* (see [2]) (where Response Status: $80 = M_RS_OK) is not received by the MS(MO). (See trigger 18 in figure 7). NOTE 1: The phase where the WAP session will be deactivated is not covered by this indicator. Some mobiles might not support the sending/receiving of the next MMS unless the WAP session 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. "MMS unsuccessful send attempt timeout" as specified in TS 102 250-5 (see bibliography).

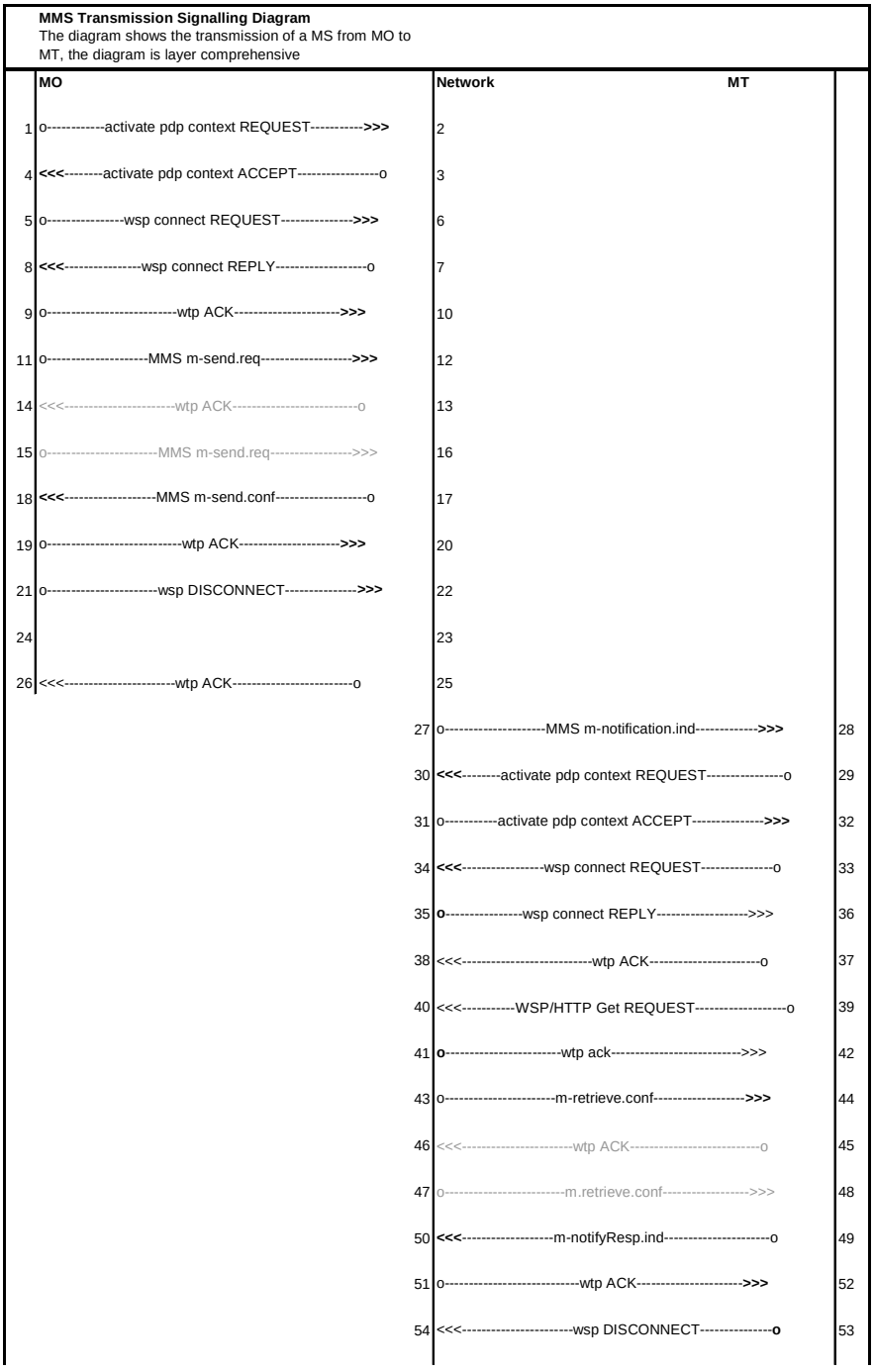


Figure 7: MMS Transaction flow

Abstract formula:

$$\text{MMS Send Failure Ratio (MO)}\,[\%] = \frac{\text{Number of unsuccessful MMS Send Attempts (MO)}}{\text{Number of All MMS Send Attempts (MO)}} \times 100\ \%$$

4.7.2 MMS retrieval failure ratio (MT) [%]

4.7.2.1 Abstract definition

The parameter MMS Retrieval Failure Ratio (MT) describes the probability that the MMS-message can not be downloaded by the MT mobile, 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 is switched to automatic mode) or enables the User to manually start this "Wap Get Request" (when the mobile is switched to manual mode). All the measurements will be done using the setting "Automatic Download".

