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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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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 quantitative 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 three parts: the abstract definition which contains a generic description of the parameter, the abstract equation and 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 and 3G networks, 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. These profiles are necessary for comparing "like-for-like" performance in case that a specific set of tests is carried out by different customers.

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 and 3G networks where the results are obtained applying inferential statistics.

Introduction

All the defined quality of service parameters and their computations are based on field measurements. This 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 three parts: the abstract definition which contains a generic description of the parameter, the abstract equation and 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 vary depending on the respective 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.

Other standardization bodies, namely the CBMS group in the DVB Forum and the BCAST group in OMA, requested an approved document for QoS parameters in Mobile Broadcast to be used as a reference in their documents. Therefore, the parameters described below should be approved even if technical trigger points still can not be defined in most of the cases. This is due to missing stable specifications in the mentioned bodies. If these specifications are available, the information will be added to enhance the parameter definitions given in the following document.

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 7)".

[11] ETSI TS 145 008: "Digital cellular telecommunications system (Phase 2+); Radio subsystem link control (3GPP TS 45.008 Release 6)".

[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 Release 6)".

[13] ETSI TS 123 246: "Universal Mobile Telecommunications System (UMTS); Multimedia Broadcast/Multicast Service (MBMS); Architecture and functional description (3GPP TS 23.246 Release 6)".

[14] Push to talk over Cellular (PoC) - Architecture, OMA Candidate Version 1.0.

[15] Push to talk over Cellular (PoC) - User Plane, OMA Draft Version 1.0.16.

[16] Push to talk over Cellular (PoC) - Control Plane, OMA Candidate Version 1.0.

[17] IETF RFC 3903: "A Session Initiation Protocol (SIP) Extension for Event State Publication".

[18] ITU-T Recommendation P.862.2: "Wideband extension to P.862 for the assessment of wideband telephone networks and speech codecs".

[19] ITU-T Recommendation P.862.3: "Application guide for objective quality measurement based on Recommendations P.862, P862.1 and P.862.2".

[20] ITU-T Recommendation E.800: "Terms and definitions related to quality of service and network performance including dependability".

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.

For the purposes of the present document, the terms and definitions given in the ETSI Directives and the following apply:

ad-hoc PoC group session: is a PoC session for multiple PoC users that does not involve the use or definition of a pre-arranged or chat PoC group

automatic answer: the terminal accepts the invitation automatically if resources are available

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

NOTE: We distinguish between Broadcast Bearer and Mobile Network Bearer. The latter one is synonymously referred to as Interactivity Channel.

bootstrapping: mechanism where the broadcast signal is accessed for the first time within a service usage

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

bootstrapping bearer: bearer on which the bootstrapping procedure is executed

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

NOTE: The broadcast signal is transmitted via this bearer.

chat PoC group: persistent group in which each member individually joins the PoC session i.e. the establishment of a PoC session to a chat PoC group does not result in other members of the chat PoC group being invited

chat PoC group session: PoC session established to a chat PoC group

confirmed indication: a confirmed indication is a signalling message returned by the PoC server to confirm that the PoC server, all other network elements intermediary to the PoC server and a terminating terminal are able and willing to receive media

content: 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

ESG retrieval bearer: bearer which is used to retrieve the ESG information

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

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

manual answer: the PoC user accepts the invitation manually

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

mobile network bearer: bearer provided by a mobile network operator (e.g. GSM, GPRS, UMTS, …) to establish interactivity within the Mobile Broadcast Service

on-demand session: PoC session set-up mechanism in which all media parameters are negotiated at PoC session establishment

NOTE: The on-demand sessions are defined by the OMA PoC specification as mandatory for PoC enabled user equipment, whereas pre-established sessions are defined as optional.

PoC session: this is an established connection between PoC users where the users can communicate using voice one at a time

PoC user: user of the PoC service

pre-arranged PoC group session: persistent PoC session that has an associated set of PoC members

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

pre-established session: SIP session established between the terminal and the PoC server that performs the participating PoC function

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

service provider: the operating company of a PLMN

service user: the end customer who uses the services of a PLMN by means of a UE, e.g. a mobile phone or data card

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

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

TBCP Talk Burst Granted: used by the PoC server to notify the terminal that it has been granted permission to send a talk burst

NOTE: Cf. [14] for possible floor states.

TBCP Talk Burst Idle: used by the PoC server to notify all terminals that no one has the permission to send a talk burst at the moment and that it may accept the "TBCP Talk Burst Request" message

NOTE: Cf. [14] for possible floor states.

TBCP Talk Burst Request: used by the terminal to request permission from the PoC server to send a talk burst

NOTE: Cf. [14] for possible floor states.

terminal: shall be used when referring to a PoC enabled user equipment which is a user equipment implementing a PoC client

NOTE: Cf. [13].

unconfirmed indication: indication returned by the PoC server to confirm that it is able to receive media and believes the terminal is able to accept media

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

3.2 Abbreviations

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

3G	3rd Generation
3GPP	Third Generation Partnership Project
ATD	ATtention Dial
CCCH	Common Control CHannel
CRLF	Carriage Return Line Feed
CS	Circuit Switched
CUT	PoC Session CUT-off (PoC)
DCCH	Dedicated Control CHannel
DCH	Data CHannel
DELAY	Talk Burst DELAY (PoC)
DeREG	PoC DeREGistration (PoC)
DROP	Talk Burst DROP (PoC)
ESG	Electronic Service Guide
FTP	File Transfer Protocol
GGSN	Gateway GPRS Support Node
GPRS	General Packet Radio Service
GSM	Global System for Mobile communications
HLR	Home Location Register
HTTP	HyperText Transfer Protocol
ICMP	Internet Control Message Protocol
INIT	PoC Session INITiation
IP	Internet Protocol
LEAVE	PoC Session LEAVing
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
OMA	Open Mobile Alliance
PDP	Packet Data Protocol
PEP	Performance Enhancement Proxy

PLMN	Public Land Mobile Network
POP3	Post Office Protocol version 3
PS	Packet Switched
PtS	Push to Speech
PUB	PoC PUBlish
QoS	Quality of Service
RACH	Random Access CHannel
RAS	Remote Access Service
REG long	PoC REGistration and Publish
REG	PoC REGistration
RRC	Radio Resource Control
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
RTSP	Real Time Streaming Protocol
SDCCH	Stand-alone Dedicated Control CHannel
SDP	Session Description Protocol
SGSN	Serving GPRS Support Node
SIP	Session Initiation Protocol
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
UE	User Equipment
UMTS	Universal Mobile Telecommunications System
VT	Video Telephony
WAP	Wireless Application Protocol
WGR	WAP Get Request
WSP	Wireless Session Protocol

4 QoS Parameter Basics

4.1 General Overview

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

The first layer is the Network Availability, which defines QoS rather from the viewpoint of the service provider than the service user. The second layer is the Network Access. From the service user's point of view this is the basic requirement for all the other QoS aspects and parameters. The third layer contains the other three QoS aspects Service Access, Service Integrity and Service Retainability. The different services are located in the fourth layer. Their outcome are the QoS parameters.

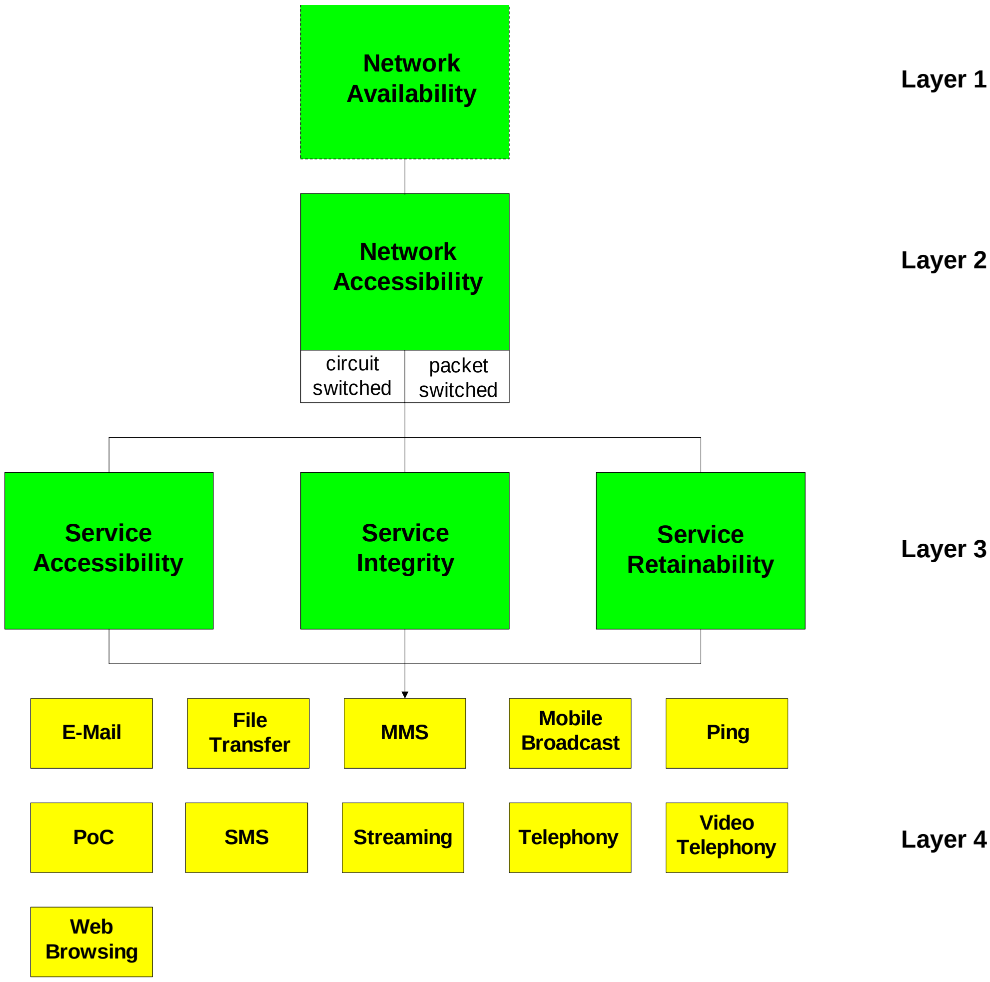


Figure 1: QoS aspects and the corresponding QoS parameters

4.2 FTP, HTTP and E-Mail Issues

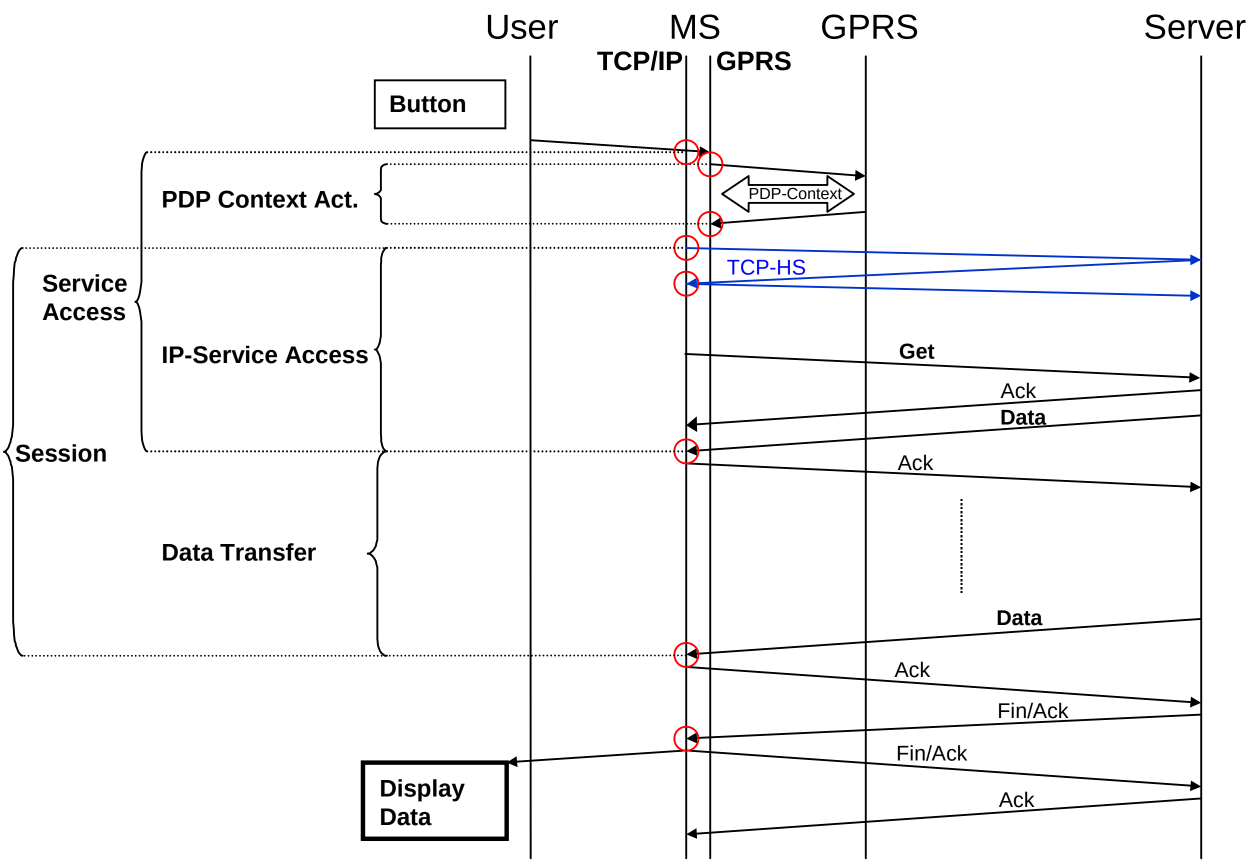
Currently two main views about the best way to reflect the user's experience for these services are in place: One preferring the payload throughput philosophy and the other preferring the transaction throughput philosophy:

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

An example of the different trigger points defined for each set is illustrated in figure 2 and figure 3: The start trigger point for the Mean Data Rate for Web browsing is either the reception of the first packet containing data content (Method A) or the sending of the HTTP GET command (Method B).

A field test system compliant to 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 that covers the attach and PDP context activation procedure. Field test systems shall be able to measure these QoS indicators.



Example: HTTP via GPRS

Figure 2: Key Performance Indicators Version A ()

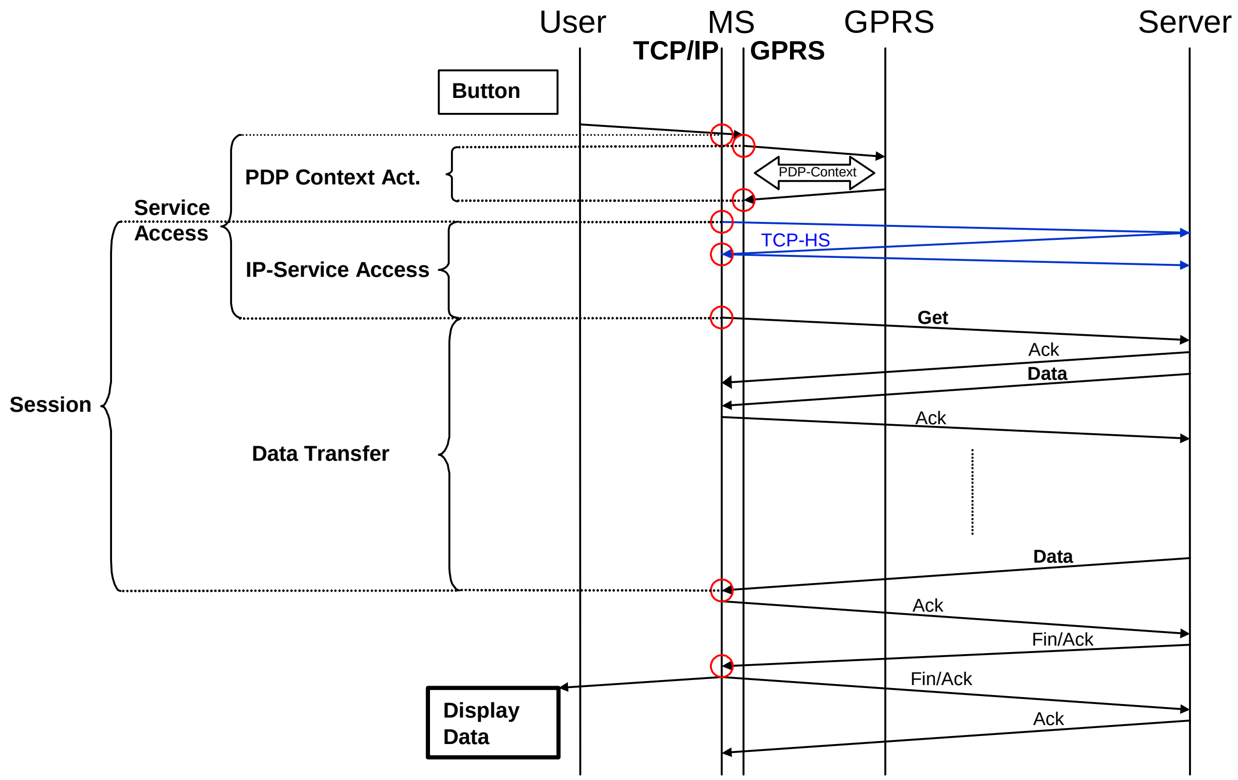


Figure 3: Key Performance Indicators Version B (Example: HTTP via GPRS)

4.2.1 Performance Enhancement Proxies

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

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

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

The following guidelines apply whenever Performance Enhancement Proxies are employed:

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

5 Service independent QoS Parameters

5.1 Radio Network Unavailability [%]

5.1.1 Abstract Definition

Probability that the mobile services are not offered to a user.

5.1.2 Abstract Equation

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

5.1.3 Trigger Points

GSM:

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

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

GPRS:

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

UMTS PS: To be defined.

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

5.2 Network Non-Accessibility [%]

5.2.1 Abstract Definition

Probability that the user cannot perform a successful registration on the PLMN.

5.2.2 Abstract Equation

$$\text{Network Non-Accessibility [\%]} = \frac{\text{unsuccessful registrations on the PLMN}}{\text{all registration attempts}} \times 100\ \%$$

5.2.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Registration attempt	Start: User turns UE on.	Start: Not assignable.
Successful registration	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.	

NOTE: 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 Non-Accessibility is checked once at the start of a probing cycle (e.g. with the AT command "at+creg?;+cops?").

5.3 Attach Failure Ratio [%]

5.3.1 Abstract Definition

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

5.3.2 Abstract Equation

$$\text{Attach Failure Ratio [\%]} = \frac{\text{unsuccessful attach attempts}}{\text{all attach attempts}} \times 100\ \%$$

5.3.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Attach attempt	Start: Customer turns the UE on.	Start: UE sends the attach request message.
Successful attach	Stop: Attach logo appears in the display of the UE.	Stop: UE receives the attach accept message.
Unsuccessful attach	Stop trigger point not reached.	

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. Radio Network Availability).

5.4 Attach Setup Time [s]

5.4.1 Abstract Definition

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

5.4.2 Abstract Equation

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

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. Radio Network Availability).

5.4.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time of attach request	Start: Customer turns the UE on.	Start: Point of time when the UE sends the attach request message.
Time when attach complete	Stop: Attach logo appears in the display of the UE.	Stop: Point of time when the UE receives the attach accept message.

5.5 PDP Context Activation Failure Ratio [%]

5.5.1 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.

5.5.2 Abstract Equation

$$\boxed{\text{PDP Context Activation Failure Ratio [\%]} = \frac{\text{unsuccessful PDP context activation attempts}}{\text{all PDP context activation attempts}} \times 100\%}$$

5.5.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates the service access.	Start: UE sends the first "Activate PDP context Request" message (Layer 3).
Successful attempt	Stop: PDP context logo appears in the display of the UE.	Stop: UE receives the "Activate PDP context Accept" message (Layer 3).
Unsuccessful attempt	Stop trigger point not reached.	

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. Radio Network Availability) and the mobile station has to be attached (cf. Attach Failure Ratio).

5.6 PDP Context Activation Time [s]

5.6.1 Abstract Definition

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

5.6.2 Abstract Equation

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

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. Radio Network Availability) and the mobile station has to be attached (cf. Attach Failure Ratio).

5.6.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time of PDP context activation request	Start: Customer initiates the service access.	Start: Point of time when the UE sends the "Activate PDP context Request" message (Layer 3).
Time when PDP context activation complete	Stop: PDP context logo appears in the display of the UE.	Stop: Point of time when the UE receives the "Activate PDP context Accept" message (Layer 3).

5.7 PDP Context Cut-off Ratio [%]

5.7.1 Abstract Definition

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

5.7.2 Abstract Equation

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

5.7.3 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.

5.8 Data Call Access Failure Ratio [%]

5.8.1 Abstract Definition

A subscriber (A-party) wants to take advantage of a given service offering (as shown by the network ID in the display of his user equipment) and establish a data call to a B-party. The failure of the data call access from imitating the data call to alerting or a busy signal is covered by this parameter.

5.8.2 Abstract Equation

$$\text{Data Call Access Failure Ratio } [\%] = \frac{\text{unsuccessf ull data call accesses}}{\text{all data call access attempts}} \times 100\ \%$$

5.8.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Data call access attempt	Start: CONNECT button is pressed.	Start: A CHANNEL_REQUEST message is sent over the RACH.
Successful data call access	Stop: Alert or busy signal occurs / connection established.	Stop: Reception of CONNECT ACK at the A-party.
Unsuccessful data call access	Stop trigger point not reached.	

5.9 Data Call Access Time [s]

5.9.1 Abstract Definition

A subscriber (A-party) wants to take advantage of a given service offering (as shown by the network ID in the display of his user equipment) and establish a data call to a B-party. The time elapsing from imitating the data call to alerting or a busy signal is covered by this parameter. This parameter is not calculated unless the call attempt is successful and not cut off beforehand.

5.9.2 Abstract Equation

$$\text{Data Call Access Time [s]} = t_{\text{successful call access}} - t_{\text{initiation of data call}}$$

5.9.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time of initiation of data call	Start: Time at which CONNECT button is pressed.	Start: A CHANNEL_REQUEST message is sent over the RACH.
Time of successful data call access	Stop: Time at which alert or busy signal occurs / connection established.	Stop: Reception of CONNECT ACK at the A-party.

6 Direct Services QoS Parameters

6.1 File Transfer Protocol(FTP)

6.1.1 FTP {Download|Upload} Service Non-Accessibility [%]

6.1.1.1 Abstract Definition

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

6.1.1.2 Abstract Equation

$$\text{FTP \{Download | Upload\} Service Non-Accessibility [\%]} = \frac{\text{unsuccessful attempts to reach the point when content is sent or received}}{\text{all attempts to reach the point when content is sent or received}} \times 100\ \%$$

6.1.1.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates the service access.	Start: ATD command.
Successful attempt	Stop: First content is received.	Stop Method A: Reception of the first data packet containing content. Stop Method B: Reception of the [ACK] from the [SYN, ACK].
Unsuccessful attempt	Stop trigger point not reached.	

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates the service access.	Start: ATD command.
Successful attempt	Stop: First content is sent.	Stop Method A: Sending of the first data packet containing content. Stop Method B: Reception of the [ACK] from the [SYN, ACK].
Unsuccessful attempt	Stop trigger point not reached.	

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

6.1.2 FTP {Download|Upload} Setup Time [s]

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.

6.1.2.2 Abstract Equation

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

6.1.2.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates the service access.	Start: ATD command.
Time when content is received	Stop: First content is received.	Stop Method A: Reception of the first data packet containing content. Stop Method B: Reception of the [ACK] from the [SYN, ACK].

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates the service access.	Start: ATD command.
Time when content is sent	Stop: First content is sent.	Stop Method A: Sending of the first data packet containing content. Stop Method B: Reception of the [ACK] from the [SYN, ACK].

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

6.1.3 FTP {Download|Upload} IP-Service Access Failure Ratio [%]

6.1.3.1 Abstract Definition

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

6.1.3.2 Abstract Equation

$$\text{FTP}\,\{\text{Download}\,|\,\text{Upload}\}\ \text{IP-Service Access Failure Ratio}\,[\%] = \frac{\text{unsuccessful attempts to establish an IP connection to the server}}{\text{all attempts to establish an IP connection to the server}} \times 100\,\%$$

6.1.3.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates file download.	Start: First [SYN] sent.
Successful attempt	Stop: File download starts.	Stop Method A: Reception of the first data packet containing content. Stop Method B: Reception of the [ACK] from the [SYN, ACK].
Unsuccessful attempt	Stop trigger point not reached.	

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates file upload.	Start: First [SYN] sent.
Successful attempt	Stop: File upload starts.	Stop Method A: Sending of the first data packet containing content. Stop Method B: Reception of the [ACK] from the [SYN, ACK].
Unsuccessful attempt	Stop trigger point not reached.	

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Radio Network 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).

6.1.4 FTP {Download|Upload} IP-Service Setup Time [s]

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.

6.1.4.2 Abstract Equation

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

6.1.4.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates file download.	Start: First [SYN] sent.
Time when content is received	Stop: File download starts.	Stop Method A: Reception of the first data packet containing content. Stop Method B: Reception of the [ACK] from the [SYN, ACK].

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates file upload.	Start: First [SYN] sent.
Time when content is sent	Stop: File upload starts.	Stop Method A: Sending of the first data packet containing content. Stop Method B: Reception of the [ACK] from the [SYN, ACK].

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Radio Network 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).

6.1.5 FTP {Download|Upload} Session Failure Ratio [%]

6.1.5.1 Abstract Definition

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

6.1.5.2 Abstract Equation

$$\text{FTP \{Download|Upload\} Session Failure Ratio [\%]} = \frac{\text{uncompleted sessions}}{\text{successfully started sessions}} \times 100\ \%$$

6.1.5.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates file download.	Start: First [SYN] sent.
Successful attempt	Stop: File download successfully completed.	Stop: Reception of the last data packet containing content.
Unsuccessful attempt	Stop trigger point not reached.	

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates file upload.	Start: First [SYN] sent.
Successful attempt	Stop: File upload successfully completed.	Stop: Reception of the [FIN, ACK] for the last data packet containing content.
Unsuccessful attempt	Stop trigger point not reached.	

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Radio Network 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).

6.1.6 FTP {Download|Upload} Session Time [s]

6.1.6.1 Abstract Definition

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

6.1.6.2 Abstract Equation

$$\text{FTP}\,\{\text{Download}\,|\,\text{Upload}\}\,\text{Session Time}\,[s] = t_{\text{Session end}} - t_{\text{Session start}}$$

6.1.6.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates file download.	Start: First [SYN] sent.
Time when content is received	Stop: File download successfully completed.	Stop: Reception of the last data packet containing content.

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates file upload.	Start: First [SYN] sent.
Time when content is sent	Stop: File upload successfully completed.	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. Radio Network 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).

6.1.7 FTP {Download|Upload} Mean Data Rate [kbit/s]

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.

6.1.7.2 Abstract Equation

$$\text{FTP}\,\{\text{Download}\,|\,\text{Upload}\}\,\text{Mean Data Rate}\,[\%] = \frac{\text{User data transferred}\,[\text{kbit}]}{\left(t_{\text{content sent or received}} - t_{\text{dial-up connection initiated}}\right)[s]} \times 100\%$$

6.1.7.3 Trigger Points

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

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: File download starts.	Start Method A: Reception of the first data packet containing content. Start Method B: Reception of the [ACK] from the [SYN, ACK].
Time when content is received	Stop: File download successfully completed.	Stop: Reception of the last data packet containing content.

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: File upload starts.	Start Method A: Sending of the first data packet containing content. Start Method B: Reception of the [ACK] from the [SYN, ACK].
Time when content is sent	Stop: File upload successfully completed.	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).

6.1.8 FTP {Download|Upload} Data Transfer Cut-off Ratio [%]

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.

6.1.8.2 Abstract Equation

$$\text{FTP \{Download | Upload\} Data Transfer Cut-off Ratio [\%]} = \frac{\text{incomplete data transfers}}{\text{successfully started data transfers}} \times 100\ \%$$

6.1.8.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: File download starts.	Start Method A: Reception of the first data packet containing content. Start Method B: Reception of the [ACK] from the [SYN, ACK].
Time when content is received	Stop: File download successfully completed.	Stop: Reception of the last data packet containing content.

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: File upload starts.	Start Method A: Sending of the first data packet containing content. Start Method B: Reception of the [ACK] from the [SYN, ACK].
Time when content is sent	Stop: File upload successfully completed.	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).

6.2 Mobile Broadcast

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

Figure 1 depicts the basis for a generic service concept for mobile broadcast. As being a composite service, two different bearers may be involved in mobile broadcast services. Unidirectional broadcast information is transmitted over the broadcast channel, whereas interactive procedures are related to the interactivity channel provided by a mobile network. The independent procedures at both bearers may interact with each other and build a common end-to-end procedure.

In general, this concept is not dedicated to specific bearer technologies. Different bearer technologies and their combinations are thinkable of.

Remark(s): However, the concept depends for example on the implementation of the Electronic Service Guide (ESG). If the ESG implementation does not allow the user to recognize the reception of ESG information, the according parameters shall have to be adapted.

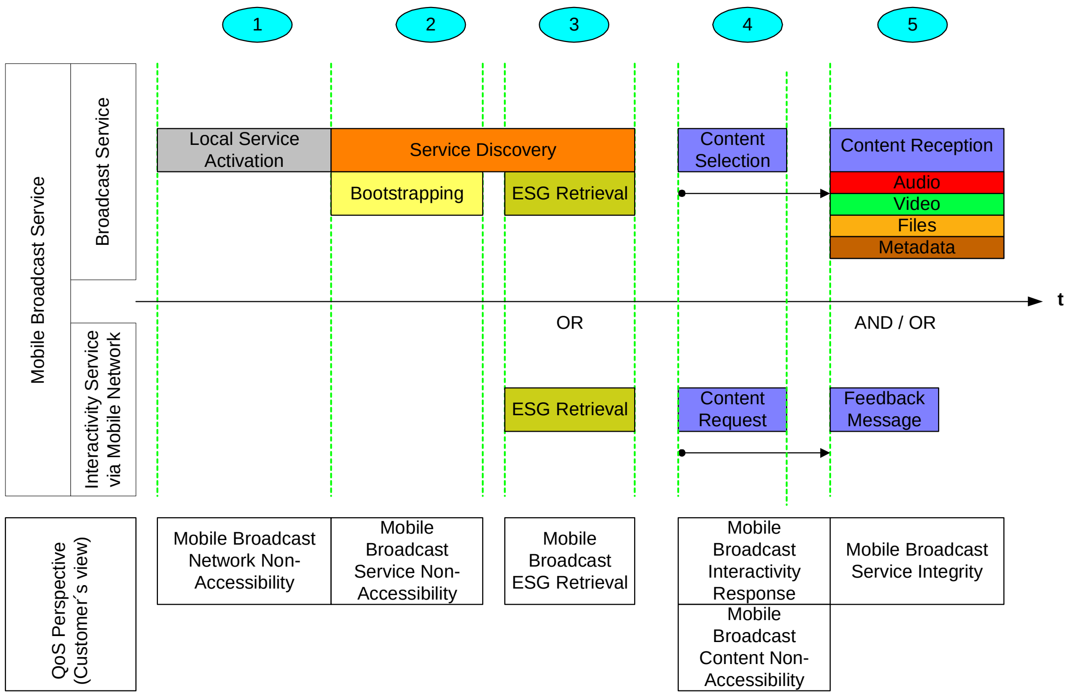


Figure 4: Service phases of mobile broadcast

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

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

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

- Mobile Broadcast Network Non-Accessibility.
- Mobile Broadcast Service Discovery.
- Mobile Broadcast Interactivity Response.
- Mobile Broadcast Service Integrity.

The mentioned phases are covered by the parameters described subsequently.

6.2.1 Mobile Broadcast Network Non-Availability {Broadcast Bearer}

6.2.1.1 Abstract Definition

Probability that the Mobile Broadcast Services are not offered to an end-customer by the target network indicators on the User Equipment (UE) in idle mode.

6.2.1.2 Abstract Equation

$$\text{Mobile Broadcast Network Non-Accessibility } [\%] = \frac{\text{unseccessful Mobile Broadcast registration attempt}}{\text{Mobile Broadcast registration attempts}} \cdot 100\ \%$$

6.2.1.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Mobile Broadcast registration attempt	Start: Start of registration procedure performed by the UE.	Tbd
Unsuccessful Mobile Broadcast registration attempt	Stop: "Mobile Broadcast icon", which indicates successfully registration, is not displayed on the UE.	Tbd

Preconditions for measurement:

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

6.2.2 Mobile Broadcast Service Discovery Failure Ratio {Bootstrapping Bearer, ESG Retrieval Bearer}

6.2.2.1 Abstract Definition

Probability that the Mobile Broadcast Services are accessible by the user. The Service Discovery mechanism, responsible among others for proving the availability of services, plays a key role. The Service Discovery procedure can be divided in two phases: bootstrapping and ESG retrieval, which are dealt in following parameters and which may use different bearers.

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

6.2.2.2 Abstract Equation

$$\text{Mobile Broadcast Service Discovery Failure Ratio } [\%] = \frac{\text{unsuccessfull Mobile Broadcast Session start sttempts}}{\text{Mobile Broadcast Session start attempts}} \cdot 100 \ \%$$

6.2.2.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Mobile Broadcast session start / Service Discovery attempt	Start: Request to use the Mobile Broadcast service on the UE.	Start: DVB-H: Tbd MBMS: MBMS Broadcast Service Activation procedure (clause 8.12 of [13])
Unsuccessful Mobile Broadcast session start /Service Discovery attempt	Stop: Mobile Broadcast service is not available on the UE.	Tbd

Preconditions for measurement:

- Mobile Broadcast Network Availability must be given.

6.2.3 Mobile Broadcast Service Discovery Time {Bootstrapping Bearer, ESG Retrieval Bearer}

6.2.3.1 Abstract Definition

The parameter Mobile Broadcast Initial Setup Time is the time period elapsed between a session start attempt of the Mobile Broadcast service and the available indication in the UE that grants access to the service. Hereby, the time the device requires to discover the available services for the first time is considered.

6.2.3.2 Abstract Equation

$$\text{Mobile Broadcast Service Discovery Time}\,[s] = t_{MB\,SessionStart} - t_{MB\,StartAttempt}$$

6.2.3.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Mobile Broadcast session start / Service Discovery attempt $t_{MB\,StartAttempt}$	Start: First request of the Mobile Broadcast service on the UE.	Start: DVB-H: Tbd MBMS: MBMS Broadcast Service Activation procedure (clause 8.12 of [16]). Tbd
Mobile Broadcast session started /Successful Service Discovery $t_{MB\,SessionStart}$	Stop: Mobile Broadcast availability is given within a pre-determined time.	Tbd

Preconditions for measurement:

- Mobile Broadcast Network Availability must be given.

6.2.4 Mobile Broadcast Bootstrapping Failure Ratio {Broadcast Bearer}

6.2.4.1 Abstract Definition

Probability that the first access to the broadcast signal fails. The failure may occur during synchronization to or decoding of the broadcast signal.

6.2.4.2 Abstract Equation

$$\text{Mobile Broadcast Bootstrapping Failure Ratio } [\%] = \frac{\text{unsuccessfull Bootstrapping procedure}}{\text{Mobile Broadcast session start / Bootstrapping attempt}} \cdot 100 \;\; \%$$

6.2.4.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Mobile Broadcast session start / Bootstrapping attempt	Start: Request to use the Mobile Broadcast service on the UE.	Tbd
Unsuccessful Mobile Broadcast session start / Bootstrapping attempt	Stop: May not be perceivable by the user.	Tbd

Preconditions for measurement:

- Mobile Broadcast Network Availability must be given.

6.2.5 Mobile Broadcast Bootstrapping Time {Broadcast Bearer}

6.2.5.1 Abstract Definition

The parameter Mobile Broadcast Bootstrapping Time measures the time it takes to perform the bootstrapping procedure consisting of the phases synchronization to the broadcast signal and broadcast signal decoding.

6.2.5.2 Abstract Equation

$$\boxed{\text{Mobile Broadcast Bootstrapping Time}\,[\text{s}] = t_{Bootstrapping\_end} - t_{Bootstrapping\_start}}$$

6.2.5.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Mobile Broadcast session start / Bootstrapping attempt $t_{Bootstrapping\_Start}$	Start: Request to use the Mobile Broadcast service on the UE.	Tbd
Successful Mobile Broadcast session start / Bootstrapping attempt $t_{Bootstrapping\_end}$	Stop: May not be perceivable by the user.	Tbd

Preconditions for measurement:

- Mobile Broadcast Bootstrapping procedure must be successful.

6.2.6 Mobile Broadcast ESG Retrieval Failure Ratio {Bearer}

6.2.6.1 Abstract Definition

Probability that the retrieval of ESG information fails. The acquisition and filtering of the Electronic Service Guide (ESG) information, as part of the Service Discovery mechanism, plays a key role.

6.2.6.2 Abstract Equation

$$\boxed{\text{Mobile Broadcast ESG Retrieval Failure Ratio}\,[\%] = \frac{\text{unsuccessfull reception of last ESG data packet}}{\text{Reception of first ESG data packet}} \times 100\ \%}$$

6.2.6.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Reception of first ESG data packet	Start: Indication of ESG retrieval.	Tbd
Unsuccessful reception of last ESG data packet needed to acquire a service	Stop: Missing indication of termination of ESG retrieval.	Tbd

Preconditions for measurement:

- For broadcast bearer:
  - Mobile Broadcast Network Availability must be given.
  - Mobile Broadcast Bootstrapping must be successful.
- For mobile network bearer:
  - Mobile Network Availability must be given.
  - Mobile Network Service Accessibility for circuit switched or packet switched data services must be given.

6.2.7 Mobile Broadcast ESG Retrieval Time {Bearer}

6.2.7.1 Abstract Definition

The Mobile Broadcast ESG Retrieval Time is the time elapsed between the start of the ESG retrieval and the successful reception of the necessary ESG information required to acquire a service. The acquisition and filtering of the Electronic Service Guide (ESG) information, as part of the Service Discovery mechanism, plays a key role.

6.2.7.2 Abstract Equation

$$\boxed{\text{Mobile Broadcast ESG Retrieval Time}\,[\text{s}] = t_{ESG\,RetrievalTermination} - t_{ESG\,RetrievalStart}}$$

6.2.7.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Reception of first ESG data packet $t_{ESG\,Retrieval\,Start}$	Start: Indication of ESG retrieval	Tbd
Reception of last ESG data packet needed to acquire a service $t_{ESG\,RetrievalTermination}$	Stop: Indication of termination of ESG retrieval	Tbd

Preconditions for measurement:

- For broadcast bearer:
  - Mobile Broadcast Network Availability must be given.
  - Mobile Broadcast Bootstrapping must be successful.
- For mobile network bearer:
  - Mobile Network Availability must be given.
  - Mobile Network Service Accessibility for circuit switched or packet switched data services must be given.
  - ESG retrieval successfully started.

6.2.8 Mobile Broadcast Content Non-Accessibility {Broadcast Bearer}

6.2.8.1 Abstract Definition

Probability that the requested Mobile Broadcast content (e.g. audio / video streaming data, files, metadata) is not started to be delivered to the user. This parameter applies also to zapping situations in which the customer changes the offered streaming content frequently in short intervals.

6.2.8.2 Abstract Equation

$$\text{Mobile Broadcast Content Selection Non-Accessibility } [\%] = \frac{\text{unsuccessful reception of first data packet}}{\text{User's / client's content request attempts}} \cdot 100 \quad \%$$

6.2.8.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
User's / client's content request attempt	Start: Request button pressed by user / request attempt from device	Start: Content Request on Interactivity Channel: Trigger points are chosen according to parameter definitions for SMS, MMS, GPRS and PS-UMTS in [TS 102 250-2].Tbd
Unsuccessful reception of first content data packet	Stop: Missing indication of reception of first content data packet	Stop: Content Request on Interactivity Channel: Trigger points are chosen according to parameter definitions for SMS, MMS, GPRS and PS-UMTS in [TS 102 250-2].Tbd

Preconditions for measurement:

- Mobile Broadcast Network Availability must be given.
- Mobile Broadcast Service Discovery must be successful.

6.2.9 Mobile Broadcast Content Access Time {Broadcast Bearer}

6.2.9.1 Abstract Definition

The parameter Mobile Broadcast Content Access Time is the time period elapsed between the user's request to receive content and the reception of the first data packet.

6.2.9.2 Abstract Equation

$$\text{Mobile Broadcast Content Reception Time } [s] = t_{ContentDeliveryStart} - t_{ContentRequestAttempt}$$

6.2.9.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
User's/client's content request attempt $t_{ContentRequestAttempt}$	Start: Request button pressed by user	Start: Content Request on Interactivity Channel: Trigger points are chosen according to parameter definitions for SMS, MMS, GPRS and PS-UMTS in [TS 102 250-2]. Tbd
Reception of first content data packet $t_{ContentDeliveryStart}$	Stop: Indication of reception of first content data packet	Stop: Content Request on Interactivity Channel: Trigger points are chosen according to parameter definitions for SMS, MMS, GPRS and PS-UMTS in [TS 102 250-2]. Tbd

Preconditions for measurement:

- Mobile Broadcast Network Availability must be given.
- Mobile Broadcast Service Discovery must be successful.

6.2.10 Mobile Broadcast Interactivity Response Failure Ratio {Mobile Network Bearer} {Broadcast Bearer}

6.2.10.1 Abstract Definition

The Mobile Broadcast Interactivity Response Failure Ratio measures the probability that a service request of a Mobile Broadcast service via an interactive channel does not result in an expected reaction (i.e., changes in content updated due to user's interaction, reception of any kind of notification to the user, …) on either the broadcast bearer or the mobile network bearer.

6.2.10.2 Abstract Equation

$$\text{Mobile Broadcast Interactivity Response Failure Ratio }[\%] = \frac{\text{unsuccessful Mobile Broadcast service outcome/response}}{\text{Mobile Broadcast service requests over interactive channel}} \times 100\%$$

6.2.10.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Mobile Broadcast service request over the interactive channel	Start: Request of the Mobile Broadcast service on the UE	Start: Content Request on Interactivity Channel: Trigger points are chosen according to parameter definitions for SMS, MMS, GPRS and PS-UMTS in [TS 102 250-2].Tbd
Unsuccessful Mobile Broadcast service outcome/response	Stop: User's interactivity is not reflected in updated content or indicated at the device	Stop: Tbd. Negative result code or timeout related to: Interactivity Channel: Trigger points are chosen according to parameter definitions for SMS, MMS, GPRS and PS-UMTS in [TS 102 250-2]. Broadcast Bearer: DVB-H: Tbd MBMS: Tbd

Preconditions for measurement:

- For broadcast bearer:
  - Mobile Broadcast Network Availability must be given.
  - Mobile Broadcast Bootstrapping must be successful.
- For mobile network bearer:
  - Mobile Network Availability must be given.
  - Mobile Network Service Accessibility for circuit switched or packet switched data services must be given.

6.2.11 Mobile Broadcast Interactivity Response Time {Mobile Network Bearer} {Broadcast Bearer}

6.2.11.1 Abstract Definition

The parameter Mobile Broadcast Interactivity Response Time is the time elapsed between a service request attempt of the Mobile Broadcast service via an interactive channel and the reception of a notification to the user.

6.2.11.2 Abstract Equation

$$\boxed{\text{Mobile Broadcast Interactivity Response Time}\,[s] = t_{MB\,Service\,Response} - t_{MBService\,Request}}$$

6.2.11.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Mobile Broadcast service request over the interactive channel $t_{MB\,Service\,Request}$	Start: Request of the Mobile Broadcast service on the UE	Start: Content Request on Interactivity Channel: Trigger points are chosen according to parameter definitions for SMS, MMS, GPRS and PS-UMTS in [TS 102 250-2] Tbd.
Successful Mobile Broadcast service outcome/response $t_{MB\,Service\,Response}$	Stop: User's interactivity is not reflected in updated content or indicated at the device	Stop: Negative result code or timeout related to: Interactivity Channel: Trigger points are chosen according to parameter definitions for SMS, MMS, GPRS and PS-UMTS in [TS 102 250-2]. Broadcast Bearer: DVB-H: Tbd MBMS: Tbd

Preconditions for measurement:

- For broadcast bearer:
  - Mobile Broadcast Network Availability must be given.
  - Mobile Broadcast Bootstrapping must be successful.
- For mobile network bearer:
  - Mobile Network Availability must be given.
  - Mobile Network Service Accessibility for circuit switched or packet switched data services must be given.

6.2.12 Mobile Broadcast Service Integrity {Broadcast Bearer}

Mobile Broadcast technology paves the way for network operators and service providers to offer a huge palette of mobile services, which can be divided in the following categories:

- Streaming services.
- Packet switched data services.
- Short Message Service (SMS).
- Multimedia Message Service (MMS).
- Wireless Application Protocol (WAP).

According to ITU-T Recommendation E.800 [20], the Service Integrity describes the Quality of Service during service use. Since the above mentioned services are already offered in other scenarios, in the present document only a reference to the already defined QoS parameters will be made. Important to bear in mind is the fact that for Mobile Broadcast Service, only the abstract definition of the parameters applies, since the underlying protocol stack may not be the same.

The ETSI document [TS 102 250-2] defines the following QoS parameters:

- For packet switched data services:
  - Completed Session Ratio.
  - Mean Data Rate.
  - Round Trip Time.
- For streaming services:
  - Streaming Audio Quality.
  - Streaming Video Quality.
  - Streaming Audio/Video De-Synchronization.
- For short message services:
  - End-to-end Delivery Time.
- For multimedia message services:
  - MMS Retrieval Failure Ratio (MT).
  - MMS Send Time (MO).
  - MMS Retrieval Time (MT).

For wireless application protocol services, a reference from the Open Mobile Alliance (OMA) should be added (if available).

6.3 Ping

6.3.1 Ping Round Trip Time [ms]

6.3.1.1 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.

6.3.1.2 Abstract Equation

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

6.3.1.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Point of time when packet is sent	Start: User starts Ping client.	Start: ICMP echo request sent.
Point of time when packet is received	Stop: Echo reply is displayed.	Stop: ICMP echo reply received by the sender.

As an alternative the measurement of the round trip time can 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.

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

6.4 Push to Talk over Cellular (PoC)

The present clause describes QoS parameters for the Push to Talk over Cellular (PoC) service as described in [14], [15], [16] and [17].

To point up the development and effectiveness of these QoS parameters, a generic PoC signal flow is given. Here, some restricted information on the application layer is given. These events only show important user interactions. In this context it is important to point out that the present document does not focus on the application layer or the user plane as described in [15].

The SDP is not mentioned as an alternative to RTP in [14], [15] and [16]. Thus, trigger points defined on the SDP layer are out of scope of the present document.

NOTE: All Quality of Service parameters defined for the PoC service which do not rely on RTP in terms of trigger point definition are to be applied when measuring a PoC service utilizing STP.

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

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

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

Remark(s):

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

6.4.1 Definitions

For PoC, there are differences between On-demand and Pre-established PoC Sessions which is needed to be taken into account. Thus, a direct comparison between these session types shall be avoided.

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

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

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

Table 1: QoS parameter and required preconditions

QoS Group		Description	QoS parameter in this group	Preconditions
REG		PoC Registration	6.4.3, 6.4.4	-
PUB		PoC Publish	6.4.5, 6.4.6	REG
REG long		PoC Registration + PoC Publish	6.4.6.3, 6.4.7.3	-
On demand	INIT	PoC Session Initiation	6.4.8.3,6.4.9.3	PUB
	SETUP	PoC Session Setup	6.4.14.3, 6.4.17	-
	PtS	Push to Speech	6.4.18, 6.4.19	PUB
	LEAVE	PoC Session Leaving	6.4.20, 6.4.21	INIT or SETUP
Pre-established	NEGO	PoC Media Parameters Negotiation	6.4.10.3, 6.4.11.3	PUB
	INIT	PoC Session Initiation	6.4.12.3, 6.4.14	NEGO
	SETUP	PoC Session Setup	6.4.16, 6.4.17	-
	PtS	Push to Speech	6.4.18, 6.4.19	PUB
	LEAVE	PoC Session Leaving	6.4.22, 6.4.23	INIT or SETUP
DeREG		PoC Deregistration	6.4.24, 6.4.25	REG or SETUP
BUSY		Busy Floor Response	6.4.26, 6.4.27	SETUP or PtS
REQ		Talk Burst Request	6.4.28, 6.4.29	SETUP or PtS
CUT		PoC Session Cut-off	6.4.30	SETUP or PtS
DROP		Talk Burst Drop	0	SETUP or PtS
DELAY		Talk Burst Delay	6.4.32, 6.4.33	SETUO or PtS

6.4.2 Generic Signal Flow

This clause gives an overview of some signal flows evolving from PoC Sessions. In figure 5, a generic signal flow is given. Here, the main parts of a PoC Session, also including the registration of the PoC Service, are visualized. These are: PoC Service Registration (including PoC service settings publishment), PoC Session Initiation, PoC Talk Session, PoC Session Leaving and finally the PoC Service Deregistration.

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

In the present document greyed lines are optional signals which do not have to be send (like the "SIP NOTIFY" message which will only be send by the PoC Server if the "norefersub" option tag was included in the "SIP REFER" request (cf. [16])). Provisional SIP responses as described in [6] (e.g. "SIP 100 Trying") are greyed for clarity. These messages are provisional responses and shall be turned off during measurements.

A generic PoC Session:

PoC Registration.

PoC Session Initiation.

PoC Talk Session.

Leaving PoC Session.

PoC Deregistration.

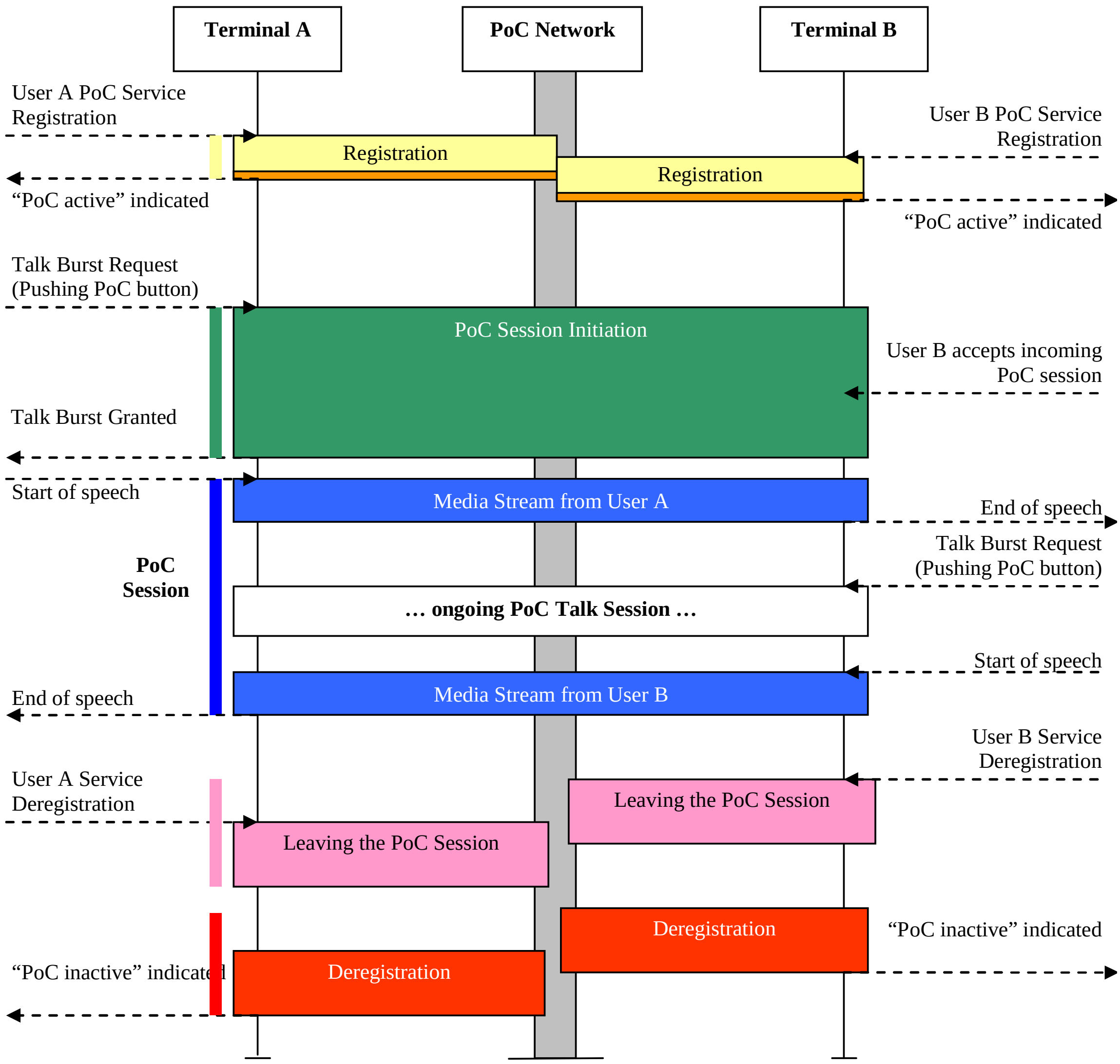


Figure 5: Generic PoC Session signal flow (including PoC Service Registration) on application layer.

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

6.4.3 PoC Registration Failure Ratio [%]

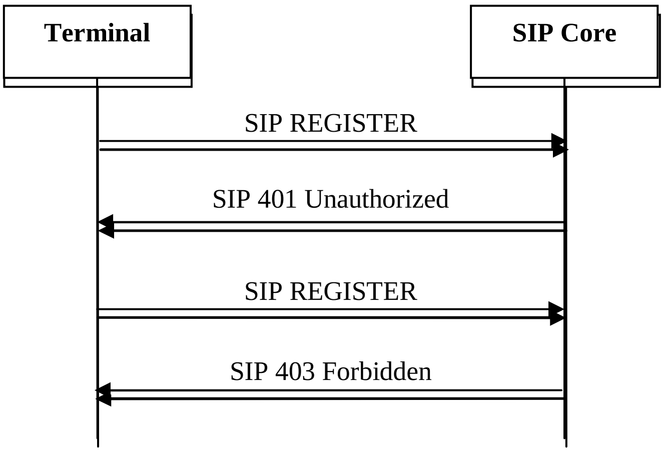
6.4.3.1 Abstract Definition

The PoC Registration Failure Ratio is the probability that the terminal can not register with the Push to Talk over Cellular Service when requested.

Remark(s): The terminal shall not be registered to the PoC Service.

6.4.3.2 Abstract Equation

$$\text{PoC Registration Failure Ratio } [\%] = \frac{\text{Number of unsuccessfull PoC Registration Attempts}}{\text{Number of all PoC Registration Attempts}} \times 100\,\%$$



NOTE: Figure 6 shows an example for an unsuccessful PoC Registration. After the first "SIP REGISTER" request the terminal has to answer to a WWW- authentication challenge (cf. [16]). If the terminal does not answer correctly to this challenge, the SIP Core will send a "SIP 403 Forbidden" message.