4.7.2.2 Computation

Trigger Points:

Event	Trigger Point	Technical description/protocol part
MMS Retrieval Attempt (MT)	Initiation of the Wap Get Request MT	After the *m-Notification.ind*. (see [2]) has been sent to the MS (MT), this mobile activates a PDP-context and contacts the MMSC via the WAP Gateway (See trigger 29 in figure 7).
Unsuccessful MMS Retrieval Attempt (MT)	No MMS-message is received	The *m-notifyResp.ind* (see [2]) is not sent by the MS (MT). (See trigger 49 in figure 7). NOTE 1: The phase where the WAP session will be deactivated is not covered by this indicator. Some mobiles might not support the sending/receiving of the next MMS unless the WAP session is disconnected properly. NOTE 2: Only MMS received within the timeouts will be considered. "MMS unsuccessful Retrieval timeout" as specified in TS 102 250-5 (see bibliography).

Abstract formula:

$$\text{MMS Delivery Failure Ratio (MT)}\,[\%] = \frac{\text{Number of unsuccessful MMS Delivery Attempts MT}}{\text{Number of All MMS Delivery Attempts (MT)}} \times 100\,\%$$

4.7.3 MMS send time (MO) [s]

4.7.3.1 Abstract definition

A subscriber uses the Multimedia Messaging Service (as indicated by the network ID in his mobile phone display). The time elapsing from pushing the send button after the editing of a MMS-message to the completion of the data transfer is described by this parameter.

NOTE: 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 increase also. Number of MMSCs involved is therefore a measurement condition to be discussed.

4.7.3.2 Computation

Trigger Points:

Event	Trigger Point	Technical description/protocol part
$t_{MMStoMMSCcomplete}$	MMS-message is completely transmitted to MMS-C	The *m-send.conf* (see [2]) (where Response Status: $80 = M_RS_OK) is not received by the MS(MO). (See trigger 18 in figure 7). NOTE 1: The phase, where the WAP session will be deactivated is not covered by this indicator. Some mobiles might not support the sending/receiving of the next MMS unless the WAP session is disconnected properly. NOTE 2: Only MMS send within the timeouts will be considered.
$t_{sendbutton}$	Send button is pushed	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). "MMS unsuccessful send transfer timeout" as specified in TS 102 250-5 (see bibliography).

Abstract formula:

$$\text{MMS Send Time } [s] = t_{MMStoMMSCcomplete} - t_{sendbutton}$$

4.7.4 MMS retrieval time (MT) [s]

4.7.4.1 Abstract definition

The reception of a MMS-message works as follows: A push-sms is sent to the receiver's mobile. 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 will be described by the parameter MMS Retrieval Time (MT).

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 increase also. Number of MMSCs involved is therefore a measurement condition to be discussed.

4.7.4.2 Computation

Trigger Points:

Event	Trigger Point	Technical description/protocol part
$t_{MMSfromMMSCcomplete}$	MMS-message is received completely	The *m-notifyResp.Ind* (see [2]) is sent by the MS (MT). (See trigger 49 in figure 7). NOTE 1: The phase, where the WAP session will be deactivated is not covered by this indicator. Some mobiles might not support the sending/receiving of the next MMS unless the WAP session is disconnected properly. NOTE 2: Only MMS received within the timeouts will be considered. "MMS successful retrieval timeout" as specified in TS 102 250-5 (see bibliography).
$t_{initWGR}$	Time when WAP Get Request is initiated	The *m-Notification.ind* (see [2] is delivered to the MS (MT). This initiates the *PDP context activation.* (See trigger 29 in figure 7).

Abstract equation:

$$\text{MMS Delivery Time MT } [s] = t_{MMSfromMMSCcomplete} - t_{initWGR}$$

4.7.5 MMS notification failure ratio [%]

4.7.5.1 Abstract definition

The parameter MMS Notification Failure Ratio [%] describes the probability that the Multimedia Messaging Service (MMS) is not able to deliver the Notification of a MMS-message to the b-parties mobile.