Figure 6: Unsuccessful PoC Registration example

6.4.3.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
PoC registration attempt	Start: Activation of the PoC service on the terminal	Start: Protocol: SIP First data packet sent by the terminal containing a "SIP REGISTER" message
Unsuccessful PoC registration attempt	Stop: PoC available indication is not given within a pre-determined time	Stop: Protocol: SIP Case 1: Second data packet received by the terminal (after sending the "SIP REGISTER" message) containing a message different to "SIP 200 OK". This message may be implementation-dependent (cf. [16]). Case 2: First data packet received by the terminal (after the authentication procedure) containing a message different to "SIP 200 OK". Case 3: No message received by the terminal within a pre-determined time.

6.4.4 PoC Registration Time [s]

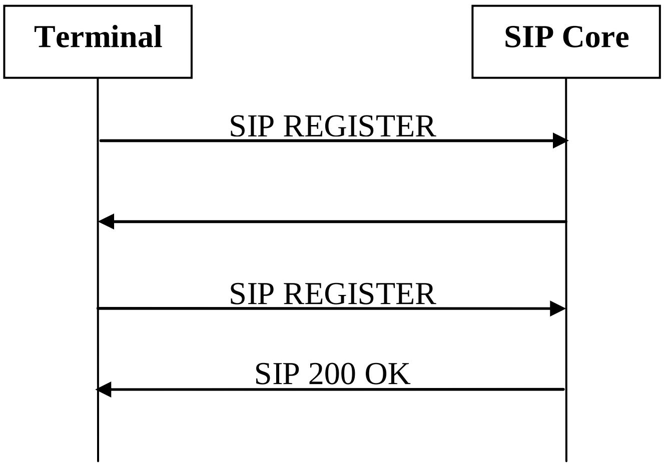
6.4.4.1 Abstract Definition

The PoC Registration Time is the time period between the registration request of the PoC Service and being registered to the PoC Service.

Remark(s): The terminal shall not be registered to the PoC Service.

6.4.4.2 Abstract Equation

$$\boxed{\text{PoC Registration Time }[\text{s}] = t_{PoCAvailable} - t_{PoCActivated}}$$



NOTE: Figure 7 shows an example of a successful PoC Registration (cf. [16]). In contrast to Figure 11, the terminal answered correctly to the authentication challenge (the 2nd "SIP REGISTER" message).

Figure 7: Successful PoC Registration example

6.4.4.3 Trigger Points

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

6.4.5 PoC Publish Failure Ratio [%]

6.4.5.1 Abstract Definition

The PoC Publish Failure Ratio is the probability that the terminal can not successfully publish his PoC service settings to the PoC Server, after the terminal is registered to the PoC Service.

Remark(s):

- To set, update or refresh the PoC service settings, the terminal generates a "SIP PUBLISH" request with XML MIME content according to rules and procedures of [17].
- The terminal shall be registered to the PoC Service.
- PoC enabled user equipment may combine the PoC Registration and the PoC Publish request and may not give the user the possibility to do these actions separately.

6.4.5.2 Abstract Equation

$$\text{PoC Publish Failure Ratio}\,[\%] = \frac{\text{Number of unsuccessfull PoC Publish Attempts}}{\text{Number of all PoC Publish Attempts}} \times 100\,\%$$

6.4.5.3 Trigger Points

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

6.4.6 PoC Publish Time [s]

6.4.6.1 Abstract Definition

The PoC Publish Time is the period of time that it takes to publish the terminal's PoC Service settings to the PoC Server.

Remark(s):

- To set, update or refresh the PoC service settings, the terminal generates a "SIP PUBLISH" request with XML MIME content according to rules and procedures of [17].
- The terminal shall be registered to the PoC Service.
- PoC enabled user equipment may combine the PoC Registration and the PoC Publish request and may not give the user the possibility to do these actions separately.

6.4.6.2 Abstract Equation

$$\text{PoC Publish Time}\,[s] = t_{PoCPublishEnd} - t_{PoCPublishStart}$$

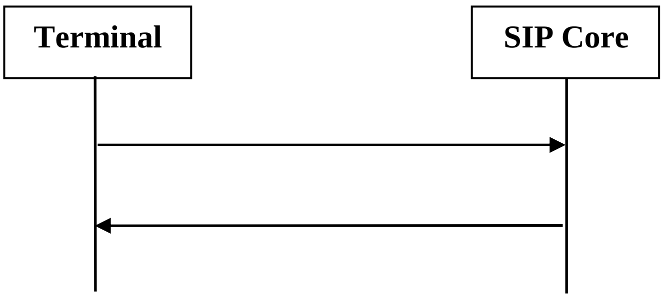


Figure 8: Example for a successful publish of PoC Service settings

6.4.6.3 Trigger Points

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

6.4.7 PoC Registration Failure Ratio (long) [%]

6.4.7.1 Abstract Definition

The PoC Registration Failure Ratio (long) is the probability that the terminal can not successfully be registered to the PoC Service and publish his PoC Service settings.

Remark(s):

- This QoS parameter is a combination of the PoC Registration parameter (cf. clause 6.4.3) and the PoC Publish parameter (cf. clause 6.4.5). It is ought to reflect the behaviour of PoC enabled user equipment that may do the PoC Publish automatically after the PoC Register.
- The terminal shall not be registered to the PoC Service.

6.4.7.2 Abstract Equation

$$\text{PoC Registration Failure Ratio (long) } [\%] = \frac{R + P}{\text{Number of all Registration (long) Attempts}} \times 100\,\%$$

6.4.7.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Start event of the PoC Registration Failure Ratio parameter	Start: Start trigger of the PoC Registration Failure Ratio parameter	Start: Start trigger of the PoC Registration Failure Ratio parameter
End event of the PoC Publish Failure Ratio parameter	Stop: End trigger of the PoC Publish Failure Ratio parameter	Stop: End trigger of the PoC Publish Failure Ratio parameter

6.4.8 PoC Registration Time (long) [s]

6.4.8.1 Abstract Definition

The PoC Registration Time (long) is the combined duration for a SIP Registration and a SIP Publish.

Remark(s):

- This QoS parameter is a combination of the PoC Registration parameter (cf. clause 6.4.3) and the PoC Publish parameter (cf. clause 6.4.5). It is ought to reflect the behaviour of PoC enabled user equipment that may do the PoC Publish automatically after the PoC Register.
- The terminal shall not be registered to the PoC service.

6.4.8.2 Abstract Equation

$$\text{PoC Registration Time (long)}\,[s] = t_{\text{Long Reg End}} - t_{\text{Long Reg Start}}$$

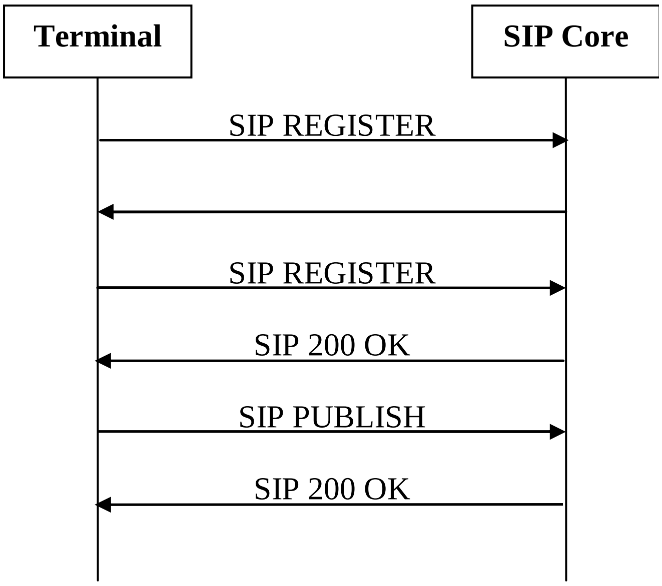


Figure 9: Example for a successful PoC Registration (long)

6.4.8.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Start event of the PoC Registration Time parameter $t_{\text{Long Reg Start}}$	Start: Start trigger of the PoC Registration Time parameter	Start: Start trigger of the PoC Registration Time parameter
End event of the PoC Publish Time parameter $t_{\text{Long Reg End}}$	Stop: End trigger of the PoC Publish Time parameter	Stop: End trigger of the PoC Publish Time parameter

6.4.9 PoC Session Initiation Failure Ratio (on-demand) [%]

6.4.9.1 Abstract Definition

The PoC Session Initiation Failure Ratio (on-demand) is the probability that a PoC Session can not be successfully initiated. A PoC Session is initiated when the user pushes the PoC button on the terminal (and thereby requests a Talk Burst) and is granted a Talk Burst (cf. figure 10).

Remark(s):

- The terminal notifies the user about the granted Talk Burst (e.g. by a "beep"-tone)
- There shall be at least one other participating terminal and the floor shall be idle. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- All terminals shall be registered to the PoC Service and shall have successfully published their PoC service settings.

- There are different signal flows for confirmed and for unconfirmed invitations. In the confirmed case, at least one of the invited users has to accept the invitation to the PoC Session in order to get the Talk Burst granted (cf. [16]). If the PoC Server supports Media Buffering, the Talk Burst confirm is send after the first received auto-answer. This automatic answer mode shall be used for the measurements and Media Buffering shall not be supported. In both cases (confirmed and unconfirmed) the trigger points for the measurement are the same. Measurement data of confirmed and unconfirmed measurements can not be directly compared.

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

- The initial "SIP INVITE" message accepted by the PoC Server is an implicit Talk Burst request.

6.4.9.2 Abstract Equation

$$\text{PoC Session Initiation Failure Ratio } [\%] = \frac{\text{Number of unsuccessful PoC Session Initiations}}{\text{Number of all PoC Session Initiations}} \times 100\ \%$$

6.4.9.3 Trigger Points

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

6.4.10 PoC Session Initiation Time (on-demand) [s]

6.4.10.1 Abstract Definition

The PoC Session Initiation Time (on-demand) is the time period between pushing the PoC Button on the terminal in order to initiate a PoC Session and being granted the Talk Burst, e.g. indicated by a "beep"-tone on the terminal.

Remark(s):

- The terminal notifies the user about the granted Talk Burst (e.g. by a "beep"-tone).

- There shall be at least one other participating terminal and the floor shall be idle. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).

- All terminals shall be registered to the PoC Service and shall have successfully published their PoC service settings.

- There are different signal flows for confirmed and for unconfirmed invitations. In the confirmed case, at least one of the invited users has to accept the invitation to the PoC Session in order to get the Talk Burst granted (cf. [15]). If the PoC Server supports Media Buffering, the Talk Burst confirm is send after the first received auto-answer. This automatic answer mode shall be used for the measurements and Media Buffering shall not be supported. In both cases (confirmed and unconfirmed) the trigger points for the measurement are the same. Measurement data of confirmed and unconfirmed measurements can not be directly compared.
- This parameter is applicable to different kinds of PoC Session Initiations, which has an impact on the comparability of the measurement data.
- The initial "SIP INVITE" message accepted by the PoC Server is an implicit Talk Burst request.

6.4.10.2 Abstract Equation

$$\text{PoC Session Initiation Time}\,[s] = t_{\text{beep received}} - t_{\text{PoC button pressed}}$$

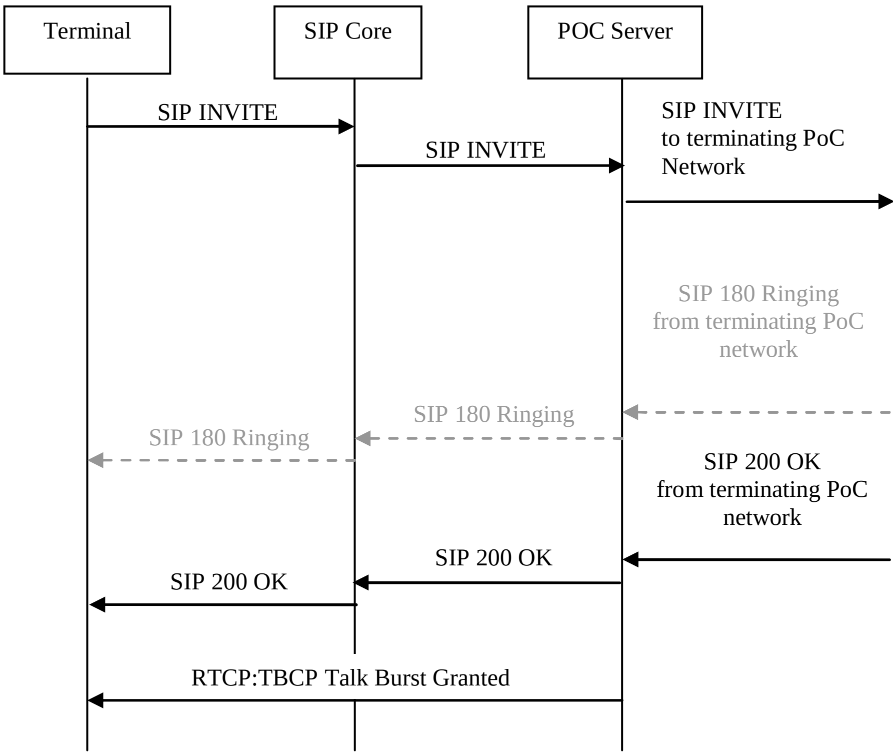


Figure 10: Implicit Talk Burst Request Procedure at the initiation of the PoC Session

Remark(s):

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

6.4.10.3 Trigger Points

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

6.4.11 PoC Pre-established Session Media Parameters Negotiation Failure Ratio [%]

6.4.11.1 Abstract Definition

Pre-established Session Parameters Negotiation Failure Ratio is the probability that a negotiation procedure of media parameters for a posterior Pre-established Session can not be successfully accomplished.

Remark(s):

- The initial "SIP INVITE" message accepted by the PoC server is not an implicit Talk Burst request.
- All terminals shall be registered to the PoC Service and shall have successfully published their PoC service settings.
- "The PoC Server performing the Controlling PoC Function shall determine the codec(s) and Media Parameters that should be used in the PoC Session. The preferred Media Parameters should be determined according to the lowest negotiated Media Parameters (e.g. bandwidth) of the terminals that have joined the PoC Session (cf. [14], page 102)."
- "User Plane adaptation may be triggered e.g. by roaming or when a new terminal with lower Media Parameters enters the PoC Session (cf. [15], page 103)."

6.4.11.2 Abstract Equation

$$\text{Media Parameter Negotiation Failure Ratio } [\%] = \frac{\text{Number of unsuccessful Negotiation Attempts}}{\text{Number of all Negotiation Attempts}} \times 100\ \%$$

6.4.11.3 Trigger Points

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

6.4.12 PoC Pre-established Session Media Parameters Negotiation Time [s]

6.4.12.1 Abstract Definition

The PoC Pre-established Session Parameters Negotiation Time describes the time period needed to accomplish a successful negotiation of media parameters.

Remark(s):

- The initial "SIP INVITE" message accepted by the PoC server is not an implicit Talk Burst request.
- All terminals shall be registered to the PoC Service and shall have successfully published their PoC service settings.
- "The PoC Server performing the Controlling PoC Function shall determine the codec(s) and Media Parameters that should be used in the PoC Session. The preferred Media Parameters should be determined according to the lowest negotiated Media Parameters (e.g. bandwidth) of the terminals that have joined the PoC Session (cf. [14], page 102)."
- "User Plane adaptation may be triggered e.g. by roaming or when a new terminal with lower Media Parameters enters the PoC Session (cf. [14], page 103)."

6.4.12.2 Abstract Equation

$$\boxed{\text{Media Parameter Negotiation Time } [s] = t_{\text{ok received}} - t_{\text{Negotiation initiation}}}$$

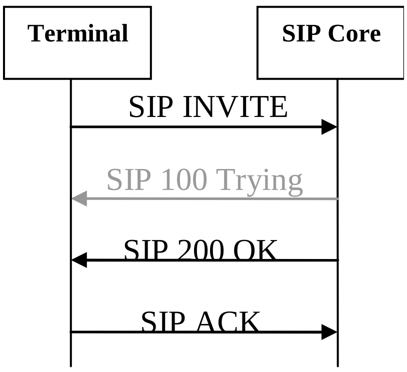


Figure 11: Media parameters negotiation for Pre-established Session

6.4.12.3 Trigger Points

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

6.4.13 PoC Session Initiation Failure Ratio (pre-established) [%]

6.4.13.1 Abstract Definition

The PoC Session Initiation Failure Ratio (pre-established) is the probability that a Pre-established Session can not be successfully initiated. After the negotiation of media parameters, a Pre-established Session is initiated when the user pushes the PoC button on the terminal (and thereby requests the Talk Burst) and is granted the Talk Burst.

Remark(s):

- The terminal notifies the user about the granted Talk Burst (e.g. by a "beep"-tone).
- The initial "SIP REFER" message accepted by the PoC server is an implicit Talk Burst request.
- There shall be at least one other participating terminal and the floor shall be idle. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- The terminals shall have negotiated the session media parameters with the PoC Server.
- All terminals in the PoC Session shall be configured to use the auto-answer mode procedure (cf. [16]).
- There are different signal flows for confirmed and for unconfirmed invitations. In the confirmed case, at least one of the invited users has to accept the invitation to the PoC Session in order to get the Talk Burst granted. The terminals on the terminating side may be configured to confirm the invitation automatically. This auto-answer mode should be used for measurements. In both cases (confirmed and unconfirmed) the trigger points for the measurement are the same. Measurement data of confirmed and unconfirmed measurements can not be directly compared.
- This parameter is applicable to different kinds of PoC Session Initiations, which has an impact on the comparability of the measurement data.

6.4.13.2 Abstract Equation

$$\text{PoC Session Leaving Failure Ratio (Pre)}\,[\%] = \frac{\text{Number of unsuccessful PoC Session Leaving Attempts}}{\text{Number of all PoC Session Leaving Attempts}} \times 100\ \%$$

6.4.13.3 Trigger Points

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

6.4.14 PoC Session Initiation Time (pre-established) [s]

6.4.14.1 Abstract Definition

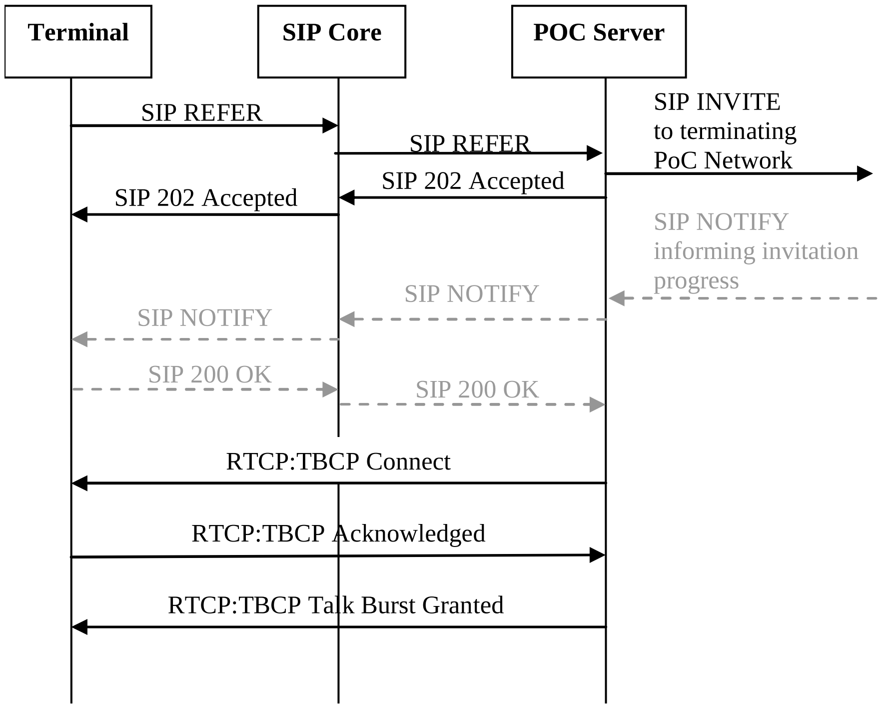
The PoC Session Initiation Time (pre-established) is the time period between pushing the PoC button on the terminal in order to initiate a Pre-established Session and being granted the Talk Burst, e.g. indicated by a "beep"-tone on the terminal.

Remark(s):

- The terminal notifies the user about the granted Talk Burst (e.g. by a "beep"-tone).
- The initial "SIP REFER" message accepted by the PoC server is an implicit Talk Burst request.
- There shall be at least one other participating terminal and the floor shall be idle. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- The terminals shall have negotiated the session media parameters with the PoC Server.
- All terminals in the PoC Session shall be configured to use the auto-answer mode procedure (cf. [16]).
- There are different signal flows for confirmed and for unconfirmed invitations. In the confirmed case, at least one of the invited users has to accept the invitation to the PoC Session in order to get the Talk Burst granted. The terminals on the terminating side may be configured to confirm the invitation automatically. This auto-answer mode should be used for measurements. In both cases (confirmed and unconfirmed) the trigger points for the measurement are the same. Measurement data of confirmed and unconfirmed measurements can not be directly compared.
- This parameter is applicable to different kinds of PoC Session Initiations, which has an impact on the comparability of the measurement data.

6.4.14.2 Abstract Equation

$$\boxed{\text{PoC Pre-established Session Initiation Time }[s] = t_{\text{beep received}} - t_{\text{PoC button pressed}}}$$



NOTE: The dashed arrows in figure 12 only occur in case of a confirmed, manual answer invitation. In this case the time period between the "SIP INVITE" message and the reception of the Talk Burst Granted message depends on how fast an invited user on the terminating side answers to the invitation. Furthermore, the "SIP NOTIFY" message is defined as optional (cf. [15]) and might not be sent by the server at all. For this reason the automatic answer mode shall be used during measurements.

Figure 12: Talk Burst Request Procedure of a Pre-established PoC Session

6.4.14.3 Trigger Points

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

6.4.15 PoC Session Setup Failure Ratio (on-demand) [%]

6.4.15.1 Abstract Definition

The PoC Session Setup Failure Ratio (on-demand) is the probability that a terminal can not successful register to the PoC Service and initialize an On-demand Session.

Remark(s):

- This QoS parameter is a combination of the PoC Registration parameter and the PoC Session Initiation parameter. It is ought to reflect the behaviour of PoC enabled user equipment.
- Data between Confirmed and Unconfirmed measurements cannot be compared directly.

6.4.15.2 Abstract Equation

Let *R* be the number of unsuccessful Registration attempts and let *S* be the number of unsuccessful Session Initiation following a successful Registration

Then:

$$\text{PoC Session Leaving Failure Ratio (Pre)}\,[\%] = \frac{\text{Number of unsuccessful PoC Session Leaving Attempts}}{\text{Number of all PoC Session Leaving Attempts}} \times 100\ \%$$

6.4.15.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Start event of the PoC Registration Failure Ratio	Start: Start trigger of the PoC Registration Failure Ratio	Start: Start trigger of the PoC Registration Failure Ratio
End event of the PoC Session Initiation Parameter	Stop: End trigger of the PoC Session Parameter	Stop: End trigger of the PoC Session Initiation Parameter

6.4.16 PoC Session Setup Failure Ratio (pre-established) [%]

6.4.16.1 Abstract Definition

The PoC Session Setup Failure Ratio (pre-established) is the probability that a terminal can not successful register to the PoC Service and initialize a Pre-established Session.

Remark(s):

- This QoS parameter is a combination of the PoC Registration parameter and the PoC Session Initiation parameter. It is ought to reflect the behaviour of PoC enabled user equipment.
- Data between Confirmed and Unconfirmed measurements cannot be compared directly.

6.4.16.2 Abstract Equation

Let *R* be the number of unsuccessful registration attempts and let *S* be the number of unsuccessful Pre-established Session Media Parameters negotiations following a successful registration. Let *T* be the number of unsuccessful Session Initiation attempts, which followed after a successful Registration and after a successful Pre-established Session Media Parameters Negotiation.

Then:

$$\text{PoC Session Leaving Failure Ratio (Pre)}\,[\%] = \frac{\text{Number of unsuccessful PoC Session Leaving Attempts}}{\text{Number of all PoC Session Leaving Attempts}} \times 100\ \%$$

6.4.16.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Start event of the PoC Registration Failure Ratio	Start: Start trigger of the PoC Registration Failure Ratio	Start: Start trigger of the PoC Registration Failure Ratio
End event of the PoC Session Initiation Parameter	Stop: End trigger of the PoC Session Initiation Parameter	Stop: End trigger of the PoC Session Initiation Parameter

6.4.17 PoC Session Setup Time [s]

- Abstract Definition:

The PoC Session Setup Time is the time period for the registration to the PoC Service plus the time period for the initiation of a PoC Session.

Remark(s):

- This QoS parameter is a combination of the PoC Registration parameter and the PoC Session Initiation parameter. It is ought to reflect the behaviour of PoC enabled user equipment.
- Data between Confirmed and Unconfirmed measurements cannot be compared directly.
- Data between On-demand Sessions and Pre-established Sessions cannot be compared directly

6.4.17.1 Abstract Equation

$$\boxed{\text{PoC Session Setup Time}\,[s] = t_{\text{Session SetupEnd}} - t_{\text{Session Setup Start}}}$$

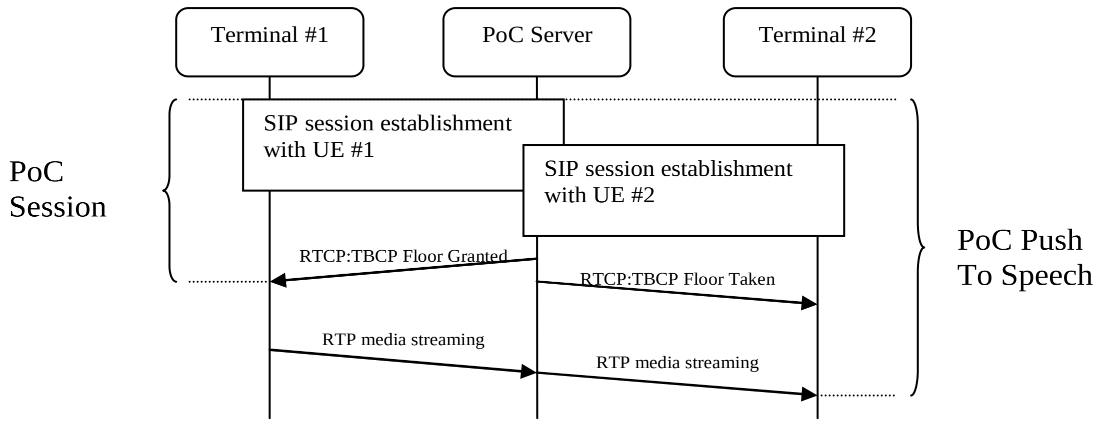


Figure 13: PoC Session Setup Time and PoC Push to Speech Time

6.4.17.2 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Start event of the PoC Registration Time $t_{\text{Session Setup Start}}$	Start: Start trigger of the PoC Registration Time	Start: Start trigger of the PoC Registration Time
Stop event of the PoC Session Initiation Time $t_{\text{Session Setup End}}$	Stop: End trigger of the PoC Session Initiation Time	Stop: End trigger of the PoC Session Initiation Time

6.4.18 PoC Push to Speech Failure Ratio [%]

6.4.18.1 Abstract Definition

The PoC Push to Speech Failure Ratio is the probability that terminal A can not successfully set up a PoC Session and start with speech leading to no other terminal receiving speech.

Remark(s):

- This QoS parameter is a combination of the PoC Session Setup parameter and the PoC Talk-burst Cut-off parameter (cf. clause 6.4.30). It is ought to reflect the behaviour of PoC enabled user equipment.
- All terminals shall be registered to the PoC Service and shall have successfully published their PoC service settings.
- Data between Confirmed and Unconfirmed measurements cannot be compared directly.

6.4.18.2 Abstract Equation

Let *S* be the number of unsuccessful PoC Session Setup attempts and let *T* be the number of Talk-burst Cut-offs following a successful PoC Session Setup.

Then:

$$\text{PoC Session Leaving Failure Ratio (Pre)}\left[\%\right] = \frac{\text{Number of unsuccessful PoC Session Leaving Attempts}}{\text{Number of all PoC Session Leaving Attempts}} \times 100\ \%$$

6.4.18.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Start event of the PoC Session Setup Failure Ratio	Start trigger of the PoC Session Setup Failure Ratio	Start trigger of the PoC Session Setup Failure Ratio
End event of the PoC Talk Burst Cut-off parameter	End trigger of the PoC Talk Burst Cut-off parameter	End trigger of the PoC Talk Burst Cut-off parameter

6.4.19 PoC Push to Speech Time [s]

6.4.19.1 Abstract Definition

The PoC Push to Speech Time is the period of time that it takes to setup a PoC Session and start with speech in addition to the delay until terminal B receives the speech (as defined in clause 6.4.32).

Remark(s):

- This QoS parameter is a combination of the PoC Session Setup Time parameter and the PoC Voice Delay parameter (cf. clause 6.4.32). It is ought to reflect the behaviour of PoC enabled user equipment.
- All terminals shall be registered to the PoC Service and shall have successfully published their PoC service settings.
- Data between Confirmed and Unconfirmed measurements cannot be compared directly.

6.4.19.2 Abstract Equation

$$\text{PoC Push to Speech Time}\,[s] = t_{\text{Push To Speech End}} - t_{\text{Push to Speech Start}}$$

6.4.19.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Start event of the PoC Session Setup Time $t_{\text{Push to Speech Start}}$	Start: Start trigger of the PoC Session Setup Time	Start: Start trigger of the PoC Session Setup Time
Stop event of the PoC Voice Delay parameter $t_{\text{Push to Speech End}}$	Stop: End trigger of the PoC Voice Delay parameter	Stop: End trigger of the PoC Voice Delay parameter

6.4.20 PoC Session Leaving Failure Ratio (on-demand) [%]

6.4.20.1 Abstract Definition

The PoC Session Leaving Failure Ratio (on-demand) is the probability that the user can not leave the PoC Session he is participating.

Remark(s):

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

6.4.20.2 Abstract Equation

$$\text{PoC Session Leaving Failure Ratio (Pre)}\,[\%] = \frac{\text{Number of unsuccessful PoC Session Leaving Attempts}}{\text{Number of all PoC Session Leaving Attempts}} \times 100\ \%$$

6.4.20.3 Trigger Points

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

6.4.21 PoC Session Leaving Time (on-demand) [s]

6.4.21.1 Abstract Definition

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

Remark(s):

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

6.4.21.2 Abstract Equation

$$\boxed{\text{PoC Session Leaving Time}\,[\text{s}] = t_{SessionLeft} - t_{SessionLeaveRequest}}$$

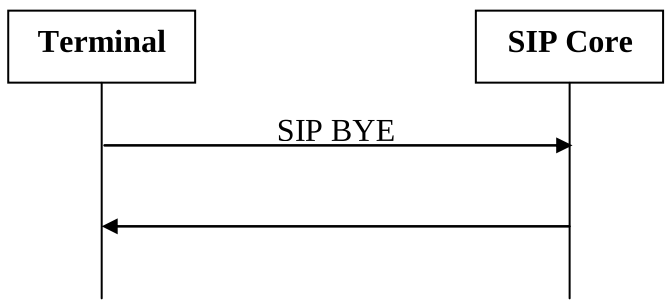


Figure 14: Successful PoC Session leaving

6.4.21.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
PoC Session Leaving Attempt $t_{SessionLeaveRequest}$	Start: Leaving the participating PoC Session	Start: Protocol: SIP First data packet sent by the terminal containing a "SIP BYE" message
Successful PoC Session Leaving Attempt $t_{SessionLeft}$	Stop: PoC Session left is indicated	Stop: Protocol: SIP First data packet received by the terminal containing a "SIP 200 OK" message.

6.4.22 PoC Session Leaving Failure Ratio (pre-established) [%]

6.4.22.1 Abstract Definition

The PoC Session Leaving Failure Ratio (pre-established) is the probability that the user can not leave the PoC Pre-established Session he is participating.

Remark(s):

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

6.4.22.2 Abstract Equation

$$\text{PoC Deregistration Failure Ratio } [\%] = \frac{\text{Number of unsuccessfull PoC Deregistration Attempts}}{\text{Number of all PoC Deregistration Attempts}} \times 100\ \%$$

6.4.22.3 Trigger Points

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

6.4.23 PoC Session Leaving Time (pre-established) [s]

6.4.23.1 Abstract Definition

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

Remark(s):

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

6.4.23.2 Abstract Equation

$$\boxed{\text{PoC Session Leaving Time (Pre)}\,[s] = t_{SessionLeft} - t_{SessionLeaveRequest}}$$

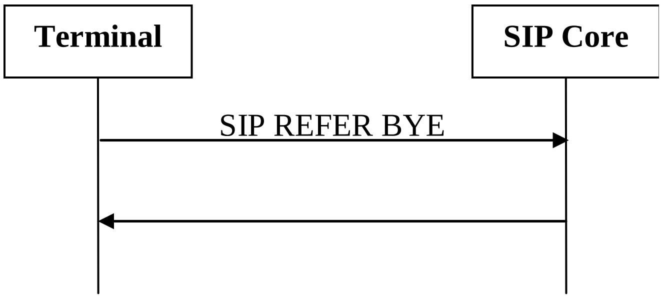


Figure 15: Successful PoC Session leaving (Pre-established Session)

6.4.23.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
PoC Session Leaving Attempt $t_{SessionLeaveRequest}$	Start: Leaving the participating PoC Session	Start: Protocol: SIP First data packet sent by the terminal containing a "SIP REFER BYE" message
Successful PoC Session Leaving Attempt $t_{SessionLeft}$	Stop: Terminal has successfully left the PoC Session	Stop: Protocol: SIP First data packet received by the terminal containing a "SIP 202 ACCEPTED" message.

6.4.24 PoC Deregistration Failure Ratio [%]

6.4.24.1 Abstract Definition

The PoC Deregistration Failure Ratio is the probability that the user can not be deregistered from the Push to Talk over Cellular Service when requested.

Remark(s):

- The terminal shall be registered to the PoC Service.

6.4.24.2 Abstract Equation

$$\boxed{\text{PoC Talk Burst Cut-off Ratio}\,[\%] = \frac{\text{Number of dropped Talk Bursts}}{\text{Number of all Talk Bursts}} \times 100\%}$$

6.4.24.3 Trigger Points

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

6.4.25 PoC Deregistration Time [s]

6.4.25.1 Abstract Definition

The PoC Deregistration Time is the time period between the de-activation request of the PoC service and being deregistered from the PoC service.

Remark(s):

- The terminal shall be registered to the PoC Service.

6.4.25.2 Abstract Equation

$$\text{PoC Deregistration Time}\,[s] = t_{PoCUnvailable} - t_{PoCDeactivated}$$

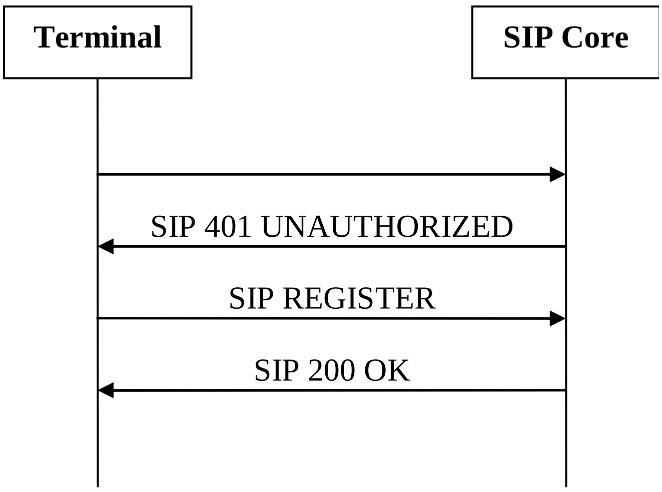


Figure 16: Successful PoC Deregistration Example

6.4.25.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
PoC De-Registration attempt $t_{PoCDeactivated}$	Start: De-activation of the PoC-Service on the terminal	Start: Protocol: SIP First data packet sent by the terminal containing a "SIP Register" message, where the "Expires" header is set to 0
Successful PoC Registration Attempt $t_{PoCUnavailable}$	Stop: PoC unavailable is indicated	Stop: Protocol: SIP First data packet received by the terminal containing a "SIP 200 OK" message.

6.4.26 PoC Busy Floor Response Failure Ratio [%]

6.4.26.1 Abstract Definition

The PoC Busy Floor Response Failure Ratio is the probability that, once in a PoC Session, the Talk Burst request from the terminal fails.

Remark(s):

- The terminal shall be within an active PoC Talk Session. Thus, there shall be at least one other participating terminal.
- For the special case of requesting the idle floor, there are defined further QoS parameters (cf. clauses 6.4.28, 6.4.29).

6.4.26.2 Abstract Equation

$$\text{PoC Talk Burst Cut-off Ratio}\,[\%] = \frac{\text{Number of dropped Talk Bursts}}{\text{Number of all Talk Bursts}} \times 100\%$$

6.4.26.3 Trigger Points

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

6.4.27 PoC Busy Floor Response Time [s]

6.4.27.1 Abstract Definition

The PoC Busy Floor Response Time is the is the time period between requesting the Talk Burst and receiving the indication the floor is busy within an already established PoC Session.

Remark(s):

- The terminal shall be within an active PoC Talk Session. Thus, there shall be at least one other participating terminal.
- For the special case of requesting the idle floor, there are defined further QoS parameters (cf. clauses 6.4.28 and 6.4.29).

6.4.27.2 Abstract Equation

$$\boxed{\text{PoC Busy Floor Response Time}\,[s] = t_{\text{floor response}} - t_{\text{floor request}}}$$

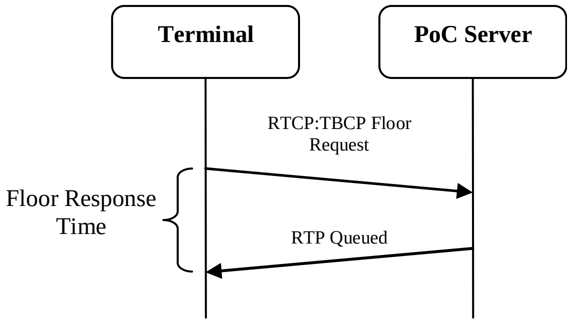


Figure 17: Example for a Busy Floor Response

6.4.27.3 Trigger Points

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

6.4.28 PoC Talk Burst Request Failure Ratio [%]

6.4.28.1 Abstract Definition

The PoC Talk Burst Request Failure Ratio is the probability that, once in a PoC Session, the terminal's request of the idle floor fails.

Remark(s):

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

6.4.28.2 Abstract Equation

$$\text{PoC Talk Burst Cut-off Ratio}\,[\%] = \frac{\text{Number of dropped Talk Bursts}}{\text{Number of all Talk Bursts}} \times 100\%$$

6.4.28.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
PoC Talk Burst request	Start: Push PoC button	Start: Protocol: RTCP:TBCP First data packet sent by the terminal containing a "RTCP:TBCP Talk Burst Request" message.
Unsuccessful PoC Talk Burst request	Stop: Talk Burst Granted is not indicated	Stop: Protocol: RTCP:TBCP Case 1: First data packet received by the terminal containing a floor state different to "RTCP:TBCP Talk Burst Granted". Possible floor states are listed in [14]. Case 2: No message received by the terminal within a pre-determined time.

6.4.29 PoC Talk Burst Request Time [s]

6.4.29.1 Abstract Definition

The PoC Talk Burst Request Time is the time period between requesting the Talk Burst and being granted the previously idle floor within an already established PoC Session.

Remark(s):

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

6.4.29.2 Abstract Equation

$$\text{PoC Talk Burst Request Time}\,[s] = t_{\text{floor granted}} - t_{\text{floor request}}$$

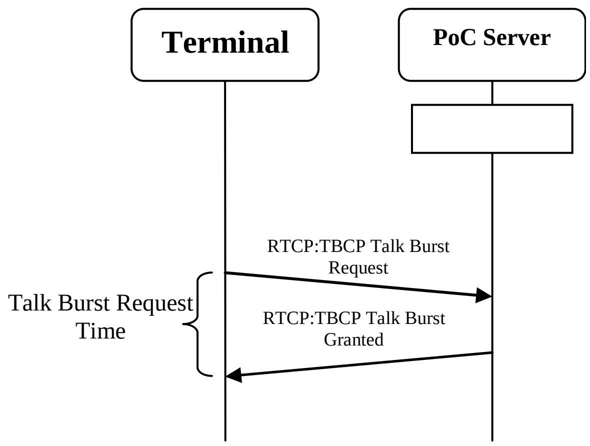


Figure 18: Example for a successful Talk Burst Request

6.4.29.3 Trigger Points

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

6.4.30 PoC Talk Burst Cut-off Ratio [%]

6.4.30.1 Abstract Definition

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

Remark(s):

- There shall be at least one other active participating terminal and the floor shall be granted to terminal A. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- The implementation of a stop-talking timer is mandatory on the server side. When a user is granted a Talk Burst, the PoC Server resets this stop-talking timer. When the timer expires, the PoC Server revokes the Talk Burst from the user (cf. [14]). Hence this situation (Talk Burst revoked because of a timeout) shall not be considered for measurements.
- The time of a Talk Burst shall be shorter than the network-defined stop-talking timeout.

6.4.30.2 Abstract Equation

$$\boxed{\text{PoC Talk Burst Cut-off Ratio}\,[\%] = \frac{\text{Number of dropped Talk Bursts}}{\text{Number of all Talk Bursts}} \times 100\%}$$

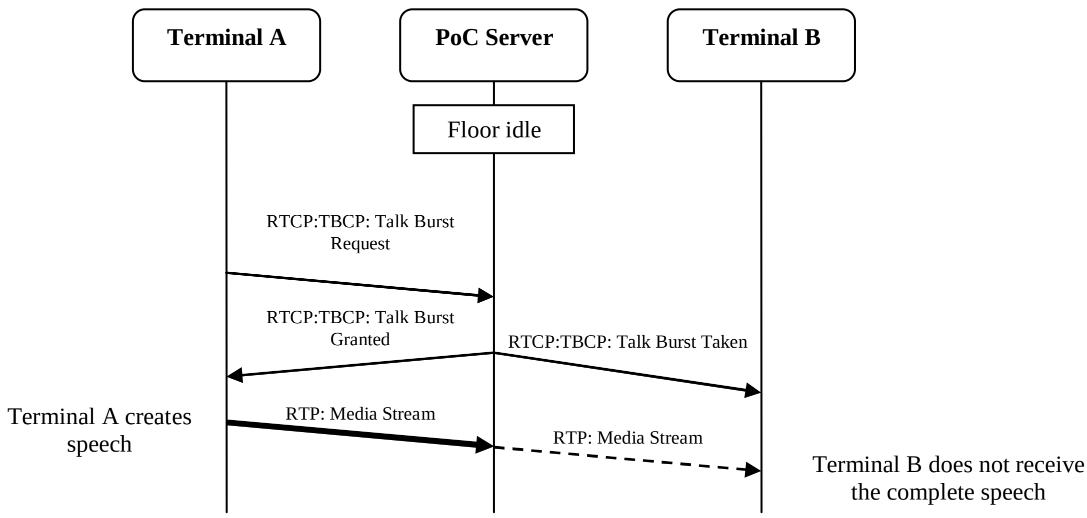


Figure 19: PoC Talk Burst Cut-off

6.4.30.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
PoC Talk Burst granted and start of speech on terminal A	Start: Talk Burst Granted is indicated. Speech starts	Start: Protocol: RTP First data packet sent by terminal A containing speech data.
Unintended speech cut-off on terminal B	Stop: Terminal B does not receive speech or does not receive the whole speech	Stop: Protocol: RTP Case 1: No packet containing speech data (RTP media stream) received by terminal B within a pre-determined time. The timeout should be chosen greater than the average voice delay (cf. clause 6.6.32). Case 2: The media stream is only partially received by terminal B. Some of the data packets containing speech data (RTP media stream) have not been received by terminal B.

6.4.31 PoC Talk Burst Packet Drop Ratio [%]

6.4.31.1 Abstract Definition

The PoC Talk Burst Packet Drop Ratio is the ratio between the number of data packets containing speech data sent by the terminal on the originating side (terminal A) and the number of data packets containing speech data received on the terminating side (terminal B).

Remark(s):

- There shall be at least one other active participating terminal and the floor shall be granted to terminal A. In particular, no other terminal shall create and send data packets containing speech data (RTP media stream).
- The implementation of a stop-talking timer is mandatory on the server side. When a user is granted a Talk Burst, the PoC server resets this stop-talking timer. When the timer expires, the PoC Server revokes the Talk Burst from the user (cf. [14]). Hence this situation (Talk Burst revoked because of a timeout) shall not be considered for measurements.
- The time of a Talk Burst shall be shorter than the network-defined stop-talking timeout.

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

6.4.31.2 Abstract Equation

$$\text{PoC Talk Burst Packet Drop Ratio}\,[\%] = \frac{\text{Number of dropped RTP Speach Packets}}{\text{Number of all Sent RTP Speach Packets}} \times 100\%$$

6.4.31.3 Trigger Points

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

6.4.32 PoC Voice transmission Delay (first) [s]

6.4.32.1 Abstract Definition

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

Remark(s):

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

6.4.32.2 Abstract Equation

$$\boxed{\text{PoC Voice Delay}\,[\text{s}] = t_{B\_hears} - t_{A\_speaks}}$$

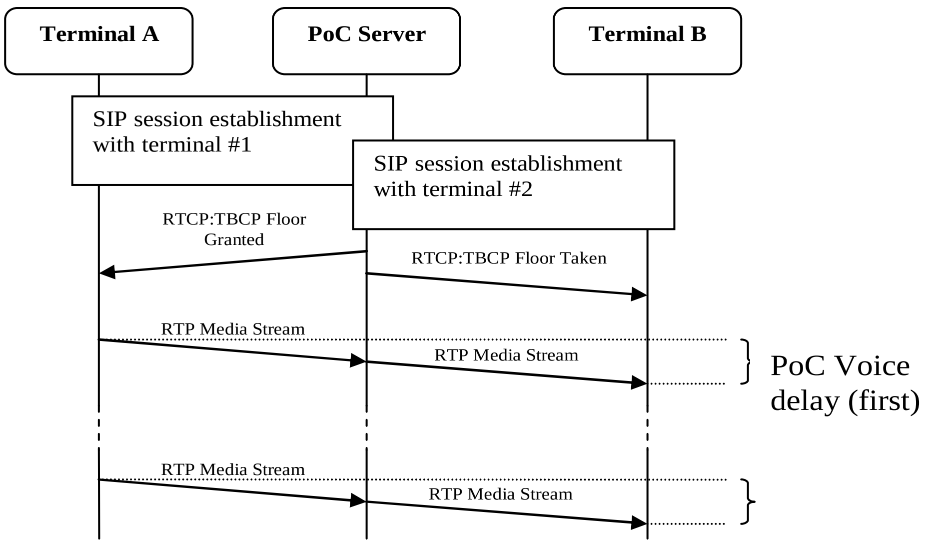


Figure 20: PoC Voice transmission Delay (first) and PoC Voice transmission Delay (others)

6.4.32.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Input at terminal A t_{A_speaks}	Start: Terminal A got the Talk Burst granted and creates an RTP media stream (starts talking)	Start: Protocol: RTP First data packet sent by terminal A containing speech data.
Output at terminal B t_{B_hears}	Stop: Sound received by terminal B	Stop: Protocol: RTP First data packet received by terminal B containing speech data.

6.4.33 PoC Voice transmission Delay (others) [s]

6.4.33.1 Abstract Definition

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

Remark(s):

- Without loss of generality, the PoC Session consists only of two active terminals (A and B) and terminal A is trying to create and send data packets containing speech data (RTP media stream). Thus, terminal B is the one who should receive the corresponding RTP media stream.
- Server side buffering has a high impact on measurement results. Depending on the configuration of the server, the PoC Voice transmission Delay (first) might in fact just describe the transmission delay between the server and terminal B. To avoid buffering at server side, Confirmed Indication shall be used.
- Terminal A shall create an RTP media stream immediately after being granted the Talk Burst.
- This parameter is measured on the transport layer. Thus the measured value may be smaller than the real user perceived voice delay. The perceived delay also depends on the encoding/decoding speed of the terminals.
- The voice delays on the terminating site depend on where the terminals are located (e.g. in another cell or another network).

6.4.33.2 Abstract Equation

$$\boxed{\text{PoC Voice Delay}\,[\text{s}] = t_{B\_hears} - t_{A\_speaks}}$$

6.4.33.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Input at terminal A t_{A_speaks}	Start: Terminal A got the Talk Burst granted and creates an RTP media stream (starts talking)	Start: Protocol: RTP First data packet sent by terminal A containing speech data.
Output at terminal B t_{B_hears}	Stop: Sound received at terminal B	Stop: Protocol: RTP First data packet received by terminal B containing speech data.

6.4.34 PoC Speech Quality

To be defined.

6.4.35 Group Management QoS Parameter

To be defined.

6.4.36 Group Document related QoS Parameter

To be defined.

6.4.37 Instant Message QoS Parameter

To be defined.

6.5 Streaming Video

6.5.1 Definitions

6.5.1.1 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 21 this means that the session starts at (B) and stops at (G).

6.5.2 Prerequisites

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

6.5.3 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.

6.5.3.1 Generic Streaming Signalling Flow

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

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

Protocol	Reference in figure 21	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.

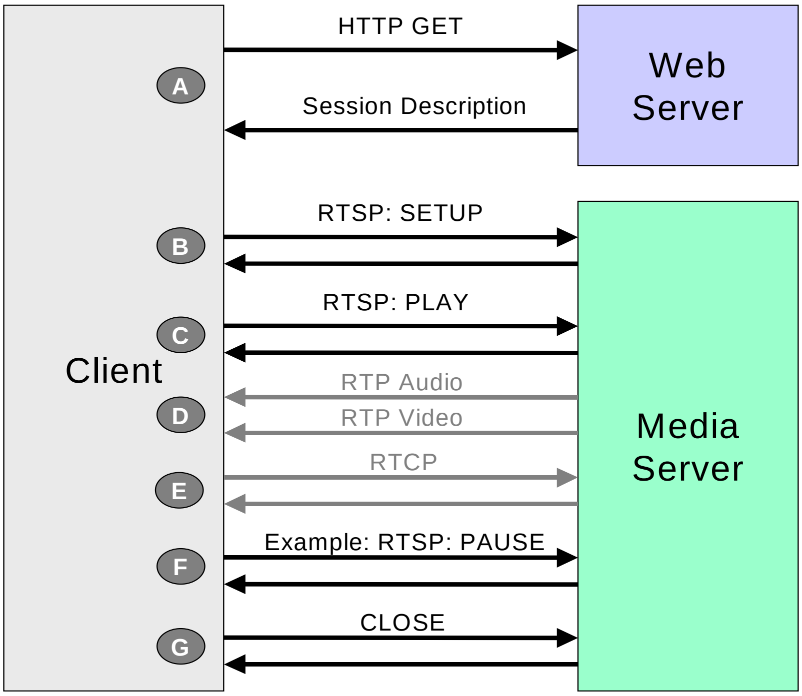


Figure 21: Generic session signalling flow, based on Schulzrinne

Referring to figure 21 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).

6.5.3.2 Parameter Overview Chart

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

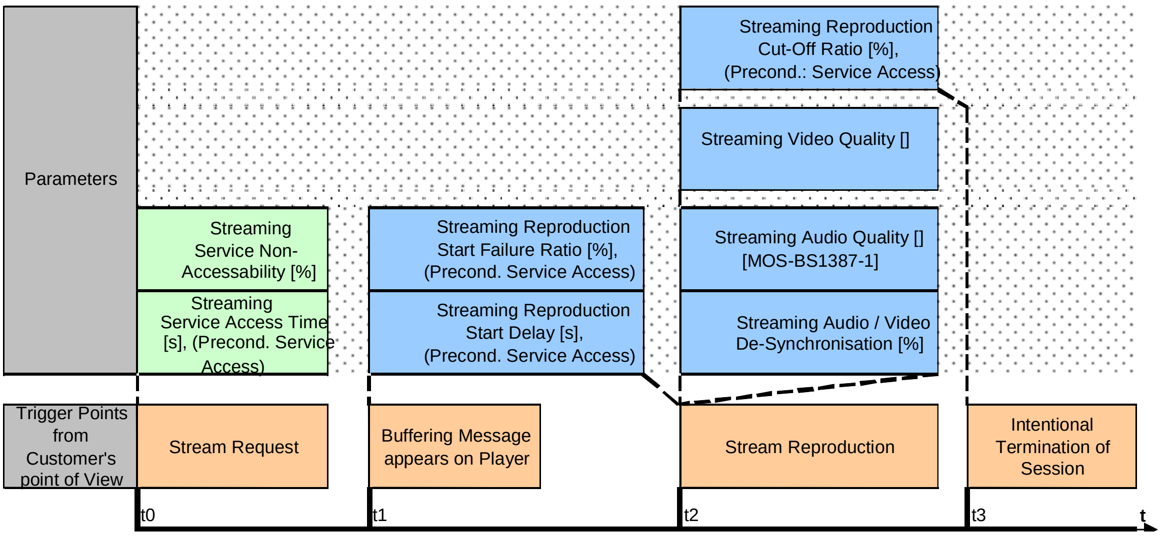


Figure 22: Parameter overview with trigger points

6.5.4 Streaming Service Non-Accessibility [%]

6.5.4.1 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.

6.5.4.2 Abstract Equation

$$\text{Sreaming Service Non-Accessibility [\%]} = \frac{\text{unsuccessful stream request attempts}}{\text{all stream request attempts}} \times 100\ \%$$

6.5.4.3 Trigger Points

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

6.5.5 Streaming Service Access Time [s]

6.5.5.1 Abstract Definition

The parameter Streaming Service Access Time describes the duration of a service access from requesting the stream at the portal until the reception of the first stream data packet at the UE.

The first data packet refers to RTP protocol.

6.5.5.2 Abstract Equation

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

6.5.5.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when stream is requested	Start: Stream request	Start: RTSP: Setup
Time when first data packet is received	Start: "Buffering" message	Stop: Reception of first data packet

6.5.6 Streaming Reproduction Cut-off Ratio [%]

6.5.6.1 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.

6.5.6.2 Abstract Equation

$$\text{Sreaming Reproduction Cut-off Ratio [\%]} = \frac{\text{unintentional terminated streaming reproductions}}{\text{all successfully started stream reproductions}} \times 100\ \%$$

6.5.6.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Successfully started media streaming reproduction	Start: Stream reproduction starts	Start: Streaming player signals the start of the stream reproduction.
Intentional terminated streaming reproduction	Stop: User presses the "Exit" button or end of stream is reached	Stop: RTSP Teardown method sent by UE and reception of confirmation "RTSP 200 OK" from media server
Unintentional terminated streaming reproduction	Stop trigger point not reached	

NOTE: Not all players may signal the reproduction start.

Some players do not send this TEARDOWN command at the end of the stream but a PAUSE command or in some cases nothing 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.

6.5.7 Streaming Audio Quality

6.5.7.1 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].