4.7.5.2 Computation

Trigger Points:

Event	Trigger Point	Technical description/protocol part
Successful submitted MMS MO	Reception of the acknowledgement from the MMS-C MO (i.e. "Message sent")	The *m-send.conf* (see [2]) (where Response Status: $80 = M_RS_OK) is not received by the MS(MO). (See trigger 18 in figure 7). NOTE 1: The phase where the WAP session will be deactivated is not covered by this indicator. Some mobiles might not support the sending/receiving of the next MMS unless the WAP session 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 added alternatively.
Failed MMS-Notifications	Failure delivery (non-delivery) of the Notification - SMS	*m- notification.ind* (see [2]) is not delivered to the MS(MT). (See trigger 28 in figure 7). NOTE 3: Only Notifications received within the timeouts will be considered as successful. "MMS successful notification timeout" as specified in TS 102 250-5 (see bibliography).

Abstract formula:

$$\text{MMS Notification Failure Ratio}[\%] = \frac{\text{Number of failed MMS - Notifications}}{\text{Number of successful submitted MMS (MO)}} \times 100\,\%$$

4.7.6 MMS notification time [s]

4.7.6.1 Abstract definition

A subscriber uses the Multimedia Messaging Service. 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 increase also. Number of MMSCs involved is therefore a measurement condition to be discussed.

4.7.6.2 Computation

Trigger Points:

Event	Trigger Point	Technical description/protocol part
$t_{MMSsubmit}$	The MMS is submitted successfully	The *m-send.conf* (see [2]) , (where Response Status: $80 = M_RS_OK) is not received by the MS(MO). (See trigger 18 in figure 7). NOTE 1: The phase, where the WAP session will be deactivated is not covered by this indicator. Some mobiles might not support the sending/receiving of the next MMS unless the WAP session is disconnected properly. NOTE 2: The phase, where the WAP session will be deactivated is not covered by this indicator. Some mobiles might not support the sending/receiving of the next MMS unless the WAP session is disconnected properly.
$t_{recNotif}$	Time when the Notification is received (MT)	*m-Notif.ind* (see [2]) is received by MS (MT) (See trigger 28 in figure 7). NOTE 3: Only Notifications received within the timeouts will be considered as successful. "MMS successful notification timeout" as specified in TS 102 250-5 (see bibliography).

Abstract equation:

$$\text{MMS Notification Time MO/MT } [s] = t_{recNotif}[s] - t_{MMSsubmit}[s]$$

4.7.7 MMS end-to-end failure ratio [%]

4.7.7.1 Abstract definition

The parameter MMS end-to-end failure ratio describes 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.

4.7.7.2 Computation

Trigger Points:

Event	Trigger Point	Technical description/protocol part
MMS Send Attempt by MS(MO)	Pushing of send button	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). 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 [2]), (where response status is $80 = M RS OK).
Unsuccessful MMS Retrieval Attempt of MS(MT)	No MMS-message is received (MT) or no acknowledgement from the MMSC is received at MS (MO).	The *m-send.conf* (where Response Status: $80 = M_RS_OK) is not received by the MS(MO). (See trigger 18 in figure 7) or the *m-notifyResp. ind* (see [2]) (see is not sent by the MS (MT)). (See trigger 18 and 49 in figure 7). NOTE 2: The phase where the WAP session will be deactivated is not covered by this indicator. Some mobiles might not support the sending/receiving of the next MMS unless the WAP session is disconnected properly. NOTE 3: Only MMS received within the timeouts will be considered. MMS unsuccessful End-to-End timeout as specified in TS 102 250-5 (see bibliography).

Abstract equation:

$$\text{MMS End - to - end Failure Ratio}[\%] = \frac{\text{Number of unsuccessful delivered MMS - messages}}{\text{Number of all MMS send attempts}} \times 100\ \%$$

End-to-end parameter measurement may optionally be derived by concatenating the component measurements.

4.7.8 MMS End-to-end Delivery Time (MO/MT) [s]

4.7.8.1 Abstract definition

A subscriber uses the Multimedia Messaging Service (as indicated by the network ID in his 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 MO/MT.

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 sized MM is used for this measurement. See Auxiliary (Network Performance-) Parameter "MMS Average Size".

NOTE 1: 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 increase also. 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.

4.7.8.2 Computation

Trigger Points:

Event	Trigger Point	Technical description/protocol part
$t_{sendattemot}$	Time when the "send button" is pushed	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). 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).
t_{MMSrec}	Time when the MMS is received at the b-parties mobile	The M-resp.ind (see [2]) is received completely by the MS (MT), and the MS (MT) sends the m-notify-resp.ind (See trigger 49 in figure 7). 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). NOTE 3: The phase where the WAP session will be deactivated is not covered by this indicator. Some mobiles might not support the sending/receiving of the next MMS unless the WAP session is disconnected properly. NOTE 4: Only MMS received within the timeouts will be considered. "MMS successful End-to-end timeout" as specified in TS 102 250-5 (see bibliography).