6.5.7.2 Abstract Equation

To be defined.

6.5.7.3 Trigger Points

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

6.5.8 Streaming Video Quality

6.5.8.1 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.

6.5.8.2 Abstract Equation

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.

6.5.8.3 Trigger Points

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

6.5.9 Streaming Audio/Video De-Synchronization

6.5.9.1 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.

6.5.9.2 Abstract Equation

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

6.5.9.3 Trigger Points

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

6.5.10 Streaming Reproduction Start Failure Ratio [%]

6.5.10.1 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.

6.5.10.2 Abstract Equation

$$\text{Sreaming Reproduction Start Failure Ratio [\%]} = \frac{\text{reproduction failures}}{\text{all successful service accesses}} \times 100\ \%$$

6.5.10.3 Trigger Points

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

6.5.11 Streaming Reproduction Start Delay [s]

6.5.11.1 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.

6.5.11.2 Abstract Equation

$$\text{Streaming Reproduction Start Delay [s]} = t_{\text{start of stream reproduction}} - t_{\text{reception of first data packet}}$$

6.5.11.3 Trigger Points

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

6.6 Telephony

6.6.1 Telephony Service Non-Accessibility [%]

6.6.1.1 Abstract Definition

Probability that the end-customer cannot access the Mobile Telephony Service when requested if it is offered by display of the network indicator on the mobile equipment.

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

6.6.1.2 Abstract Equation

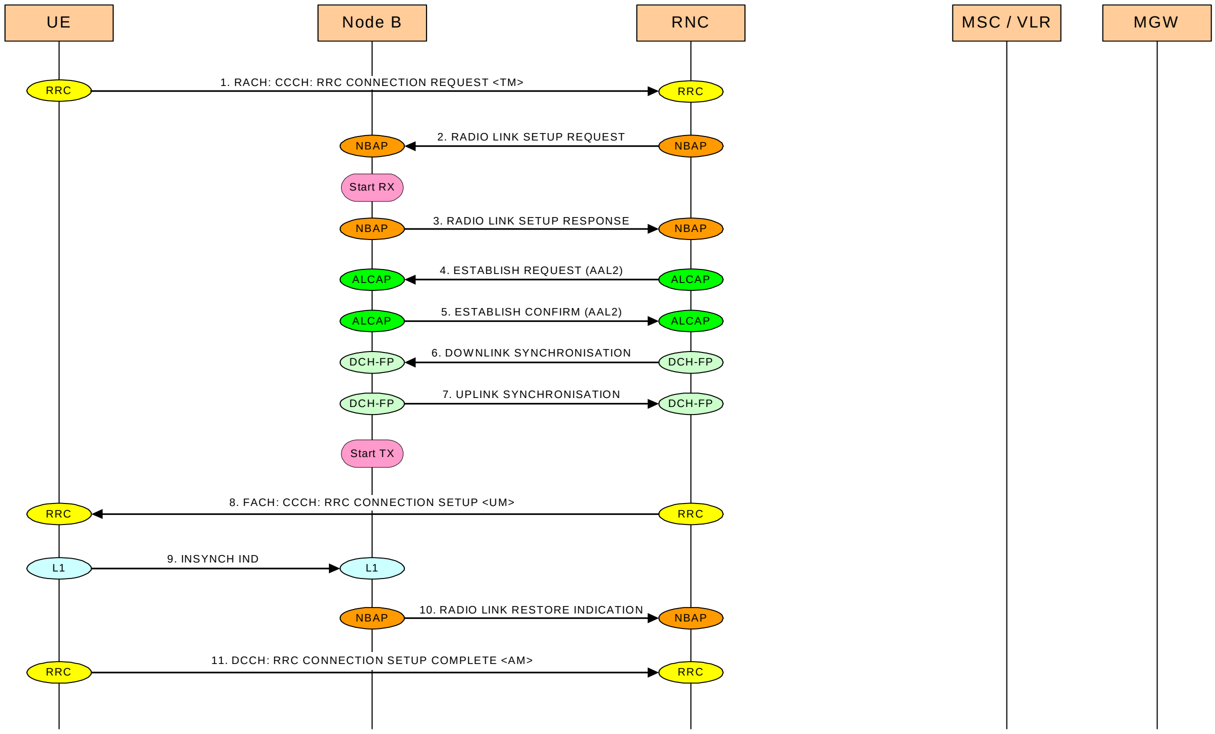
$$\text{Telephony Service Non-Accessibility [\%]} = \frac{\text{unsuccessful call attempts}}{\text{all call attempts}} \times 100\ \%$$

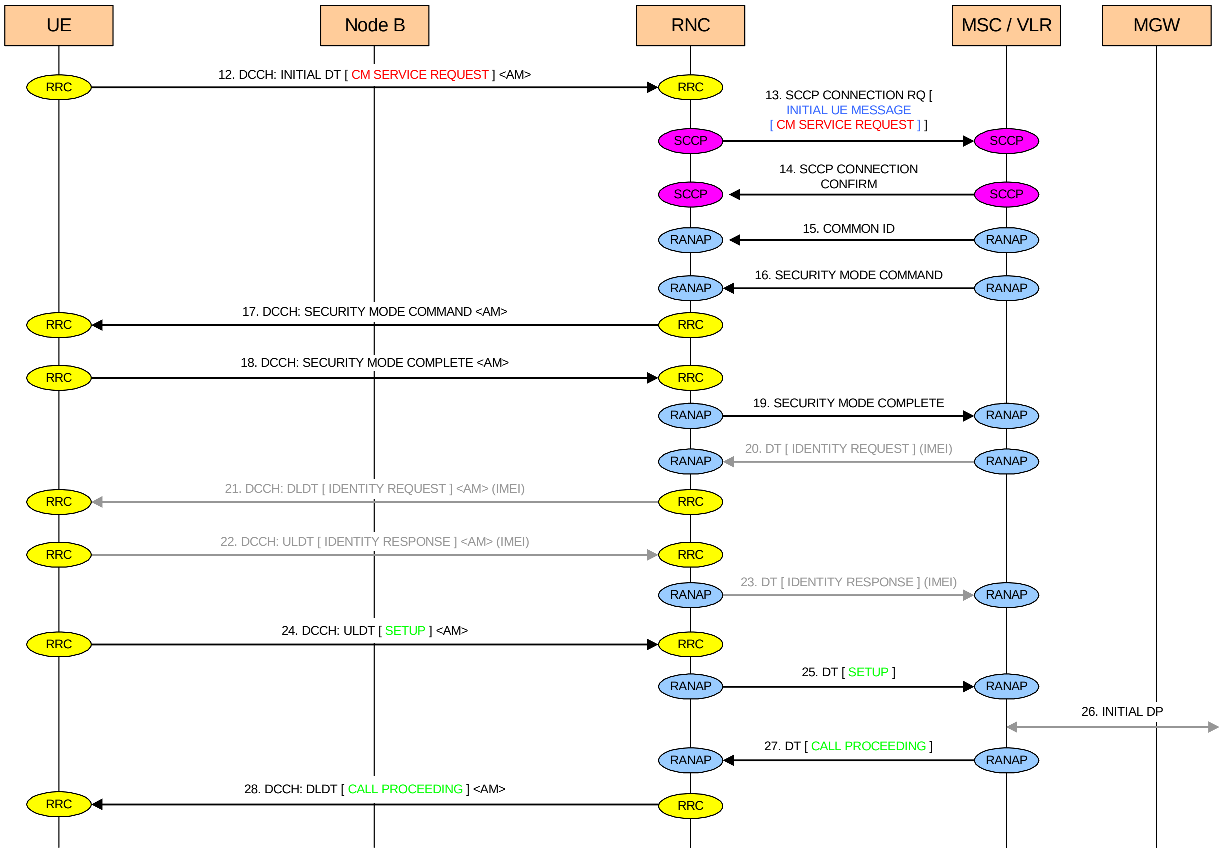
6.6.1.3 Trigger Points

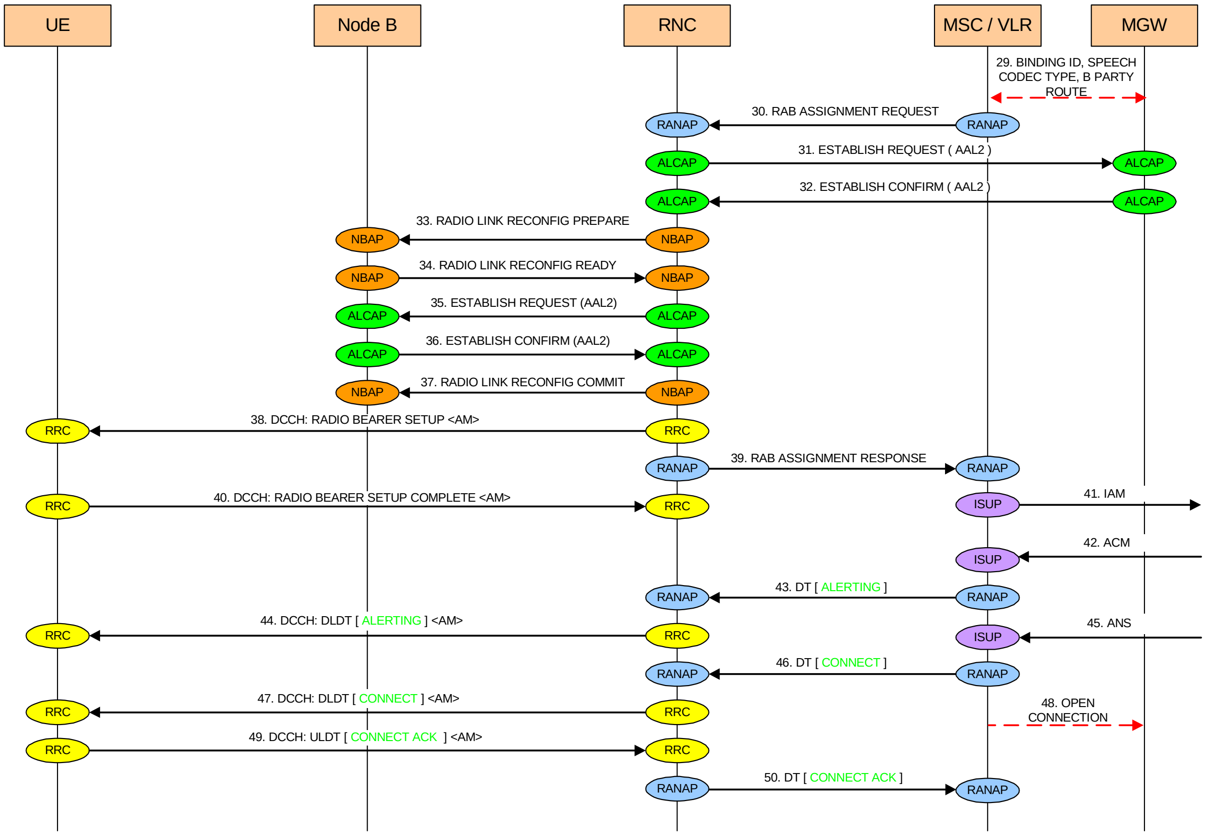
Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part over 3G
Call attempt	Start: Push Send button	Start: The first RRC CONNECTION REQUEST with Establishment Cause "Originating Conversational Call" message carried on the CCCH logical channel and mapped to the RACH transport channel is sent. (figure 23; signalling point number 1). Comment: It is possible that more than one RRC CONNECTION REQUEST message per call attempt is sent. Only the first RRC CONNECTION REQUEST with Establishment Cause "Originating Conversational Call" should be taken into account for the calculation. It is possible that the RRC connection is already established because of an e.g. Location Update, then the start trigger is not reachable. In this case the current test sample should be deleted.
Successful call attempt	Stop: Alerting tone is heard by the A-party coming from B-party AND B-party rings.	Stop: The ALERTING message on the DCCH logical channel is passed: 1. from the B-party to the MSC (uplink) AND 2. from the MSC to the A-party (downlink) to indicate that the A-party rings (Figure 2; signalling point number 4). 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.
Unsuccessful call attempt	Stop trigger point not reached	
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.		

Preconditions for measurement:

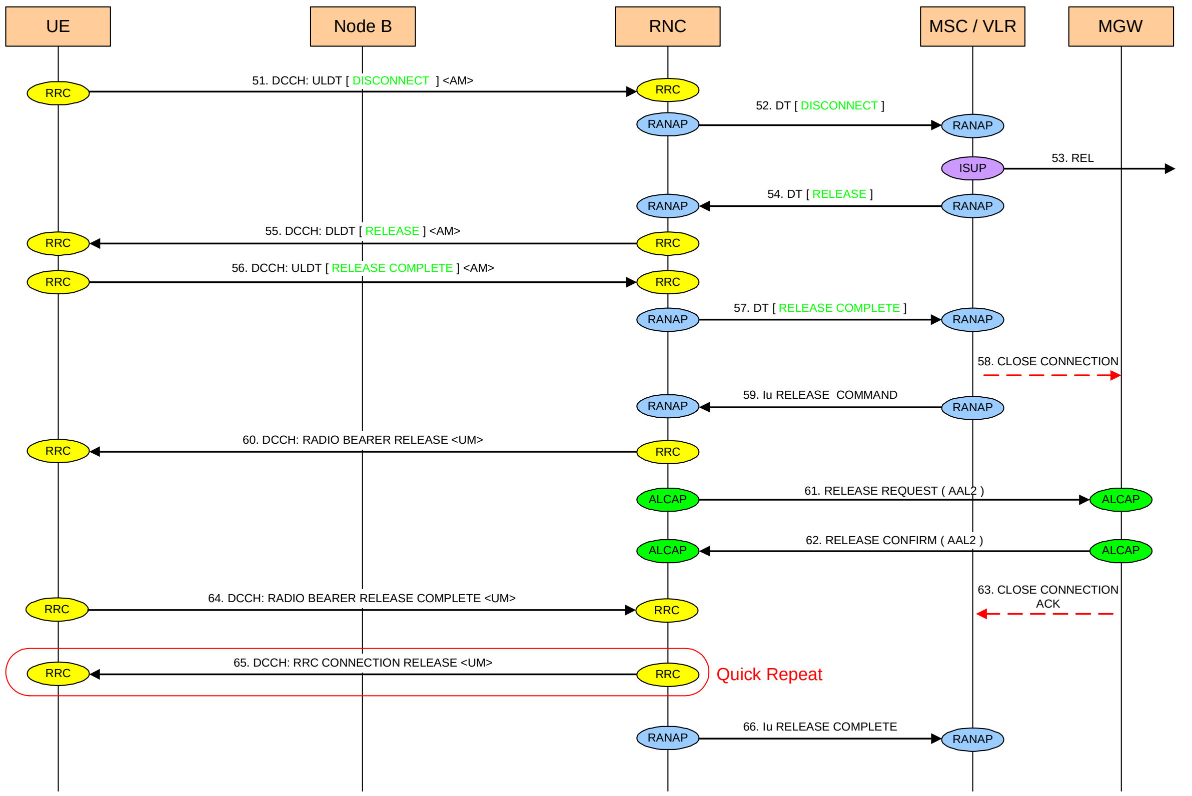
Precondition	Covered by	Reference document
CS network available	Radio Network Unavailability	
CS attach successful		
B-party shall not be busy		


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


UE
Node B
RNC
MSC / VLR
MGW
12. DCCH: INITIAL DT [CM SERVICE REQUEST] <AM>
13. SCCP CONNECTION RQ [INITIAL UE MESSAGE [CM SERVICE REQUEST]]
14. SCCP CONNECTION CONFIRM
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>
RRC
SCCP
RANAP



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]
RANAP
ALCAP
NBAP
RRC
ISUP



UE
Node B
RNC
MSC / VLR
MGW
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
RRC
RANAP
ISUP
ALCAP

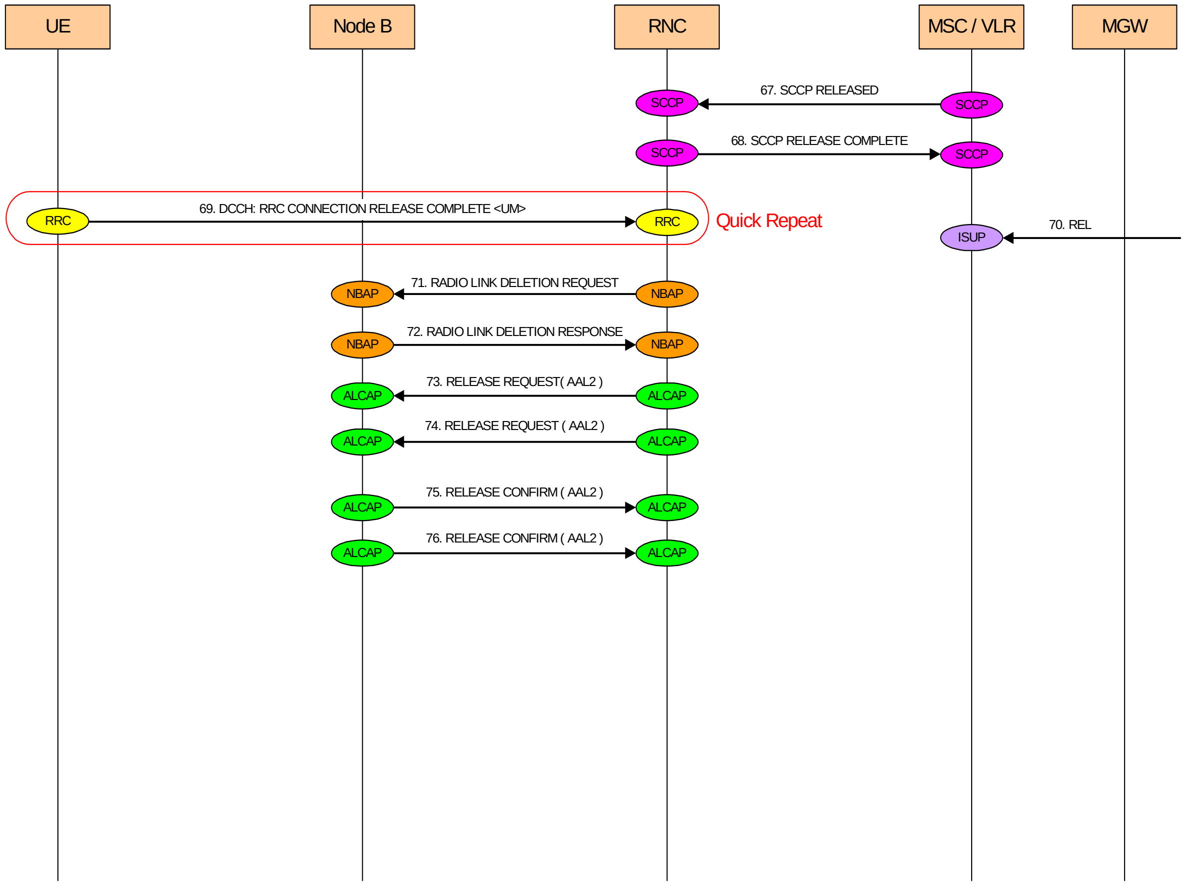


Figure 23: 3G Voice Signalling Flow Chart: Mobile Originated Call Establishment Procedure

6.6.2 Telephony Setup Time [s]

6.6.2.1 Abstract Definition

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

6.6.2.2 Abstract Equation

$$\boxed{\text{Telephony Setup Time [s]} = t_{\text{connect established}} - t_{\text{customer presses send button on UE}}}$$

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.

6.6.2.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part over 3G
Call Attempt	Start: Push Send button	Start: The first RRC CONNECTION REQUEST with Establishment Cause "Originating Conversational Call" message carried on the CCCH logical channel and mapped to the RACH transport channel is sent. (Figure 2; signalling point number 1). Comment: It is possible that more than one RRC CONNECTION REQUEST message per call attempt is sent. Only the first RRC CONNECTION REQUEST with Establishment Cause "Originating Conversational Call" should be taken into account for the calculation. It is possible that the RRC connection is already established because of an e.g. Location Update, then the start trigger is not reachable. In this case the current test sample should be deleted.
Connection established (Successful call attempt)	Stop: Alerting tone is heard by the A-party coming from the B-party AND B-party rings	Stop: The ALERTING message on the DCCH logical channel is passed: 3. from the B-party to the MSC (uplink) AND 4. from the MSC to the A-party (downlink) to indicate that the A-party rings (Figure 2; signalling point number 44). 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.
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.		

Preconditions for measurement:

Precondition	Covered by	Reference document
CS network available	Radio Network Unavailability	
CS attach successful		
CS service access successful	Telephony Service Non-Accessibility	

6.6.3 Telephony Speech Quality on Call Basis

6.6.3.1 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.

6.6.3.2 Abstract Equation

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].

$$\text{Telephony Speech Quality on Call Basis (received A-side)} = f(MOS\_{LQO})$$
$$\text{Telephony Speech Quality on Call Basis (received B-side)} = f(MOS\_{LQO})$$

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).

6.6.3.3 Trigger Points

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

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part over 3G
Not applicable.	Start: Interchange speech samples between a-party and b-party	Start: 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).
Not applicable.	Stop: Release of connection	Stop: 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).

6.6.4 Telephony Speech Quality on Sample Basis

6.6.4.1 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.

6.6.4.2 Abstract Equation

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.

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

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

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).

6.6.4.3 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.

6.6.5 Telephony Cut-off Call Ratio [%]

6.6.5.1 Abstract Definition

Probability that a successful call attempt is ended by a cause other than the intentional termination by A- or B-party.

6.6.5.2 Abstract Equation

$$\text{Telephony Cut-off Call Ratio [\%]} = \frac{\text{unintentionally terminated telephony calls}}{\text{all successful telefphony call attempts}} \times 100\ \%$$

6.6.5.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Successful telephony call attempt	Start: Alerting or busy tone heard by the A-party coming from B-party.	Start: 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	Stop: Release of connection directly by A- or B-party	Stop: 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).
Unintentionally terminated telephony call	Stop trigger point not reached.	

6.7 Video Telephony

6.7.1 Network Accessibility/Availability

Network availability and network accessibility are measured independently from the service, and will not be described further in this chapter. Network availability and network accessibility are pre-conditions for the performance of the measurement of QoS.

6.7.2 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.

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. The technical triggers are defined for positive successful cases, if the VT works fine. For failures the triggers are the opposite, this means the nonexistence of the message indicates the failure. The bold lines behind the trigger points tx are the used one and the dashed one are unused.

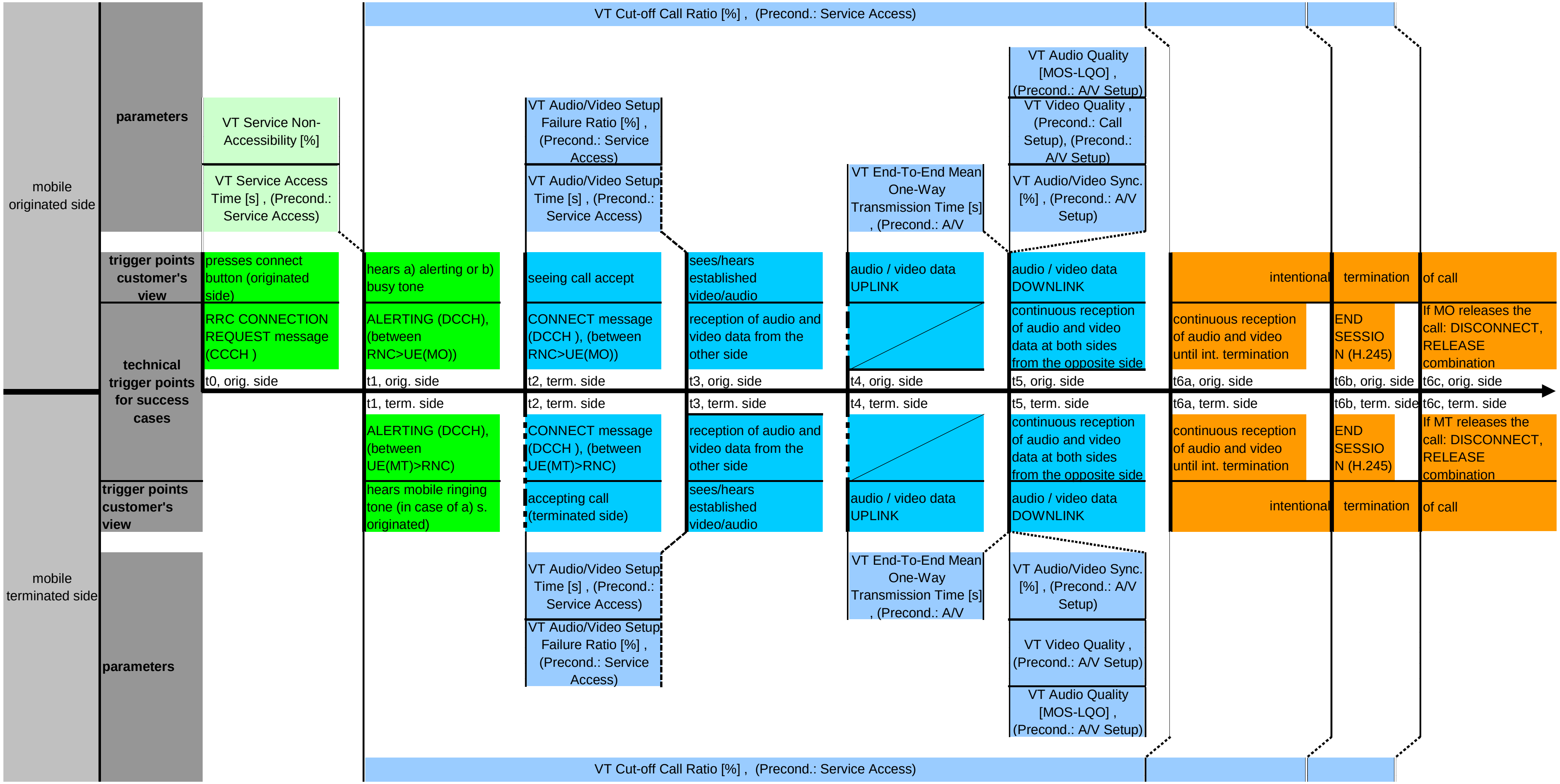


Figure 24: Parameter overview with trigger points

6.7.3 VT Service Non-Accessibility [%]

6.7.3.1 Abstract Definition

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

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

6.7.3.2 Abstract Equation

$$\text{VT Service Non-Accessibility [\%]} = \frac{\text{unsuccessful video telephony call access attempts}}{\text{all video telephony call access attempts}} \times 100\ \%$$

6.7.3.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Video Telephony call attempt	Start: Push Send button	Start: The first RRC CONNECTION REQUEST with Establishment Cause „originating conversational call message carried on the CCCH logical channel and mapped to the RACH transport channel is sent. (clause 6.7.13, signalling point number 1) Comment: It is possible that more than one RRC CONNECTION REQUEST message per call attempt is sent. Only the first RRC CONNECTION REQUEST with Establishment Cause „Originating Conversational Call should be taken into account for the calculation. It is possible that the RRC connection is already established because of an e.g. Location Update, then the start trigger is not reachable. In this case the current test sample should be deleted.
Successful Video Telephony call attempt	Stop: Alerting tone is heard by the MO side coming from the MT side AND MT side rings.	Stop: The ALERTING message on the DCCH logical channel is passed: 1. from the UE at MT side to MSC (uplink) AND 2. from the MSC to the UE at MO side (downlink) to indicate that the MT side rings. (clause 6.7.13, signalling point number 44)
Unsuccessful Video Telephony call attempt	Stop trigger point not reached.	Stop trigger point not reached.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio Network Unavailability	
UMTS CS attach successful		
MT side shall not be busy		

6.7.4 VT Service Access Time [s]

6.7.4.1 Abstract Definition

Time between pushing send button after input of MSISDN and receipt of alerting at MO side.

Remark(s): This parameter is not calculated unless the video telephony call access attempt is successful. At MT side the mobile shall ring.

6.7.4.2 Abstract Equation

$$\text{VT Service Access Time [s]} = t_{\text{alerting tone}} - t_{\text{push send button}}$$

6.7.4.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time of Video Telephony call Attempt	Start: Push Send button	Start: The first RRC CONNECTION REQUEST with Establishment Cause „originating conversational call message carried on the CCCH logical channel and mapped to the RACH transport channel is sent. (clause 6.7.13, signalling point number 1) Comment: It's possible that more than one RRC CONNECTION REQUEST message per call attempt is sent. Only the first RRC CONNECTION REQUEST with Establishment Cause „Originating Conversational Call should be taken into account for the calculation. It is possible that the RRC connection is already established because of an e.g. Location Update, then the start trigger is not reachable. In this case the current test sample should be deleted.
Time of successful Video Telephony call attempt	Stop: Alerting tone is heard by the MO side coming from the MT side AND MT side rings	Stop: The ALERTING message on the DCCH logical channel is passed: 1. from the UE at MT side to MSC (uplink) AND 2. from the MSC to the UE at MO side (downlink) to indicate that the MT side rings. (clause 6.7.13, signalling point number 44)

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio Network Unavailability	
UMTS CS attach successful		
UMST CS service access	VT Service Access Failure Ratio	

6.7.5 VT Audio/Video Setup Failure Ratio [%]

6.7.5.1 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).

6.7.5.2 Abstract Equation

$$\text{VT Audio/Video Setup Failure Ratio [\%]} = \frac{\text{audio/video setup failures}}{\text{all accepted calls at MT side}} \times 100\ \%$$

6.7.5.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Audio/video setup attempt	Start: MO sees the call acceptance from the MT side.	Start: The CONNECT message on the DCCH logical channel is passed from the MSC to the UE at MO side to indicate that the connection has been established. (clause 6.7.13, signalling point number 47)
Audio/Video Setup Success	Stop: Start of the audio and video output at both sides	Stop: Start of reception of audio and video data at both sides from the opposite side. Comment: All four data streams shall be received for a success.
Audio/Video Setup Failure	Stop trigger point not reached.	Stop trigger point not reached.

Preconditions of measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio Network Unavailability	
UMTS CS attach successful		
UMTS CS service access successful	VT Service Non-Accessibility	

6.7.6 VT Audio/Video Setup Time [s]

6.7.6.1 Abstract Definition

The elapsed time from the MT call acceptance indicated at MO 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).

6.7.6.2 Abstract Equation

$$\text{VT Audio/Video Setup Time [s]} = t_{\text{audio/video start}} - t_{\text{MT accept call}}$$

6.7.6.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time of beginning of audio/video setup	Start: MO sees the call acceptance from the MT side.	Start: The CONNECT message on the DCCH logical channel is passed from the MSC to the UE at MO side to indicate that the connection has been established. (clause 6.7.13, signalling point number 47)
Time of successful audio/video setup	Stop: Start of the audio and video output at both sides.	Stop: Start of reception of audio and video data at both sides from the opposite side + constant time value for decoding. Comment: All four data streams shall be received for a success.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio Network Unavailability	
UMTS CS attach successful		
UMTS CS audio/video setup successful	VT Audio/Video Setup Failure Ratio	

6.7.7 VT Cut-off Call Ratio [%]

6.7.7.1 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 the call acceptance fails after alerting;
- if audio/video setup fails; or
- if either the audio, the video or both are lost at one or both sides for an interruption timeout and before the end of "predefined call duration".

The "predefined call duration" is the difference between the indication of the call acceptance at MO side and the intentional release of the call.

6.7.7.2 Abstract Equation

$$\text{VT Cut-off Call Ratio [\%]} = \frac{\text{video telephony dropped calls}}{\text{all successful video telephony service access attemps}} \times 100\ \%$$

6.7.7.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Successful Video Telephony service access attempt	Start: Alerting tone is heard by the MO side coming from MT side AND MT side rings	Start: The ALERTING message on the DCCH logical channel is passed: 1. from the UE at MT side to MSC (uplink) AND 2. from the MSC to the UE at MO side (downlink) to indicate that the MT side rings. (clause 6.7.13, signalling point number 1)
Video Telephony successful call	Stop: No loss of video and/or audio without any intention by MO or MT side longer than the interruption timeout within the predefined call duration	Stop: 1. If the test system can capture audio / video information: Continuous reception of audio and video data at both sides from the opposite side without an interruption longer than the interruption timeout until intentional call release. 2. If the test system cannot capture audio / video information: The following information shall not be seen in signalling before intentional call release but they shall be seen after the intentional call release: • H.245 EndSession command (endSessionCommand disconnect) OR • the following trigger combination (all triggers on the DCCH logical channel): [M1: DISCONNECT (uplink).] AND [M2: DISCONNECT (downlink) or RELEASE (downlink)] (clause 6.7.13, signalling point number 51) Comment: In some cases the mobiles use not the EndSession command but only the DISCONNECT or RELEASE command
Video Telephony dropped calls	Stop trigger point not reached.	Stop trigger point not reached.

NOTE: If the reception of audio and/or video is interrupted shortly before the predefined call duration, then the call duration shall be extended to check if the interruption persists for the interruption timeout or not. If the interruption is shorter than the interruption timeout the call shall be released immediately and rated as success otherwise the sample shall be rated as failure and the call will be released.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio Network Unavailability	
UMTS CS attach successful		
UMTS CS service access successful	VT Service Non-Accessibility	

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

6.7.8.1 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.

Remark(s):

- This parameter is not calculated unless the VT audio/video setup attempt is successful.
- The speech quality measurement is taken per call. An aggregation for measurement campaigns or parts of it should be made on speech sample basis.
- The acoustic behaviour of terminals is not part of this audio quality measurement. The modeling of the acoustic part of the handset-terminals (e.g. frequency shaping) is incorporated in the speech quality assessment algorithm. Therefore the test mobiles used have to be connected at their electrical interfaces and not coupled acoustically. it has to be taken into account that a detailed way for insertion and capturing of audio signals is described in ITU-T Recommendation P.862.3 [19].
- For wideband (7 kHz) applications a standardized algorithm is available in ITU-T Recommendation P.862.2 [18].
- Evaluation of a MO DL or MT DL and also for these both directions (sum) is possible by calculating the mean value of the results from all samples.
- Experience has shown a high variable delay in video calls.
- ITU-T Recommendation P.862 is not approved for testing such video call applications. It has to be taken into account that further studies including auditory tests of video calls have to be conducted.

6.7.8.2 Computation

ITU-T Recommendation P.862 [1] (02/2001) together with the related mapping given in ITU-T Recommendation P.862.1 [9] (10/2003) is recommended. This algorithm describes the opinion of customers related to voice transmission quality (300 through 3400 Hz) and its connected impairments (background noise, unnatural voice, temporal clipping and interruptions etc).

The speech quality measurement is taken per call (the evaluation algorithm is currently under study in ETSI STQ MOBILE WG) and per direction (DL at MO, DL at MT).

After mapping the raw P.862 results according to Rec. P.862.1, the speech quality assessment is presented in a MOS-like scale between 1 and 5 called MOS Listening Quality Objective (MOS-LQO), as defined in ITU-T Rec. P.800.1.

6.7.8.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Successful Audio/Video Setup Attempt	Start: Start of the audio and video output at both sides	Start: Start of reception of audio and video data at both sides from the opposite side. Comment: All four data streams shall be received for a success.
End of call (only intentional)	Stop: End of call	Stop: End of continuous reception of audio and video data at both sides from the opposite side because of: • intentional call release.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio Network Unavailability	
UMTS CS attach successful		
UMTS CS service access successful	VT Service Non-Accessibility	
UMTS CS audio/video setup successful	VT Audio/Video Setup Failure Ratio	

6.7.9 VT Speech Quality on Sample Basis [MOS-LQO]

6.7.9.1 Abstract Definition

Indicator representing the quantification of the end-to-end speech transmission quality as perceived by the user. This parameter computes the speech quality on a sample basis.

Remark(s):

- This parameter is not calculated unless the VT audio/video setup attempt is successful.
- Speech quality values from all video telephony calls should be taken into consideration for statistical quality analysis
- The speech quality measurement is taken per sample. An aggregation for measurement campaigns or parts of it should be made on speech sample basis. Only complete received samples of a dropped call are evaluable.
- The acoustic behaviour of terminals is not part of this audio quality measurement. The modeling of the acoustic part of the handset-terminals (e.g. frequency shaping) is incorporated in the speech quality assessment algorithm. Therefore the test mobiles used have to be connected at their electrical interfaces and not coupled acoustically. it has to be taken into account that a detailed way for insertion and capturing of audio signals is described in the new ITU-T Recommendation P.862.3 [19].
- For wideband (7 kHz) applications a standardized algorithm is available in ITU-T Recommendation P.862.2 [18].
- Evaluation of a MO DL or MT DL and also for these both directions (sum) is possible by calculating the mean value of the results from all samples.
- Experience has shown a high variable delay in video calls.
- P.862 is not approved for testing such video call applications. It has to be taken into account that further studies including auditory tests of video calls have to be conducted.

6.7.9.2 Abstract Equation

$$\boxed{f(Speech Quality\ Assessment\ \mathrm{Alg}orithm, \{x_1,...,x_n\}, RL) = \{y_1,...,y_n\}[MOS - LQO]}$$

where:

- Speech Quality Assessment Algorithm: is the algorithm applied;
- $\{x_1,...,x_n\}$: are the completed speech samples belonging to both completed and dropped calls;
- $\{y_1,...,y_n\}$:are the results generated by the algorithm;
- RL: Radio Link (DL at MO, DL at MT, sum).

ITU-T Recommendation P.862 [1] (02/2001) together with the related mapping given in ITU-T Recommendation P.862.1 [9] (10/2003) is recommended. This algorithm describes the opinion of customers related to voice transmission quality (300 through 3400 Hz) and its connected impairments (background noise, unnatural voice, temporal clipping and interruptions etc).

The speech quality measurement is taken per sample and per direction (DL at MO, DL at MT).

After mapping the raw P.862 results according to Recommendation P.862.1 [9], the speech quality assessment is presented in a MOS–like scale between 1 and 5 called MOS Listening Quality Objective (MOS-LQO), as defined in ITU-T Recommendation P.800.1.

6.7.9.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Successful Audio/Video Setup Attempt	Start: Start of the audio and video output at both sides	Start: Start of reception of audio and video data at both sides from the opposite side. Comment: All four data streams shall be received for a success.
End of call (intentional or dropped)	Stop: End of call	Stop: End of continuous reception of audio and video data at both sides from the opposite side because of: • an interruption for a predefined duration or longer OR • intentional call release.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio Network Unavailability	
UMTS CS attach successful	Attach Failure Ratio	
UMTS CS service access successful	VT Service Non-Accessibility	
UMTS CS audio/video setup successful	VT Audio/Video Setup Failure Ratio	

6.7.10 VT Video Quality

6.7.10.1 Abstract Definition

End-to-end quality of the video signal as perceived by the end user during a VT call. This parameter computes the video quality on a sample basis.

Remark(s):

- This parameter is not calculated unless the VT audio/video setup attempt is successful.
- Video quality values from all video telephony calls should be taken into consideration for statistical quality analysis.
- The video quality measurement is taken per sample. An aggregation for measurement campaigns or parts of it should be made on video sample basis. Only complete received samples of a dropped call are evaluable.
- Evaluation of a MO DL or MT DL and also for these both directions (sum) is possible by calculating the mean value of the results from all samples.

6.7.10.2 Abstract Equation

To be specified.

6.7.10.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Successful audio/video setup attempt	Start: Start of the audio and video output at both sides	Start: Start of reception of audio and video data at both sides from the opposite side. Comment: All four data streams shall be received for a success.
End of call (intentional or dropped)	Stop: End of call	Stop: End of continuous reception of audio and video data at both sides from the opposite side because of: • an interruption for a predefined duration or longer OR • intentional call release

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio Network Unavailability	
UMTS CS attach successful	Attach Failure Ratio	
UMTS CS service access successful	VT Service Non-Accessibility	
UMTS CS audio/video Setup successful	VT Audio/Video Setup Failure Ratio	

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

6.7.11.1 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.

6.7.11.2 Abstract Equation

Time from input of the signal at MS (MO/MT) to output at MS (MT/MO).

Aggregation Algorithm: ((Transmission Time MO ->MT) + (Transmission Time MT->MO))/2.

In case of a symmetrical channel one party could be configured as loopback device. The other one can determine the double delay by correlating transmit and receive signal. The delay should be measured after the loopback at the top of the radio bearer.

As the delay of the codec is almost constant for a specific mobile implementation, the codec delay could be considered by a mobile depending offset. In each direction one shall add the encoder and the decoder times. For the whole loopback one shall calculate the following times:

MO>MT	Encoding of audio / video (slowest is used)	a
	Transmission of audio / video (slowest is used)	b
	Decoding of audio / video (slowest is used)	c
MT>MO	Encoding of audio / video (slowest is used)	d
	Transmission of audio / video (slowest is used)	e
	Decoding of audio / video (slowest is used)	f

$$\text{VT End-To-End Mean One-Way Transmission Time } [s] = \frac{a+b+c+d+e+f}{2}$$

6.7.11.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Successful audio/video setup attempt	Start: Start of the audio and video output at both sides	Start: Start of reception of audio and video data at both sides from the opposite side. Comment: All four data streams shall be received for a success.
End of call (intentional or dropped)	Stop: End of call	Stop: End of continuous reception of audio and video data at both sides from the opposite side because of: • an interruption for a predefined duration or longer OR • intentional call release.

Preconditions for measurement:

Precondition	Covered by	Reference document
UMTS CS available	Radio Network Unavailability	
UMTS CS attach successful	Attach Failure Ratio	
UMTS CS service access successful	VT Service Non-Accessibility	
UMTS CS audio/video Setup successful	VT Audio/Video Setup Failure Ratio	

6.7.12 VT Audio/Video Synchronization [%]

6.7.12.1 Abstract Definition

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

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.

6.7.12.2 Abstract Equation

To be specified.

6.7.12.3 Trigger Points

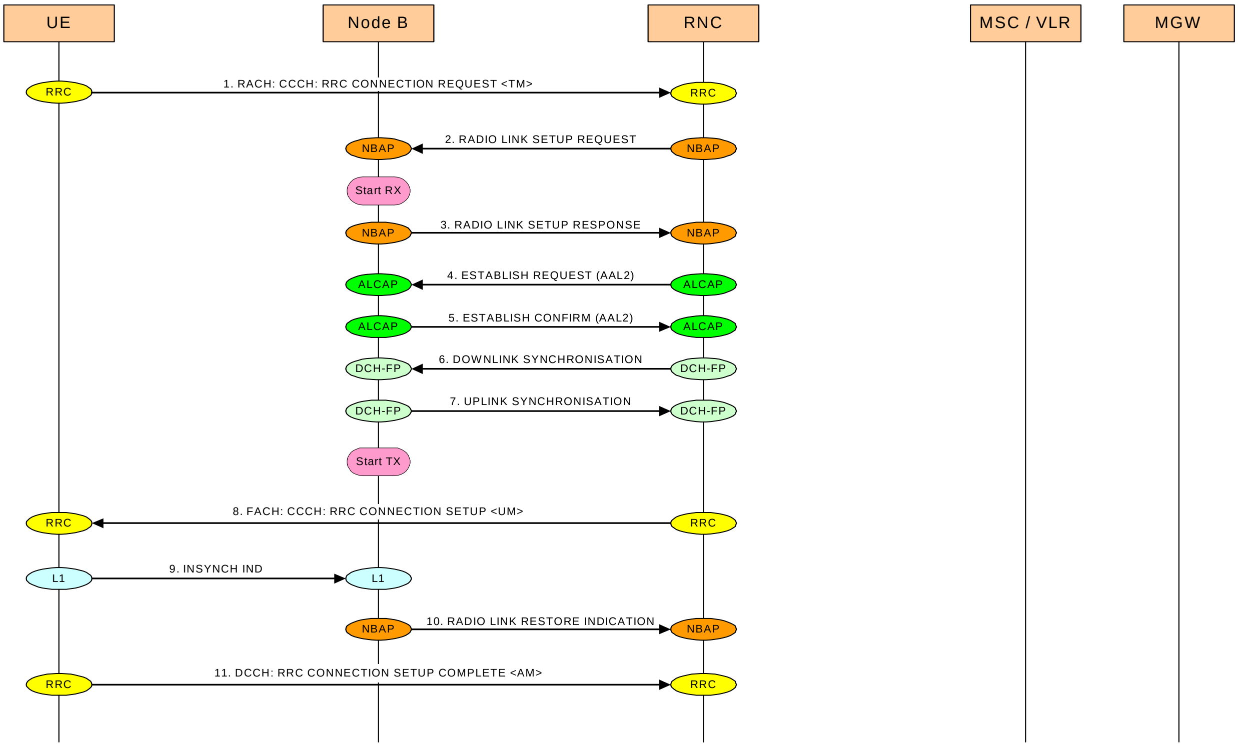
Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Successful audio/video setup attempt	Start: Start of the audio and video output at both sides	Start: Start of reception of audio and video data at both sides from the opposite side. Comment: All four data streams shall be received for a success.
End of call (intentional or dropped)	Stop: End of call	Stop: End of continuous reception of audio and video data at both sides from the opposite side because of: • an interruption for a predefined duration or longer OR • intentional call release.

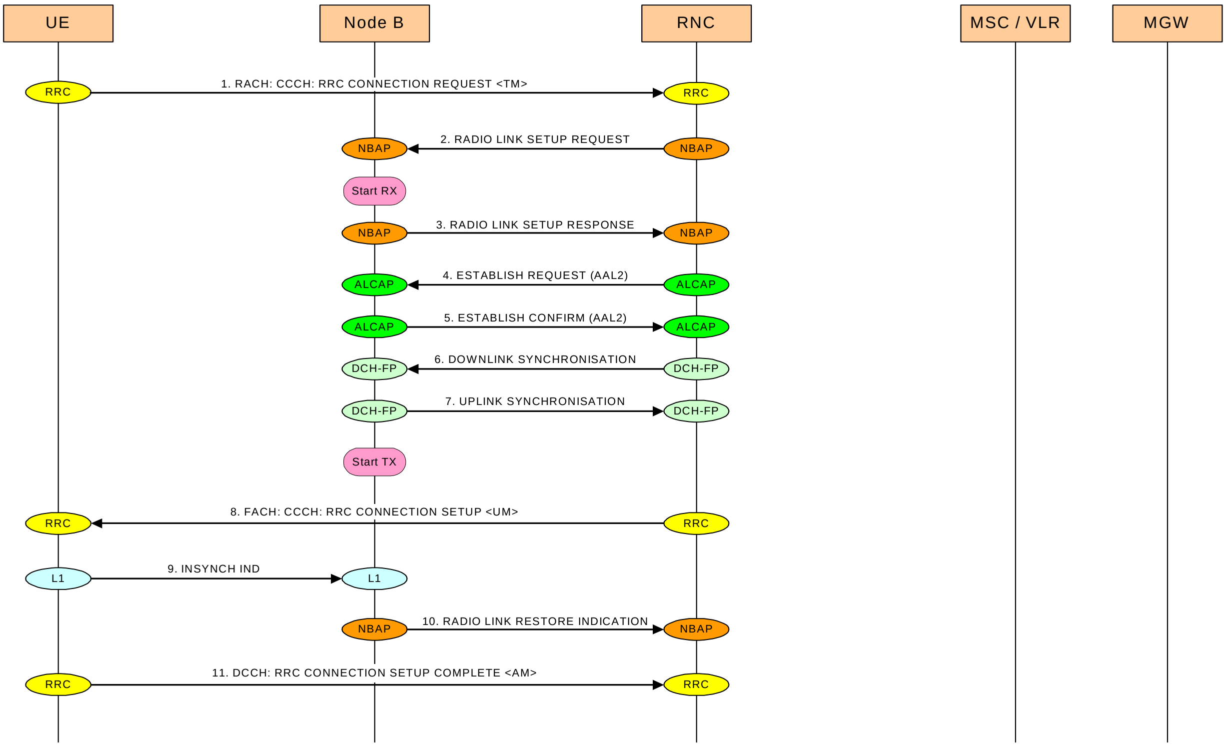
Preconditions for measurement:

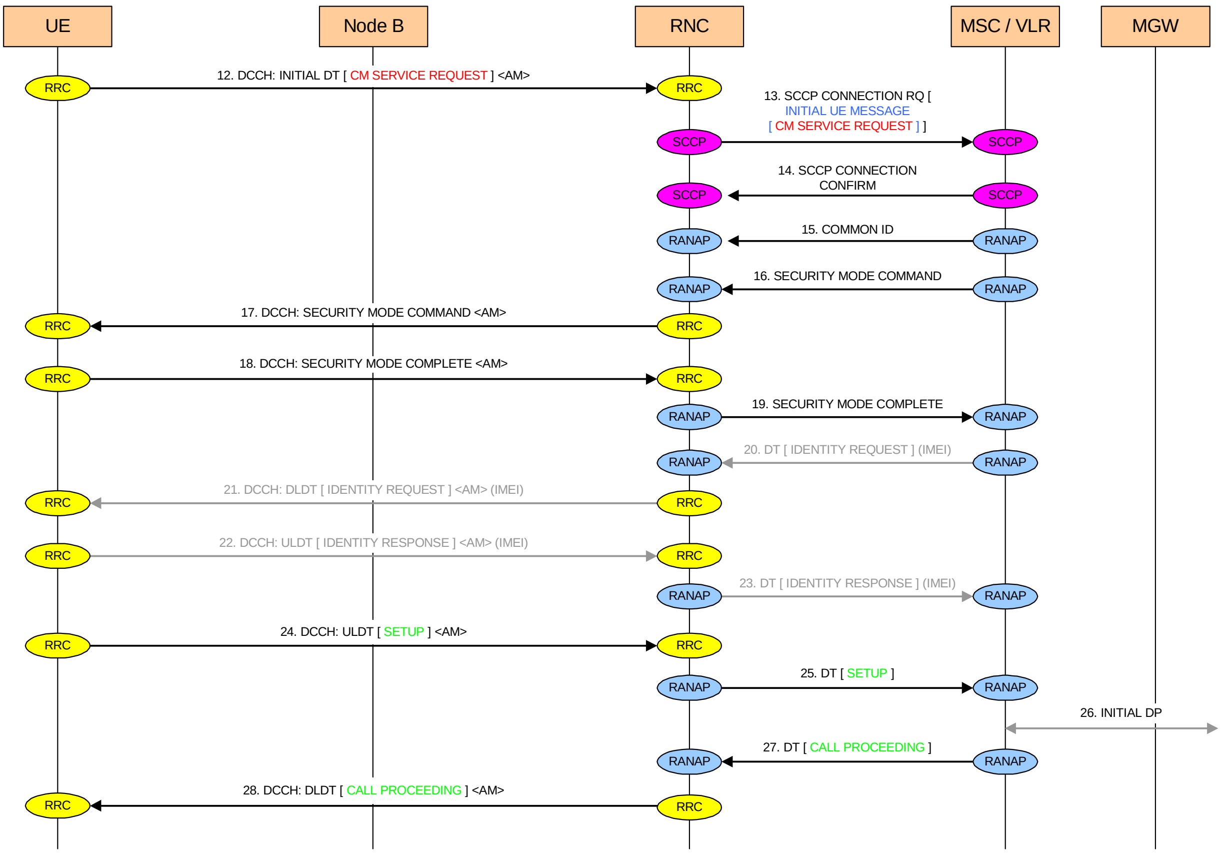
Precondition	Covered by	Reference document
UMTS CS available	Radio Network Unavailability	
UMTS CS attach successful	Attach Failure Ratio	
UMTS CS service access successful	VT Service Non-Accessibility	
UMTS CS audio/video Setup successful	VT Audio/Video Setup Failure Ratio	

6.7.13 Signalling Diagrams

These are the flow charts of a mobile originated call until the call release. The point of view is the MO side.




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


UE
Node B
RNC
MSC / VLR
MGW
12. DCCH: INITIAL DT [CM SERVICE REQUEST] <AM>
13. SCCP CONNECTION RQ [
INITIAL UE MESSAGE
[CM SERVICE REQUEST]]
14. SCCP CONNECTION
CONFIRM
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>
RRC
SCCP
RANAP

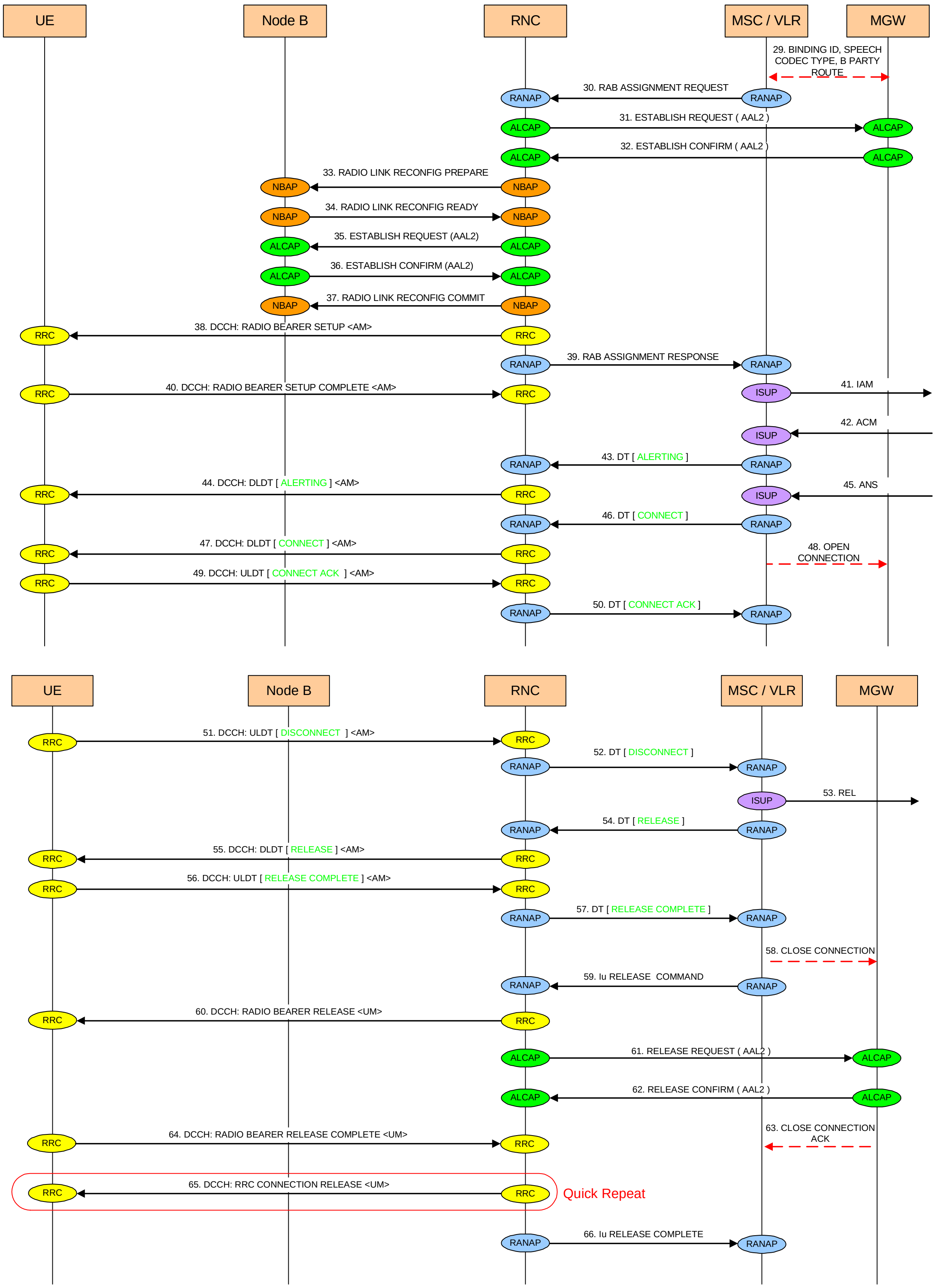


Figure 25: Parameter overview with trigger points

6.8 Web Browsing (HTTP)

6.8.1 HTTP Service Non-Accessibility [%]

6.8.1.1 Abstract Definition

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

6.8.1.2 Abstract Equation

$$\text{HTTP Service Non-Accessibility [\%]} = \frac{\text{unsuccessful attempts to reach the point when content is received}}{\text{all attempts to reach the point when content is sent or received}} \times 100\ \%$$

6.8.1.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates the service access.	Start: ATD command.
Successful attempt	Stop: First content is received.	Stop Method A: Reception of the first data packet containing content. Stop Method B: Sending of the first GET command.
Unsuccessful attempt	Stop trigger point not reached.	