Abstract equation:

$$\text{MMS End - to - end Delivery Time (MO/MT)}\ [\text{s}] = t_{\text{MMSrec}} - t_{\text{sendAttempt}}$$

4.8 Streaming

4.8.1 Definitions

4.8.1.1 Streaming Session or Session

RFC 2326 [8] 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 8 this means that the session starts at (B) and stops at (G).

4.8.2 Prerequisites

Precondition	Covered by	Reference document	Comment
Network Accessibility given	Network Accessibility Indicator		
PDP context activated			

4.8.3 Streaming Scenarios

The following two clauses describe different streaming scenarios. The first one is a generic approach in order to understand the main principles and identify the relevant protocols and communication procedures.

4.8.3.1 Generic Streaming Signalling Flow

A generic signal flow description for streaming is shown in figure 8. 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.

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

Protocol	Reference in figure 8	Description
HTTP	A	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. NOTE 1: RTSP is not responsible for the delivery of the data, this is done by RTP.
RTP	D	RTP is used for the transmission of real-time data, e.g. audio/video. NOTE 2: RTP is only used for the delivery of the data. No control and/or QoS are included.
RTCP	E	RTCP is the control protocol for RTP. Ist main function is the provision of a quality feedback.

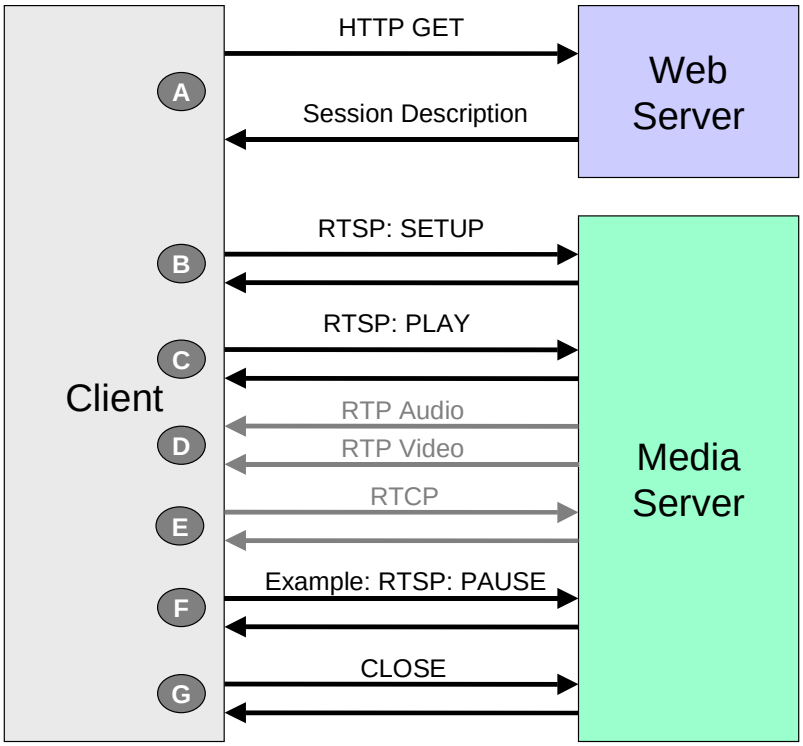


Figure 8: Generic session signalling flow, based on Schulzrinne

Referring to figure 8 and the definition of a session in clause 4.8.1.1 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 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).

4.8.3.2 Parameter Overview Chart

The following diagram gives an overview of the defined QoS parameters with their trigger points from customer's point of view.

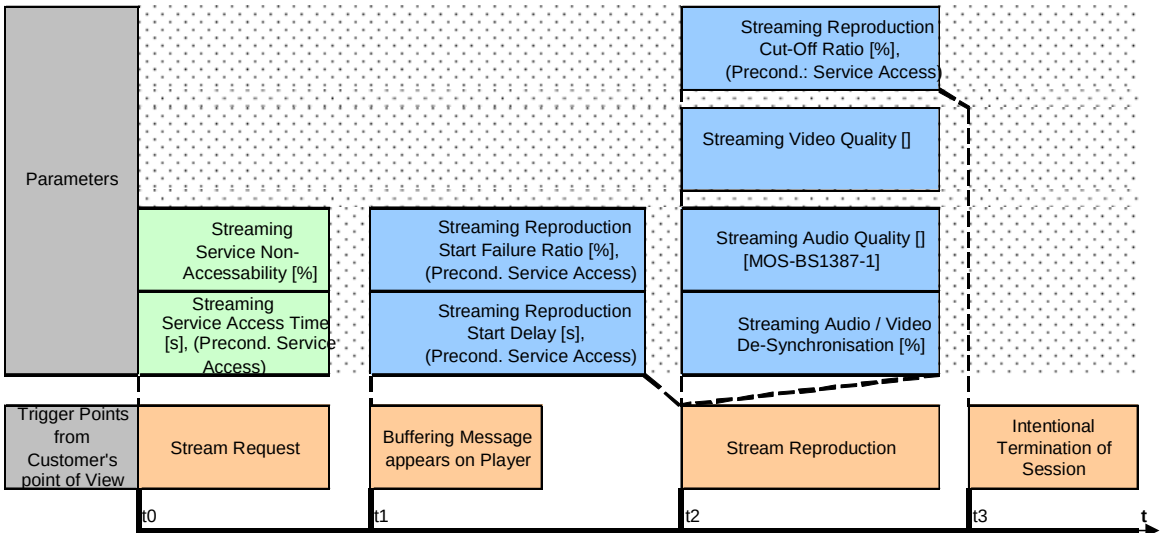


Figure 9: Parameter overview with trigger points

4.8.4 Streaming Service Non-Accessibility [%]

4.8.4.1 Abstract Definition

The parameter Streaming Service Non-Accessibility describes 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 user side.

The first data packet refers to RTP protocol.

4.8.4.2 Computation

Trigger Points:

Trigger Point from customer's point of view	Technical description/protocol part
Begin: Stream request	Start: RTSP: Setup
End: "Buffering" message	Stop: Receipt of first data packet

Abstract formula:

$$\text{Streaming Service Non - Accessibility}[\%] = \frac{\text{Number of unsucessful stream request attempts due to first data packet non - receipt}}{\text{Number of total stream request attempts}} \times 100\%$$

4.8.5 Streaming Service Access Time [s]

4.8.5.1 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.

4.8.5.2 Computation

Trigger Points:

Trigger Point from customer's point of view	Technical description/protocol part
Begin: Stream request	Start: RTSP: Setup
End: "Buffering" message	Stop: Receipt of first data packet

Abstract Formula:

$$\text{Streaming Service Access Time}[s] = [\text{Time of 1st data packet reception}] - [\text{Time of stream request}]$$

4.8.6 Streaming Reproduction Cut-off Ratio [%]

4.8.6.1 Abstract Definition

The parameter *Streaming Reproduction Cut-off Ratio* describes the probability that a successfully started stream reproduction is ended by a cause other than the intentional termination by the user.

Causes for Reproduction Cut-off

The following list represents possible causes for session cut-off scenarios:

- radio bearer loss;
- synchronization errors;
- streaming server/system failure/errors;
- protocol errors;
- streaming player failure/errors.

4.8.6.2 Computation

Trigger points:

Trigger Point from customer's point of view	Technical description/protocol part
Begin: Appearance of the buffering information (after stream request) on the player screen	Start: Receipt of 1st data packet
End: unintentional stop of stream reproduction	Stop: if RTSP:TEARDOWN method is not received

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.

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

Abstract equation:

$$\text{Streaming Reproduction Cut - off Ratio}[\%] = \frac{\text{Number of lost media streaming reproductions}}{\text{Number of all media streaming reproductions successfully started}} \times 100\%$$

4.8.7 Streaming Audio Quality

4.8.7.1 Abstract Definition

The parameter Streaming Audio Quality describes the audio quality as perceived by the end-user. Since the streams can contain and not only speech information, an algorithm like P.862 is not suitable for all scenarios.

ITU-R has defined an algorithm defined for audio information. It can be found in [6].

4.8.7.2 Computation

Trigger Points:

Trigger Point from customer's point of view	Technical description/protocol part
Begin: Begin of audio stream reproduction	Start: Streaming players signal when the reproduction of the stream starts
End: End of audio stream reproduction	Stop: RTSP: TEARDOWN

4.8.8 Streaming Video Quality

4.8.8.1 Abstract Definition

The parameter *Streaming Video Quality* measures the quality of the video stream.

NOTE 1: Although evaluation algorithms exist, there are no standardized solutions yet.

NOTE 2: Standardization process of evaluation algorithms is on-going and new Recommendations are expected during the ITU study period 2005-2008.

4.8.8.2 Computation

Trigger Points:

Trigger Point from customer's point of view	Technical description/protocol part
Begin: Begin of video stream reproduction	Start: Streaming players signal when the reproduction of the stream starts
End: End of video stream reproduction	Stop: RTSP: TEARDOWN

Abstract Formula:

NOTE 1: Although evaluation algorithms exist, there are no standardized solutions yet.

NOTE 2: Standardization process of evaluation algorithms is on-going and new Recommendations are expected during the ITU study period 2005-2008.

4.8.9 Streaming Audio/Video De-Synchronization

4.8.9.1 Abstract Definition

The parameter *Streaming Audio/Video De-Synchronization* describes the percentage of times that time difference of the audio and video signal at the user side exceeds a predefined threshold.