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

6.8.2 HTTP Setup Time [s]

6.8.2.1 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.

6.8.2.2 Abstract Equation

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

6.8.2.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates the service access.	Start: ATD command.
Time when content is received	Stop: First content is received.	Stop Method A: Reception of the first data packet containing content. Stop Method B: Sending of the first GET command.

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

6.8.3 HTTP IP-Service Access Failure Ratio [%]

6.8.3.1 Abstract Definition

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

6.8.3.2 Abstract Equation

$$\text{HTTP IP-Service Access Failure Ratio [\%]} = \frac{\text{unsuccessful attempts to establish an IP connection to the server}}{\text{all attempts to establish an IP connection to the server}} \times 100\ \%$$

6.8.3.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer enters the URL and hits "Return".	Start: First [SYN] sent.
Successful attempt	Stop: Web page download starts.	Stop Method A: Reception of the first data packet containing content. Stop Method B: Sending of the first GET command.
Unsuccessful attempt	Stop trigger point not reached.	

Remark(s)**:** The PS bearer has to be active in the cell used by a subscriber (cf. Radio Network 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).

6.8.4 HTTP IP-Service Setup Time [s]

6.8.4.1 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.

6.8.4.2 Abstract Equation

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

6.8.4.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer enters the URL and hits "Return".	Start: First [SYN] sent.
Time when content is received	Stop: Web page download starts.	Stop Method A: Reception of the first data packet containing content. Stop Method B: Sending of the first GET command.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Radio Network 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).

6.8.5 HTTP Session Failure Ratio [%]

6.8.5.1 Abstract Definition

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

6.8.5.2 Abstract Equation

$$\text{HTTP Session Failure Ratio [\%]} = \frac{\text{uncompleted sessions}}{\text{successfully started sessions}} \times 100\ \%$$

6.8.5.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer enters the URL and hits "Return".	Start: First [SYN] sent.
Successful attempt	Stop: The complete web page appears in the browser window.	Stop: Reception of the last data packet containing content.
Unsuccessful attempt	Stop trigger point not reached.	

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Radio Network 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).

6.8.6 HTTP Session Time [s]

6.8.6.1 Abstract Definition

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

6.8.6.2 Abstract Equation

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

6.8.6.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer enters the URL and hits "Return".	Start: First [SYN] sent.
Time when content is received	Stop: The complete web page appears in the browser window.	Stop: Reception of the last data packet containing content.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Radio Network 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).

6.8.7 HTTP Mean Data Rate [kbit/s]

6.8.7.1 Abstract Definition

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

6.8.7.2 Abstract Equation

$$\text{HTTP Mean Data Rate [\%]} = \frac{\text{User data transferred [kbit]}}{\left(t_{\text{content received}} - t_{\text{dial-up connection initiated}}\right)[s]} \times 100\ \%$$

6.8.7.3 Trigger Points

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

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Web page download starts.	Start Method A: Reception of the first data packet containing content. Start Method B: Sending of the first GET command.
Time when content is received	Stop: Web page download successfully completed.	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).

6.8.8 HTTP Data Transfer Cut-off Ratio [%]

6.8.8.1 Abstract Definition

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

6.8.8.2 Abstract Equation

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

6.8.8.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Web page download starts.	Start Method A: Reception of the first data packet containing content. Start Method B: Sending of the first GET command.
Time when content is received	Stop: Web page download successfully completed.	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).

7 Store-and-forward (S&F) Services QoS Parameters

7.1 E-Mail

7.1.1 E-Mail {Download|Upload} Service Non-Accessibility [%]

7.1.1.1 Abstract Definition

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

7.1.1.2 Abstract Equation

$$\text{E-Mail \{Download | Upload\} Service Non-Accessibility [\%]} = \frac{\text{unsuccessful attempts to reach the point when content is sent or received}}{\text{all attempts to reach the point when content is sent or received}} \times 100\ \%$$

7.1.1.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates the service access.	Start: ATD command.
Successful attempt	Stop: First content is received.	Stop Method A: Reception of the first data packet containing content. Stop Method B: Send RETR command.
Unsuccessful attempt	Stop trigger point not reached.	

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates the service access.	Start: ATD command.
Successful attempt	Stop: First content is sent.	Stop Method A: Sending of the first data packet containing content. Stop Method B: 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.
Unsuccessful attempt	Stop trigger point not reached.	

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

7.1.2 E-Mail {Download|Upload} Setup Time [s]

7.1.2.1 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.

7.1.2.2 Abstract Equation

$$\boxed{\text{E-Mail}\{\text{Download}|\text{Upload}\}\text{Setup Time}[s] = t_{\text{Content sent or received}} - t_{\text{Dial-up connection initiated}}}$$

7.1.2.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates the service access.	Start: ATD command.
Time when content is received	Stop: First content is received.	Stop Method A: Reception of the first data packet containing content. Stop Method B: Send RETR command.

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates the service access.	Start: ATD command.
Time when content is sent	Stop: First content is sent.	Stop Method A: Sending of the first data packet containing content. Stop Method B: 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.

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

7.1.3 E-Mail {Download|Upload} IP-Service Access Failure Ratio [%]

7.1.3.1 Abstract Definition

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

7.1.3.2 Abstract Equation

$$\text{E-Mail \{Download | Upload\} IP-Service Access Failure Ratio [\%]} = \frac{\text{unsuccessfull attempts to establish an IP connection to the server}}{\text{all attempts to establish an IP connection to the server}} \times 100\ \%$$

7.1.3.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates e-Mail download.	Start: First [SYN] sent.
Successful attempt	Stop: E-Mail download starts.	Stop Method A: Reception of the first data packet containing content. Stop Method B: Send RETR command.
Unsuccessful attempt	Stop trigger point not reached.	

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates e-Mail upload.	Start: First [SYN] sent.
Successful attempt	Stop: E-Mail upload starts.	Stop Method A: Sending of the first data packet containing content. Stop Method B: 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.
Unsuccessful attempt	Stop trigger point not reached.	

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Radio Network 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).

7.1.4 E-Mail {Download|Upload} IP-Service Setup Time [s]

7.1.4.1 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.

7.1.4.2 Abstract Equation

$$\text{E - Mail}\{\text{Download}\,|\,\text{Upload}\}\text{IP - Service Setup Time}\,[s] = t_{\text{Content sent or received}} - t_{\text{Query sent}}$$

7.1.4.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates e-Mail download.	Start: First [SYN] sent.
Time when content is received	Stop: E-Mail download starts.	Stop Method A: Reception of the first data packet containing content. Stop Method B: Send RETR command.

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates e-Mail upload.	Start: First [SYN] sent.
Time when content is sent	Stop: E-Mail upload starts.	Stop Method A: Sending of the first data packet containing content. Stop Method B: 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.

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Radio Network 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).

7.1.5 E-Mail {Download|Upload} Session Failure Ratio [%]

7.1.5.1 Abstract Definition

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

7.1.5.2 Abstract Equation

$$\text{E - Mail}\{\text{Download}\,|\,\text{Upload}\}\text{Session Failure Ratio}\,[\%] = \frac{\text{uncompleted sessions}}{\text{successfully started sessions}} \times 100\,\%$$

7.1.5.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates e-Mail download.	Start: First [SYN] sent.
Successful attempt	Stop: E-Mail download successfully completed.	Stop Method A: Reception of the last data packet containing content. Stop Method B: Reception of the last data packet containing the finish sequence (CRLF.CRLF).
Unsuccessful attempt	Stop trigger point not reached.	

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Service access attempt	Start: Customer initiates e-Mail upload.	Start: First [SYN] sent.
Successful attempt	Stop: E-Mail upload successfully completed.	Stop Method A: Reception of the [FIN, ACK] for the last data packet containing content. Stop Method B: Reception of the positive acknowledgement (250) for the EOM command.
Unsuccessful attempt	Stop trigger point not reached.	

Remark(s): The PS bearer has to be active in the cell used by a subscriber (cf. Radio Network 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).

7.1.6 E-Mail {Download|Upload} Session Time [s]

7.1.6.1 Abstract Definition

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

7.1.6.2 Abstract Equation

$$\boxed{\text{E - Mail}\,\{\text{Download}\,|\,\text{Upload}\}\,\text{Session Time}\,[\text{s}] = t_{\text{Session end}} - t_{\text{Session start}}}$$

7.1.6.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates e-Mail download.	Start: First [SYN] sent.
Time when content is received	Stop: E-Mail download successfully completed.	Stop Method A: Reception of the last data packet containing content. Stop Method B: Reception of the last data packet containing the finish sequence (CRLF.CRLF).

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: Customer initiates e-Mail upload.	Start: First [SYN] sent.
Time when content is sent	Stop: E-Mail upload successfully completed.	Stop Method A: Reception of the [FIN, ACK] for the last data packet containing content. Stop Method B: 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. Radio Network 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).

7.1.7 E-Mail {Download|Upload} Mean Data Rate [kbit/s]

7.1.7.1 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.

7.1.7.2 Abstract Equation

$$\text{E-Mail}\,\{\text{Download}\mid\text{Upload}\}\,\text{Mean Data Rate}[\%] = \frac{\text{User data transferred}\,[\text{kbit}]}{\left(t_{\text{content sent or received}} - t_{\text{dial-up connection initiated}}\right)[s]} \times 100\%$$

7.1.7.3 Trigger Points

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

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: E-Mail download starts.	Start Method A: Reception of the first data packet containing content. Start Method B: Send RETR command.
Time when content is received	Stop: E-Mail download successfully completed.	Stop Method A: Reception of the last data packet containing content. Stop Method B: Reception of the data packet containing the finish sequence (CRLF.CRLF).

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: E-Mail upload starts.	Start Method A: Sending of the first data packet containing content. Start Method B: 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.
Time when content is sent	Stop: E-Mail upload successfully completed.	Stop Method A: Reception of the [FIN, ACK] for the last data packet containing content. Stop Method B: Reception of the positive acknowledgement (250) for the EOM command.

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).

7.1.8 E-Mail {Download|Upload} Data Transfer Cut-off Ratio [%]

7.1.8.1 Abstract Definition

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

7.1.8.2 Abstract Equation

$$\text{E-Mail \{Download | Upload\} Data Transfer Cut-off Ratio [\%]} = \frac{\text{incomplete data transfers}}{\text{successfully started data transfers}} \times 100\ \%$$

7.1.8.3 Trigger Points

Download:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: E-Mail download starts.	Start Method A: Reception of the first data packet containing content. Start Method B: Send RETR command.
Time when content is received	Stop: E-Mail download successfully completed.	Stop Method A: Reception of the last data packet containing content. Stop Method B: Reception of the data packet containing the finish sequence (CRLF.CRLF).

Upload:

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Time when dial-up connection is initiated	Start: E-Mail upload starts.	Start Method A: Sending of the first data packet containing content. Start Method B: 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.
Time when content is sent	Stop: E-Mail upload successfully completed.	Stop Method A: Reception of the [FIN, ACK] for the last data packet containing content. Stop Method B: Reception of the positive acknowledgement (250) for the EOM command.

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).

7.2 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]).

7.2.1 MMS Send Failure Ratio [%]

7.2.1.1 Abstract Definition

The parameter MMS Send Failure Ratio 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".

7.2.1.2 Abstract Equation

$$\text{MMS Send Failure Ratio [\%]} = \frac{\text{unsuccessful MMS send attempts}}{\text{all MMS send attempts}} \times 100\ \%$$

7.2.1.3 Trigger Points

Event from abstract equation	**Trigger point from customer's point of view**	**Technical description / protocol part**
MMS Send Attempt	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 26).
Unsuccessful MMS Send Attempt	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 26). 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).

MMS Transmission Signalling Diagram
The diagram shows the transmission of a MS from MO to MT, the diagram is layer comprehensive

	MO		Network MT	
1	o------------activate pdp context REQUEST----------->>>	2		
4	<<<--------activate pdp context ACCEPT-----------------o	3		
5	o----------------wsp connect REQUEST--------------->>>	6		
8	<<<----------------wsp connect REPLY-------------------o	7		
9	o---------------------------wtp ACK---------------------->>>	10		
11	o---------------------MMS m-send.req------------------->>>	12		
14	<<<-----------------------wtp ACK-------------------------o	13		
15	o----------------------MMS m-send.req----------------->>>	16		
18	<<<-------------------MMS m-send.conf-------------------o	17		
19	o----------------------------wtp ACK--------------------->>>	20		
21	o----------------------wsp DISCONNECT--------------->>>	22		
24		23		
26	<<<-----------------------wtp ACK------------------------o	25		
		27	o---------------------MMS m-notification.ind------------->>>	28
		30	<<<--------activate pdp context REQUEST----------------o	29
		31	o-----------activate pdp context ACCEPT--------------->>>	32
		34	<<<-----------------wsp connect REQUEST---------------o	33
		35	o----------------wsp connect REPLY------------------->>>	36
		38	<<<---------------------------wtp ACK-----------------------o	37
		40	<<<-----------WSP/HTTP Get REQUEST-------------------o	39
		41	o------------------------wtp ack--------------------------->>>	42
		43	o-----------------------m-retrieve.conf------------------->>>	44
		46	<<<-----------------------wtp ACK--------------------------o	45
		47	o------------------------m.retrieve.conf------------------>>>	48
		50	<<<-------------------m-notifyResp.ind----------------------o	49
		51	o----------------------------wtp ACK---------------------->>>	52
		54	<<<-----------------------wsp DISCONNECT---------------o	53

Figure 26: MMS Transaction flow

7.2.2 MMS Retrieval Failure Ratio [%]

7.2.2.1 Abstract Definition

The parameter MMS Retrieval Failure Ratio describes the probability that the MMS-message can not be downloaded by the MT mobile, which received a MMS Notification before.

Remark(s)**:** 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".

7.2.2.2 Abstract Equation

$$\text{MMS Delivery Failure Ratio [\%]} = \frac{\text{unsuccessful MMS delivery attempts}}{\text{all MMS delivery attempts}} \times 100\ \%$$

7.2.2.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
MMS Retrieval Attempt (MT)	Start: Initiation of the Wap Get Request MT	Start: After the *m-Notification.ind*. (see [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 26).
Unsuccessful MMS Retrieval Attempt (MT)	Stop: No MMS-message is received	Stop: The *m-notifyResp.ind* (see [2]) is not sent by the MS (MT). (See trigger 49 in figure 26). 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).

7.2.3 MMS Send Time [s]

7.2.3.1 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.

7.2.3.2 Abstract Equation

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

7.2.3.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
$t_{MMStoMMSCcomplete}$	Start: MMS-message is completely transmitted to MMS-C	Start: 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 26). 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.
$t_{sendbutton}$	Stop: Send button is pushed	Stop: 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 26). "MMS unsuccessful send transfer timeout" as specified in TS 102 250-5 (see bibliography).

7.2.4 MMS Retrieval Time [s]

7.2.4.1 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.

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.

7.2.4.2 Abstract Equation

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

7.2.4.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
$t_{MMSfromMMSCcomplete}$	Start: MMS-message is received completely	Start: The *m-notifyResp.Ind* (see [2]) is sent by the MS (MT). (See trigger 49 in figure 26). 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).
$t_{initWGR}$	Stop: Time when WAP Get Request is initiated	Stop: The *m-Notification.ind* (see [2] is delivered to the MS (MT). This initiates the *PDP context activation.* (See trigger 29 in figure 26).

7.2.5 MMS Notification Failure Ratio [%]

7.2.5.1 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.

7.2.5.2 Abstract Equation

$$\text{MMS Notification Failure Ratio [\%]} = \frac{\text{failed MMS-notifications}}{\text{successful submitted MMS}} \times 100\ \%$$

7.2.5.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
Successful submitted MMS MO	Start: Reception of the acknowledgement from the MMS-C MO (i.e. "Message sent")	Start: 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 26). 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	Stop: Failure delivery (non-delivery) of the Notification - SMS	Stop: *m- notification.ind* (see [2]) is not delivered to the MS(MT). (See trigger 28 in figure 26). 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).

7.2.6 MMS Notification Time [s]

7.2.6.1 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.

7.2.6.2 Abstract Equation

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

7.2.6.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
$t_{MMSsubmit}$	Start: The MMS is submitted successfully	Start: 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 26). 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.
$t_{recNotif}$	Stop: Time when the Notification is received (MT)	Stop: *m-Notif.ind* (see [2]) is received by MS (MT) (See trigger 28 in figure 26). 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).

7.2.7 MMS End-to-End Failure Ratio [%]

7.2.7.1 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.

7.2.7.2 Abstract Equation

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

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

7.2.7.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
MMS Send Attempt by MS(MO)	Start: Pushing of send button	Start: The send button initiates the *PDP context activation* of the MS, followed by a connection to the WAP Gateway. (See trigger 1 in figure 26). 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)	Stop: No MMS-message is received (MT) or no acknowledgement from the MMSC is received at MS (MO).	Stop: The *m-send.conf* (where Response Status: $80 = M_RS_OK) is not received by the MS(MO). (See trigger 18 in figure 26) or the *m-notifyResp. ind* (see [2]) (see is not sent by the MS (MT)). (See trigger 18 and 49 in figure 26). 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).

7.2.8 MMS End-to-End Delivery Time [s]

7.2.8.1 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.

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.

7.2.8.2 Abstract Equation

$$\boxed{\text{MMS End-to-End Delivery Time [s]} = t_{MMSrec} - t_{sendAttempt}}$$

7.2.8.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
$t_{sendattemot}$	Start: Time when the "send button" is pushed	Start: The send button initiates the *PDP context activation* of the MS (MO), followed by a connection to the WAP Gateway. (See trigger 1 in figure 26). 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}	Stop: Time when the MMS is received at the b-parties mobile	Stop: 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 26). 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 26). 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).

7.3 Short Message Service (SMS)

7.3.1 SMS Service Non-Accessibility MO [%]

7.3.1.1 Abstract Definition

Probability that the end-customer cannot access the Short Message Service when requested while it is offered by display of the network indicator on the Mobile Equipment.

7.3.1.2 Abstract Equation

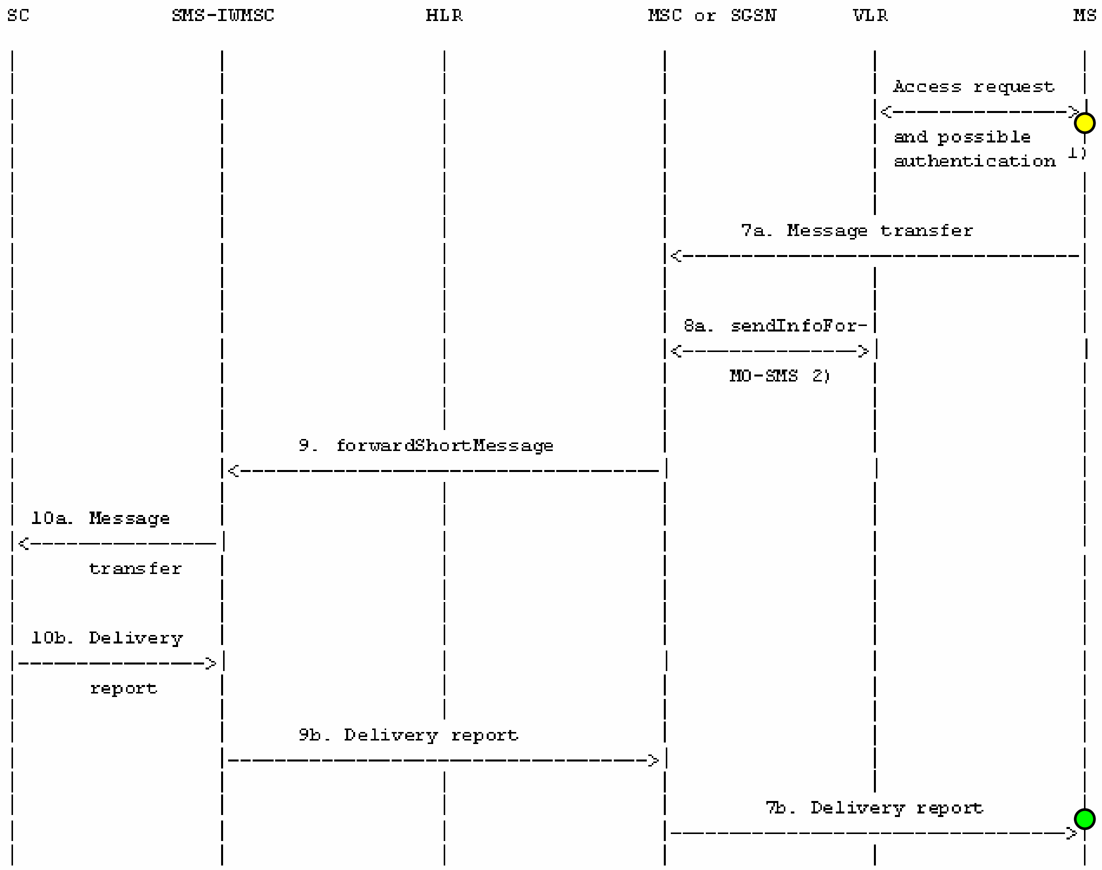
NOTE: For the trigger point explained here, the connection over the air interface must be tested (e.g. Layer-3).

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

$$\text{SMS Service Non-Accessibility MO}[\%] = \frac{\text{unsuccessful SMS service attempts}}{\text{all SMS service attempts}} \times 100\%$$

7.3.1.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
SMS service attempt	Start: Push send button (initiate sending a SMS).	Stop: 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	Stop: Receive the acknowledgement from the SMSC in the MO-party.	Stop: "Delivery Report" is received in the MS (MO) (green point in figure 6). Detailed: CP_DATA (RP_ACK) is received by MO.
Unsuccessful SMS service attempt	Stop trigger point not reached.	



--------> : Operation invocation or message transfer
<--------> : Successful operation invocation or message transfer including report

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

Figure 27: SMS Transaction flow - MO

7.3.2 SMS Access Delay MO [s]

7.3.2.1 Abstract Definition

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

7.3.2.2 Abstract Equation

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

$t_{receive}$: 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.

7.3.2.3 Trigger Points

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

7.3.3 SMS End-to-End Delivery Time [s]

7.3.3.1 Abstract Definition

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

7.3.3.2 Abstract Equation

$$\boxed{\text{SMS End-to-End Delivery Time [s]} = t_{receiveSMS} - t_{sendSMS}}$$

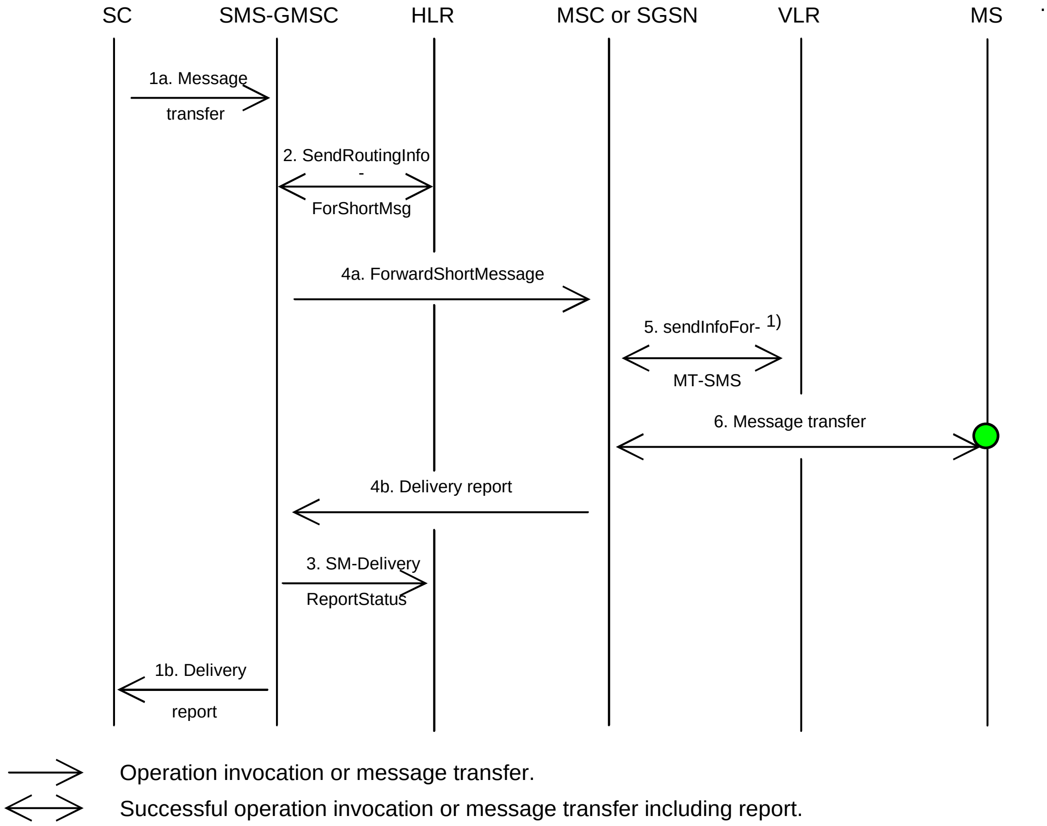
$t_{receive\ SMS}$: point of time the mobile equipment 2 receives the Short Message from mobile equipment 1.

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

7.3.3.3 Trigger Points

- Start SMS service attempt: Initiate sending a SMS.
- End SMS service attempt: Receiving SMS on Mobile Equipment 2.

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



NOTE: This operation is not used by the SGSN.

Figure 28: SMS Transaction flow - MT

7.3.4 SMS Completion Failure Ratio [%]

7.3.4.1 Abstract Definition

Ratio of not received and sent 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]).

7.3.4.2 Abstract Equation

$$\text{SMS Completion Failure Ratio [\%]} = \frac{\text{unsuccessfully received test SMS} - \text{duplicate received test SMS} - \text{corrupted test SMS}}{\text{all sent test SMS}} \times 100\%$$

7.3.4.3 Trigger Points

Event from abstract equation	Trigger point from customer's point of view	Technical description / protocol part
SMS service attempt	Start: Push send button (Initiate sending a SMS)	Start: The "Access request" is sent by the MS (MO). (Yellow point in figure 3). Detailed: CM Service Request is sent from MO.
Successfully received test SMS	Stop: The Short Message is received by MT-party's mobile	Stop: "Message Transfer" is received in the MS (MT). (Green point in figure 4). Detailed: CP_DATA (RP_ACK) is received by MT.
Unsuccessfully received test SMS	Stop trigger point not reached.	