4.8.9.2 Computation

Trigger Points:

Trigger Point from customer's point of view	Technical description/protocol part
Begin: Begin of audio stream reproduction	Start: Streaming players signal when the reproduction of the stream starts
End: End of audio stream reproduction	Stop: RTSP: TEARDOWN

Abstract Formula:

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

4.8.10 Streaming Reproduction Start Failure Ratio [%]

4.8.10.1 Abstract Definition

The parameter *Streaming Reproduction Start Failure Ratio* describes the probability of unsuccessful stream reproduction.

NOTE: This parameter can be affected:
- by the player;
- by the UE performance.

4.8.10.2 Computation

Trigger Points:

Trigger Point from customer's point of view	Technical description/protocol part
Begin: "buffering" message	Start: Receipt of 1st data packet
End: Stream reproduction	Stop: Streaming players signal when the reproduction of the stream starts

Abstract Formula:

$$\text{Streaming Reproduction Start Failure Ratio}[\%] = \frac{\text{No. of reproduction failures}}{\text{No of all successful service accesses}} \times 100\%$$

4.8.11 Streaming Reproduction Start Delay [s]

4.8.11.1 Abstract Definition

The parameter *Streaming Reproduction Delay* describes 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.

4.8.11.2 Computation

Trigger Points:

Trigger Point from customer's point of view	Technical description/protocol part
Begin: "buffering" message	Start: Receipt of 1st data packet
End: Stream reproduction	Stop: Streaming players signal when the reproduction of the stream starts

Abstract Equation:

$$\text{Streaming Reproduction Delay}[s] = [\text{Time of stream reproduction start}][s] - [\text{Time of 1st data packet reception}][s]$$

4.9 Video Telephony

4.9.1 Network Accessibility/Availability

Network Availability and Network Accessibility are measured independently from the Service, and will not be described further in the present document. Network Availability and Network Accessibility are pre-conditions for the performance of the measurement of QoS.

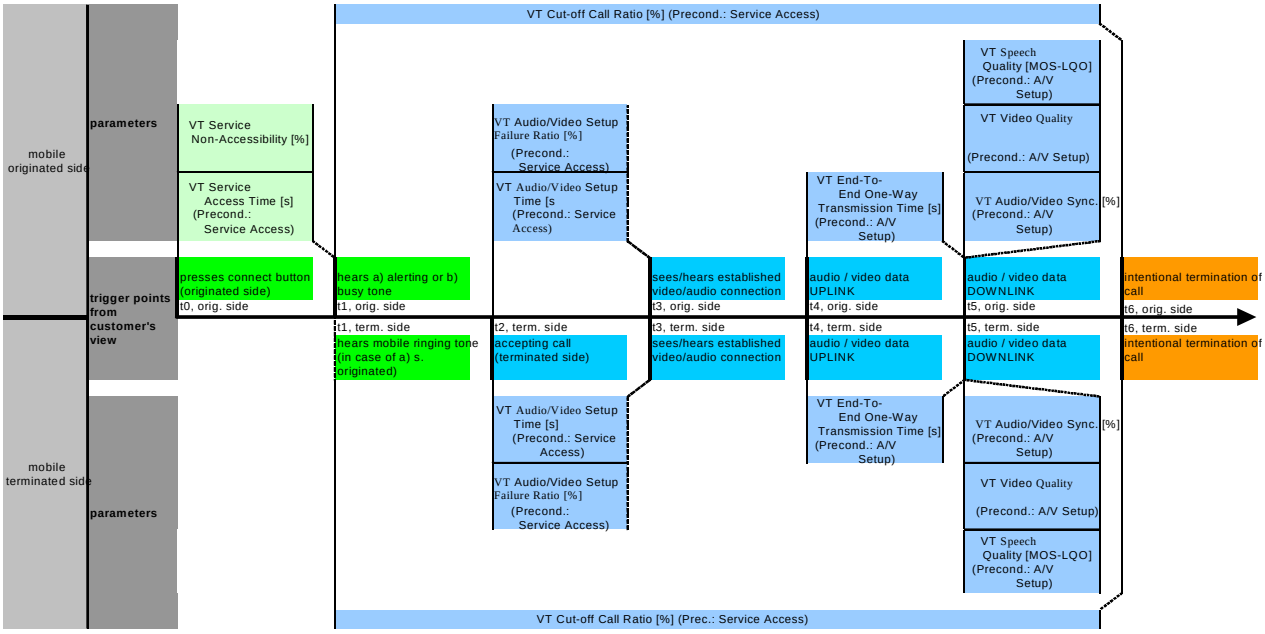
4.9.2 Parameter Overview Chart

To get a better overview of the following parameters, the diagram below shows all steps of a Video Telephony call from origin to destination, and the related QoS parameters.

Preconditions for the measurements:

It should be a bi-directional Video Telephony call. Both sides should allow the transmission of both audio and video.

Video-telephony parameters overview with trigger points



Author: Ralf Winkler, VF D2, Dep.: TGQ

Explanation: The upper half considers the triggerpoints and parameters at the originated side and the lower half at the terminated side. The rectangles are connected to the triggerpoints that are relevant for analysis. For example: "t3, orig. side" (triggerpoint at originated side) and "t3, term. side" (triggerpoint 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 behind the parameter name.

Figure 10: Parameter overview with trigger points

4.9.3 VT Service Non-Accessibility [%]

4.9.3.1 Abstract definition

Probability that the end-customer cannot access the service when requested while it is offered by network indication on the mobile equipment.

Remark(s): A successful 'Service Access' is when the A-party hears the alerting or busy tone after the send button is pushed.

4.9.3.2 Computation

Abstract formula:

$$\text{VT Service Non - Accessibility}\,[\%] = \frac{\text{Number of Unsuccessful l Video Telephony Call Access Attempts}}{\text{Number of All Video Telephony Call Access Attempts}} \text{x}\,100\%$$

Trigger Points:

Event (from equation)	Trigger Point (from customer's point of view)
Video Telephony call access attempt	Push Send button
Unsuccessful Video Telephony call access attempt	Alerting or busy tone is not heard by the A-party coming from B-party

4.9.4 VT Service Access Time [s]

4.9.4.1 Abstract definition

Time between sending of complete address information and receipt of VT call set-up notification.

Remark(s): This parameter is not calculated unless the video telephony call access attempt is successful.

4.9.4.2 Computation

Abstract formula:

$$\text{VT Service Access Time}\,[\text{s}] = t_{ConnectionEstablished} - t_{PushSendButton}$$

Trigger Points:

Event (from equation)	Trigger Point (from customer's point of view)
Video Telephony call access attempt	Push Send button
Connection established (Successful Video Telephony call attempt)	Alerting or busy tone is heard by the A-party coming from B-party

4.9.5 VT Audio/Video Setup Failure Ratio [%]

4.9.5.1 Abstract definition

Probability of audio/video setup failure after service access. The audio/video setup is successful if audio and video output is performed at both sides.

Remark(s):

- 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 used and on the multimedia protocol stack implemented (e.g. answer fast feature).

4.9.5.2 Computation

Abstract formula:

$$\text{VT Audio/Video Setup Failure Ratio}\,[\%] = \frac{\text{Number of Audio/Video Setup Failures}}{\text{Number of All Accepted Calls at MT side}} \text{ x } 100$$

Trigger Points:

Event (from equation)	Trigger Point (from customer's point of view)
Accepted call at MT side	Pressing the accept key at the MT side to accept the incoming call
Audio/Video Setup Failure	Not start of the audio and video output at both sides

4.9.6 VT Audio/Video Setup Time [s]

4.9.6.1 Abstract definition

The elapsed time from accepting the call at the MT side until audio and video output starts at both sides.

Remark(s):

- This parameter should report the worse time of both sides.
- This parameter is not calculated unless the VT audio/video setup attempt is successful.
- This parameter depends on the mobile used and on the multimedia protocol stack implemented (e.g. answer fast feature).

4.9.6.2 Computation

Abstract formula:

$$\text{VT Audio/Video Setup Time}\,[\text{s}] = t_{Audio/Videostart} - t_{MTacceptcall}$$

Trigger Points:

Event (from equation)	Trigger Point (from customer's point of view)
Accepted call at MT side	Pressing the accept key at the MT side to accept the incoming call
Audio/Video Start	Start of the audio and video output at both sides

4.9.7 VT Cut-off Call Ratio [%]

4.9.7.1 Abstract definition

Probability that a successful service access is ended by a cause other than the intentional termination of the user (calling or called party).

Remark(s): This parameter is not calculated unless the VT service access attempt is successful. A VT call is considered dropped if both audio and video are not established within the audio/video setup timeout or if either the audio, the video or both are lost for at least 10 seconds.

4.9.7.2 Computation

Abstract formula:

$$\text{VT Cut - off Call Ratio}\,[\%] = \frac{\text{Number of VT Dropped Calls}}{\text{All successful VT Call Access Attempt}} \text{ x } 100$$

Trigger Points:

Event (from equation)	Trigger Point (from customer's point of view)
Dropped Call	Lost of video and/or audio without any intention by A- or B-party
Call Access Attempt	Alerting or busy tone heard by the A-party

4.9.8 VT Speech Quality on Call Basis [MOS-LQO]

4.9.8.1 Abstract definition

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.

4.9.8.2 Computation

The validation of the end-to-end quality is made using the $MOS_{\_LQO}$ scale. This scale describes the opinion of customers with voice transmission and its troubles (noise, robot voice, echo, dropouts etc). The speech quality measurement is taken per call. An aggregation should be made on one value for speech quality per call.

Reference: ITU-T Recommendation P.862 (Algorithm) [1] in conjunction with ITU-T Recommendation P.862.1 [9].

Abstract formula:

$$\text{VT Speech Quality on Call Basis (received A - side)} = f(MOS_{\_LQO})$$
$$\text{VT Speech Quality on Call Basis(received B - side)} = f(MOS_{\_LQO})$$

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).

Trigger points:

Begin: Start of the audio and video output at both sides

End: Release of connection

NOTE 1: The acoustic behaviour of terminals is not part of this speech quality measurement.

NOTE 2: For wideband (7 kHz) applications no standardized algorithm is available yet.

4.9.9 VT Speech Quality on sample basis [MOS-LQO]

4.9.9.1 Abstract definition

Indicator representing the quantification of the end-to-end speech transmission quality of the Video Telephony Service. This parameter computes the speech quality on a sample basis.

4.9.9.2 Computation

The validation of the end-to-end quality is made using the MOS scale. This scale describes the opinion of customers with voice transmission and its troubles (noise, robot voice, echo, dropouts etc). The speech quality measurement is taken per sample. An aggregation for measurement campaigns or parts of it should be made on speech sample basis.

Reference: ITU-T Recommendation P.862 (Algorithm) [1] in conjunction with ITU-T Recommendation P.862.1 [9].

Abstract formula:

$$\text{VT Speech Quality on Sample Basis (received A - side)} = f(MOS\_LQO)$$
$$\text{VT Speech Quality on Sample Basis (received B - side)} = f(MOS\_LQO)$$

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).

Trigger points:

Begin: Start of the audio and video output at both sides

End: Release of connection

NOTE 1: The acoustic behaviour of terminals is not part of this speech quality measurement.

NOTE 2: For wideband (7 kHz) applications no standardized algorithm is available yet.

4.9.10 VT Video Quality

4.9.10.1 Abstract definition

End-to-end quality of the video signal as perceived by the end user during a VT call.

Remark(s): This parameter is not calculated unless the VT audio/video setup attempt is successful.

4.9.10.2 Computation

Abstract formula: To be specified.

Trigger Points:

Event (from equation)	Trigger Point (from customer's point of view)
Successful Audio/Video Setup Attempt	Start of the audio and video output at both sides
Release of connection	Release of connection

4.9.11 VT End-To-End Mean One-Way Transmission Time [s]

4.9.11.1 Abstract definition

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(s): This parameter is not calculated unless the VT audio/video setup attempt is successful.

4.9.11.2 Computation

Abstract formula:

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

Remark(s): This parameter is not calculated unless the VT audio/video setup attempt is successful.

Trigger Points:

Event (from equation)	Trigger Point (from customer's point of view)
	Signal in MS (MO/MT) side
	The same signal in MS (MT/MO) side

4.9.12 VT Audio/Video Synchronization [%]

4.9.12.1 Abstract definition

Percentage of times that the time differences of the audio and video signal at the user side exceeds a predefined threshold.

Remark(s):

- This parameter is not calculated unless the VT audio/video setup attempt is successful.
- Only if audio and video use different bearers this indicator would reflect the behaviour of the network and the mobiles.

4.9.12.2 Computation

Abstract formula: To be specified.

Trigger Points:

Event (from equation)	Trigger Point (from customer's point of view)
Successful Audio/Video Setup Attempt	Start of the audio and video output at both sides
Release of connection	Release of connection

Annex A (informative): Examples for measuring trigger points

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

Annex B (informative): Streaming explanations

RTP - Real Time Protocol

The Real Time Protocol 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 [7].

RTCP - Real Time Control Protocol

The Real Time Control Protocol as 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 [7].

RTSP - Real Time Streaming Protocol

The Real Time Streaming Protocol is used for the overall control of the streaming session.

For a complete description of the RTSP refer to [8].

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 [8].

SETUP:

> Causes the server to allocate resources for a stream and start an RTSP session [8].

PLAY:

> Play is send from the client to the server and informs the server to start the transmission of data as specified by the SETUP method [8].

PAUSE:

> Send from client to server. Temporarily halts the stream transmission without freeing server resources. These resources can only be freed after a specified time [8].

RECORD:

> This method initiates recording a range of media data according to the presentation description [8].

TEARDOWN:

> Frees resources associated with the stream. The RTSP session ceases to exist on the server [8].

B.1 Streaming Hyperlink Description

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

Annex C (informative): Bibliography

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