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): Push to Talk over Cellular Information

Figures 29 to 32 visualise signal flows of typical PoC Sessions. The figures include the signal flows on the transport layer as well as some restricted information on the application layer. To keep the flows concise, some signals are not pictured. So it is possible to obtain signal flows universally valid for different kinds of PoC Sessions. Figures 29 to 32 show particularities using Unconfirmed Indication with Media Buffering as well as differences between Pre-established and On-demand PoC Sessions.

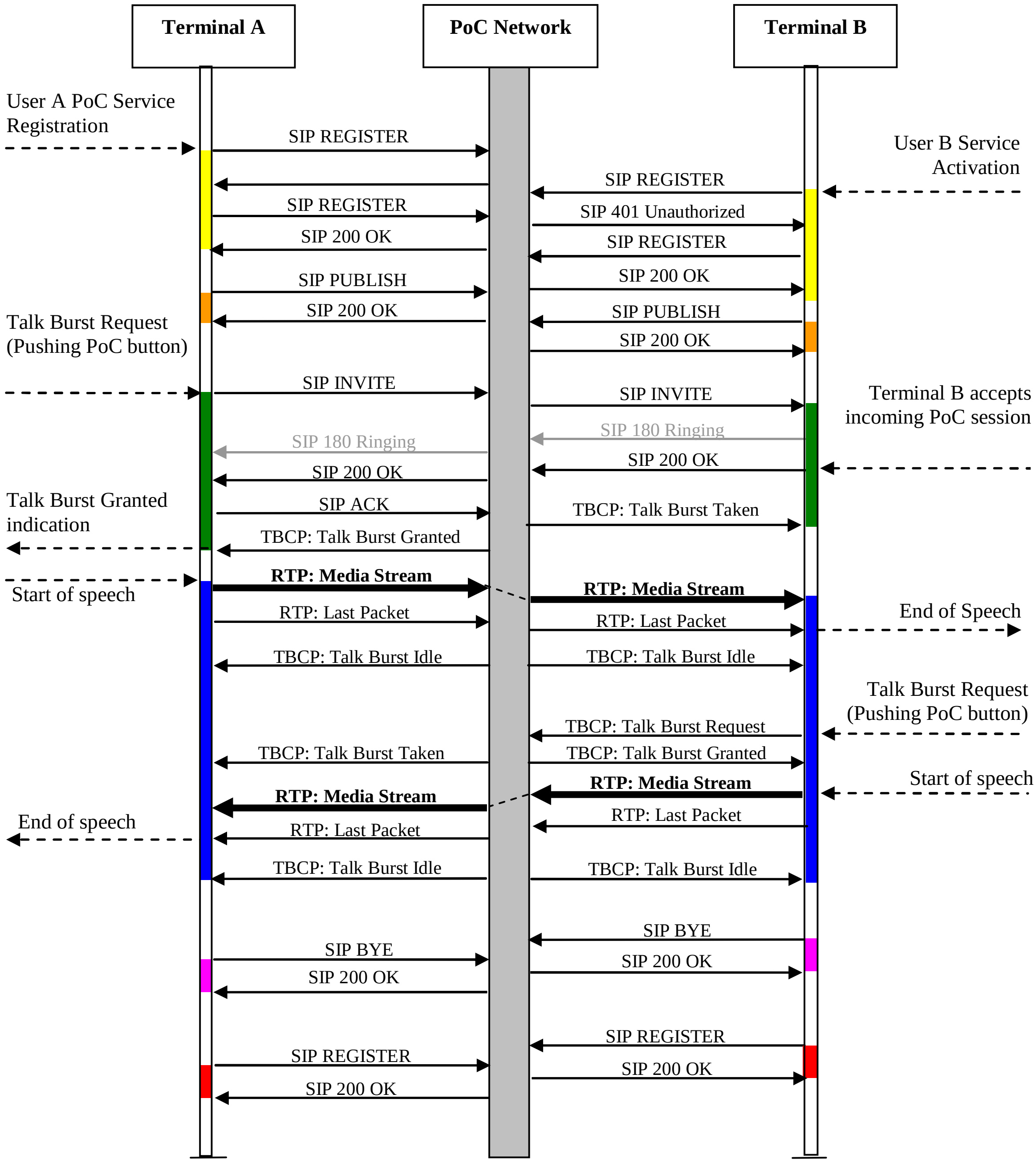
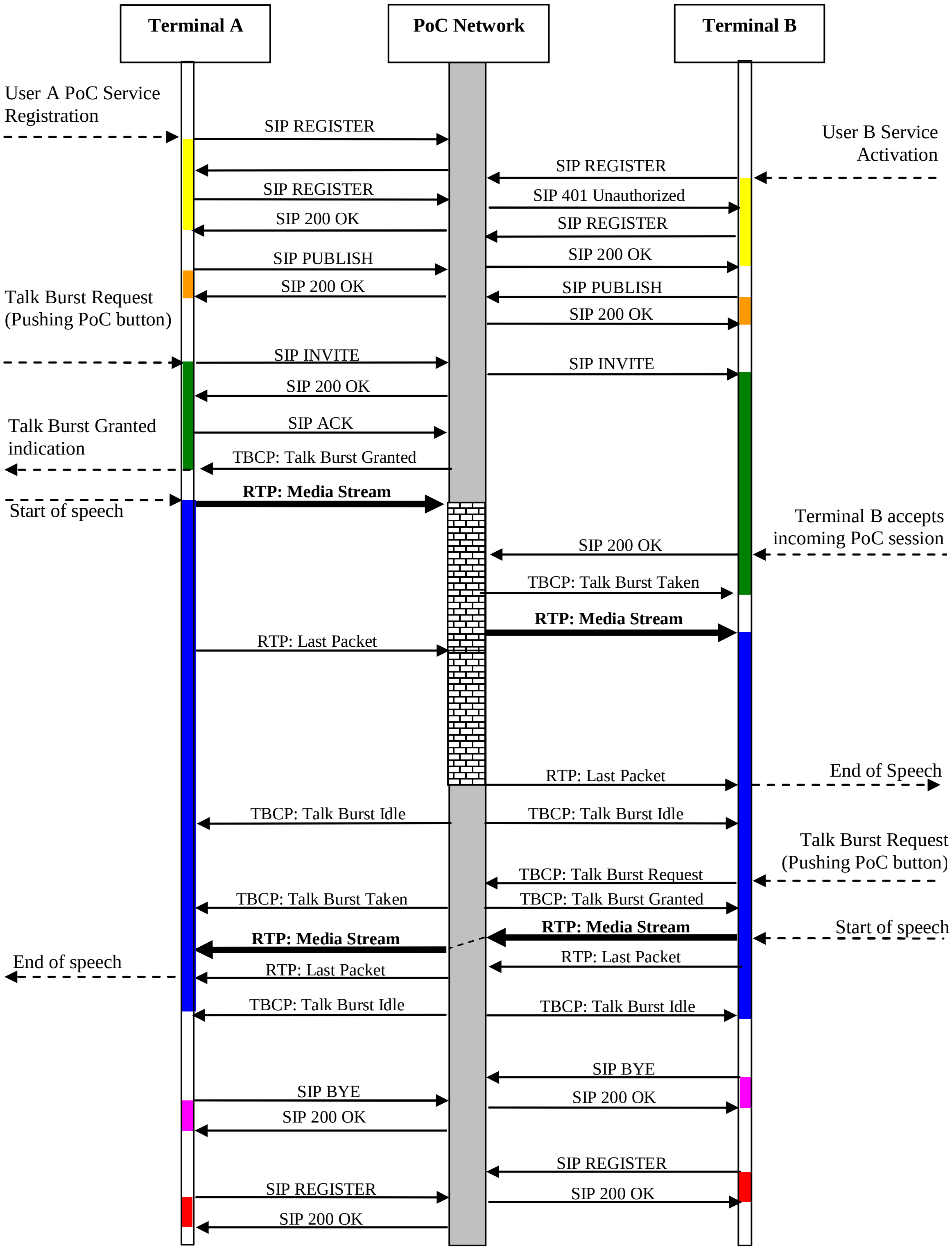


Figure 29: On-demand PoC Session with manual-answer



NOTE: The PoC Server supports Media Buffering and sends the Talk Burst confirm message after receiving the first automatic-answer message.

Figure 30: Unconfirmed On-demand Ad-hoc PoC Group Session with automatic-answer

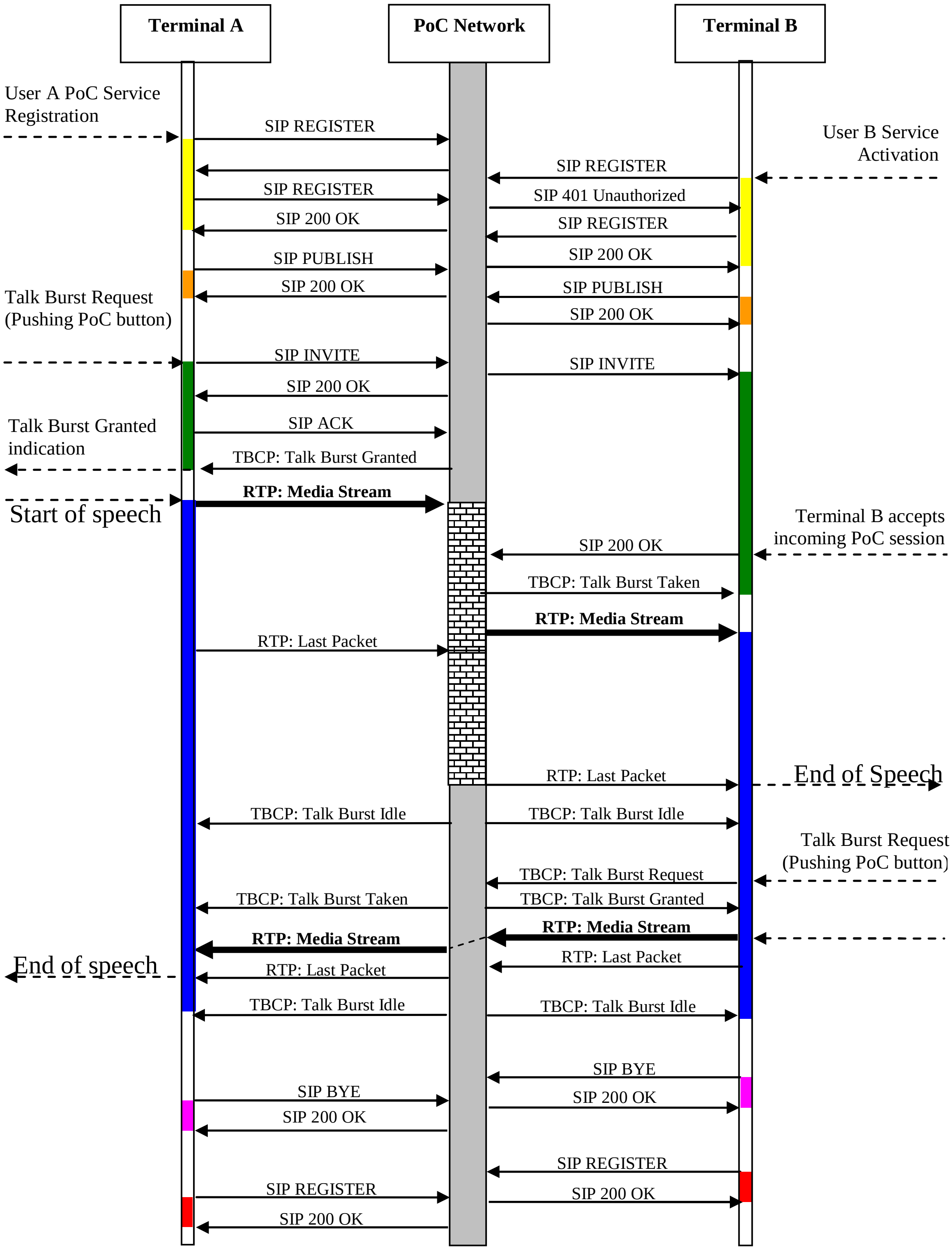


Figure 31: Confirmed Pre-established session with manual-answer

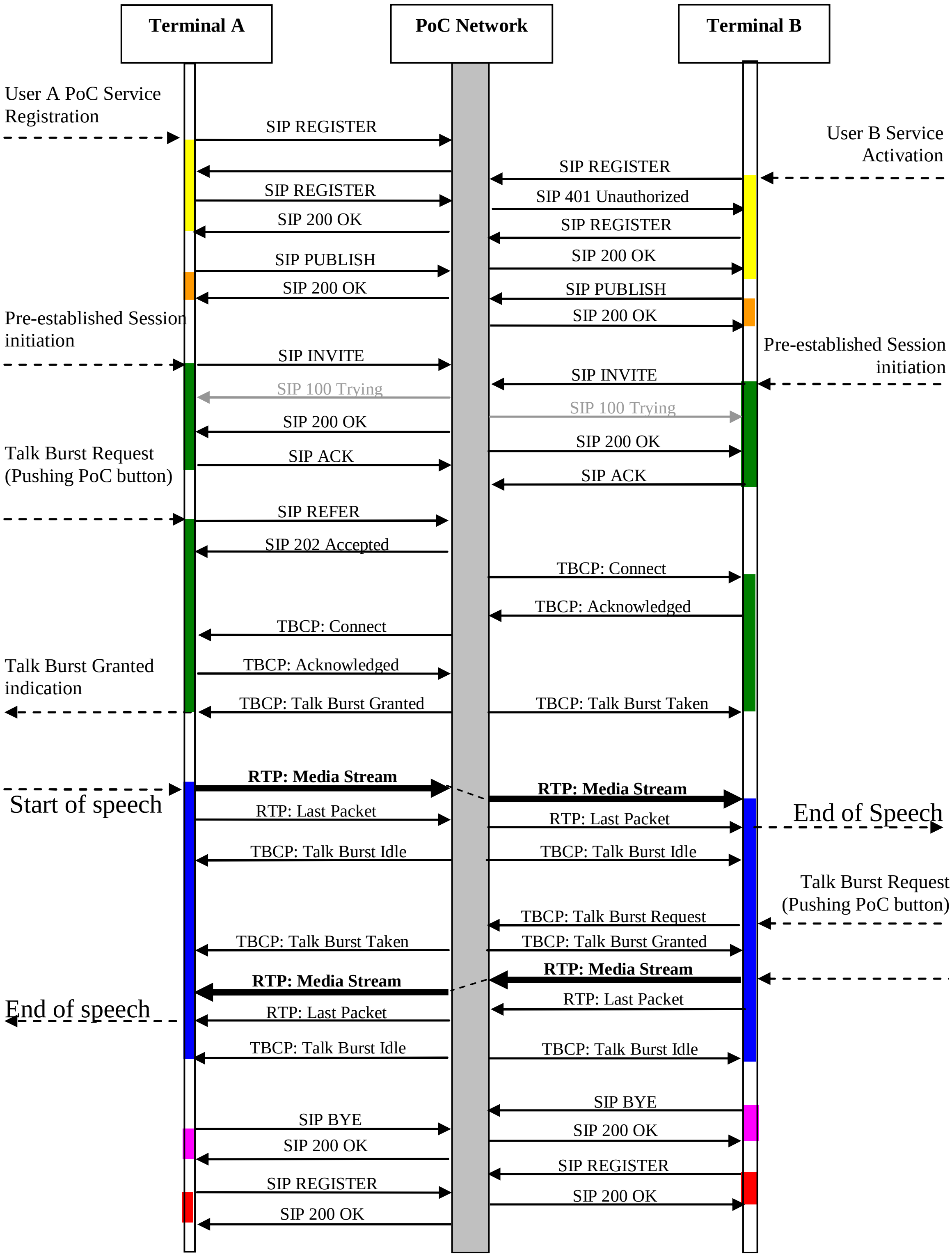


Figure 32: Unconfirmed Pre-established session with automatic-answer

C.1 Signal Grouping

This clause defines groups of signals which will in the following be referred to as building blocks of PoC signal flows, or just building blocks. These building blocks are derived from [13], [14], [15], representing only parts of a complete signal flow as seen in figures 29 to 32. Here, different building blocks of the same kind correspond to the same QoS group. The aim of the definition of such building blocks is to give detailed information on the different signal flows.

Remark(s): In the QoS parameter defining clause of the present document, most signal flows shown are less detailed. The reason for this is that these flows are only used to visualize the relevant trigger points of the corresponding QoS parameter with respect to their occurrence over time.

The relationship between building blocks and QoS groups is pictured in the following table. In contrast to the signal flows given to illustrate QoS parameter definition, only flows leading to a positive result are given. The only exception from this is the signal flow for a queued talk burst request which was added for sufficiency.

A distinction has been made between On-demand and Pre-established PoC Sessions since here different building blocks are needed. Crosses are indicating the blocks needed for the corresponding QoS group. For simplicity some crosses are greyed. These crosses indicate that a choice between Confirmed and Unconfirmed Indication has to be made.

Further parameters for the "Session SETUP" are the following:

- Session SETUP alternative 1: confirmed with auto-answered on terminating side.
- Session SETUP alternative 2: confirmed with manual answered on terminating side.
- Session SETUP alternative 3: unconfirmed with auto-answered on terminating side.
- Session SETUP alternative 4: unconfirmed with manual answered on terminating side.

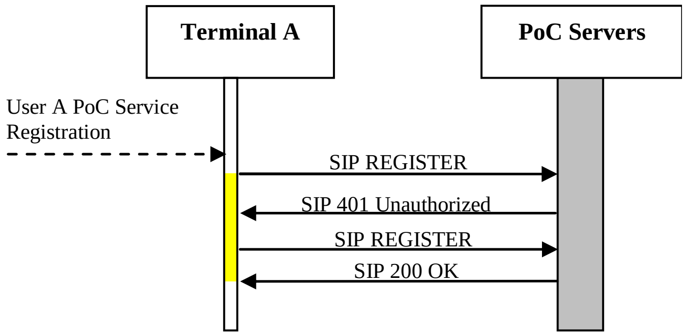
Remark(s):

- Only the QoS groups relevant to the building blocks are shown in table 2.
- Building blocks not related to any QoS group are omitted in table 2.
- Building blocks can be identified by their number as specified in table 2.

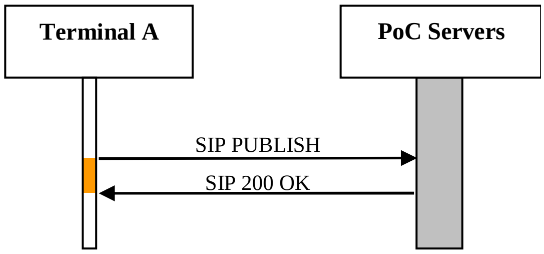
Table 2: Assignment of PoC Session parts to building blocks

Building blocks (below) & QoS groups (right hand side)		REG	PUB	REG long	On-demand							Pre-established								DeREG
					INIT	Session SETUP				PtS	LEAVE	NEGO	INIT	Session SETUP				PtS	LEAVE	
						1	2	3	4					1	2	3	4			
1	PoC Service Registration	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x
2	PoC Publish		x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	
3a	PoC On-demand Session Initiation, confirmed				x	x	x			x										
3b	PoC On-demand Session Initiation, unconfirmed				x			x	x	x										
3c	PoC Pre-established Session Media Parameters Negotiation											x	x	x	x	x	x	x		
3d	PoC Pre-established Session Initiation, confirmed												x	x	x			x		
3e	PoC Pre-established Session Initiation, unconfirmed												x			x	x	x		
4a	PoC On-demand Session Initiation, User B auto-answer					x		x		x										
4b	PoC On-demand Session Initiation, User B manual-answer						x		x	x										
4c	PoC Pre-established Session Initiation, User B auto-answer													x		x		x		
4d	PoC Pre-established Session Initiation, User B manual-answer														x		x	x		
5a	Media Stream from User A to PoC Server									x								x		
5b	Media Stream from PoC Server to User B, without Buffer									x								x		
5c	Media Stream from User B to User A, without Buffer									x								x		
6b	Talk Burst Request									x								x		
6c	Queued Talk Burst Request									x								x		
7a	Leaving PoC Session (On-demand)										x									
7b	Leaving PoC Session (Pre-established)																		x	
8	Deregistration																			x

C.2 PoC Service Registration

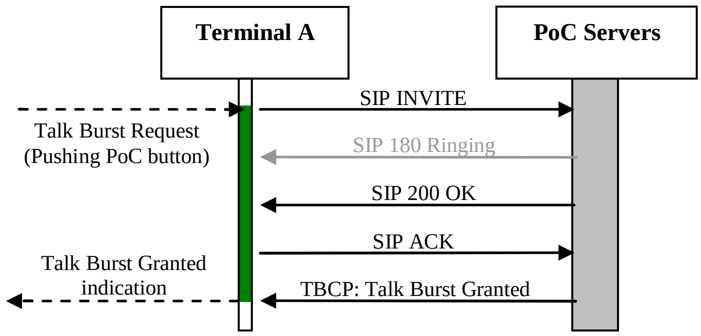


C.3 PoC Publish

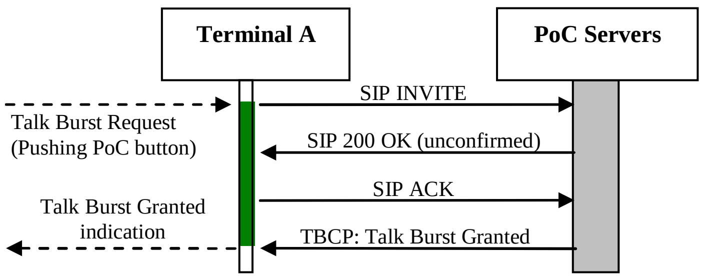


C.4 PoC Session Initiation, Originating Part

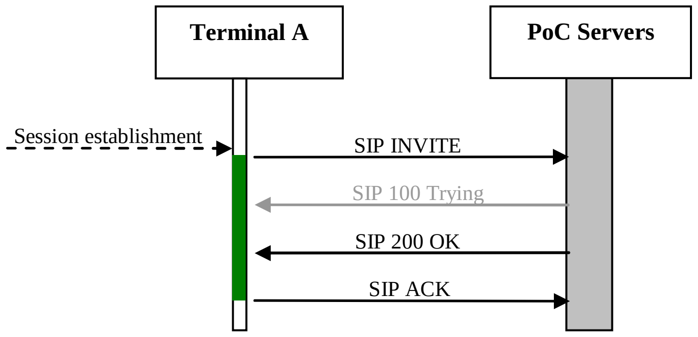
a) PoC On-demand Session Initiation, confirmed



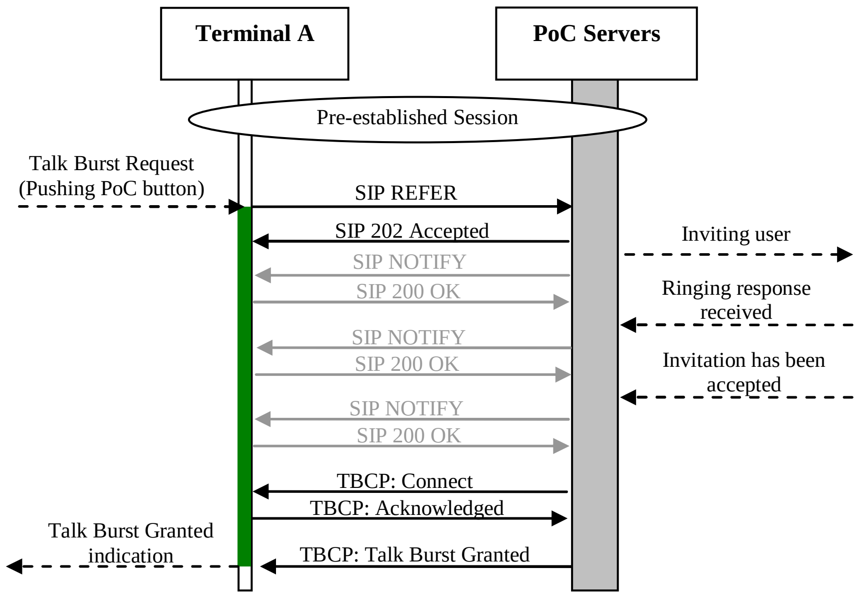
b) PoC On-demand Session Initiation, unconfirmed



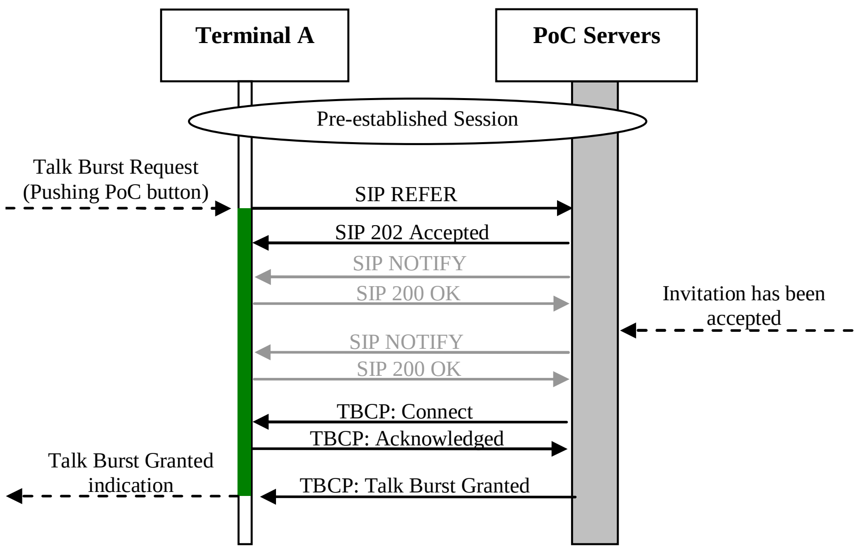
c) PoC Pre-established Session Media Parameters Negotiation



d) PoC Pre-established Session Initiation, confirmed

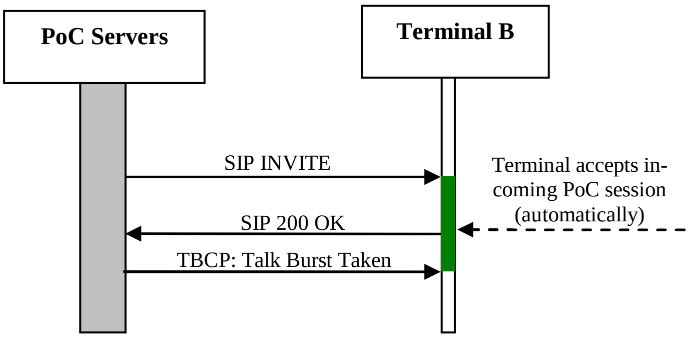


e) PoC pre-established Session Initiation, unconfirmed

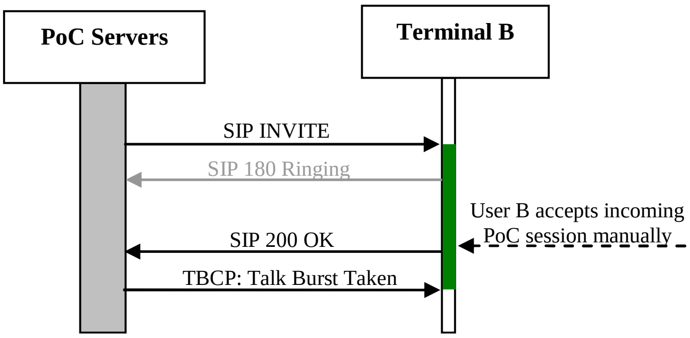


C.5 PoC Session Initiation, Terminating Part

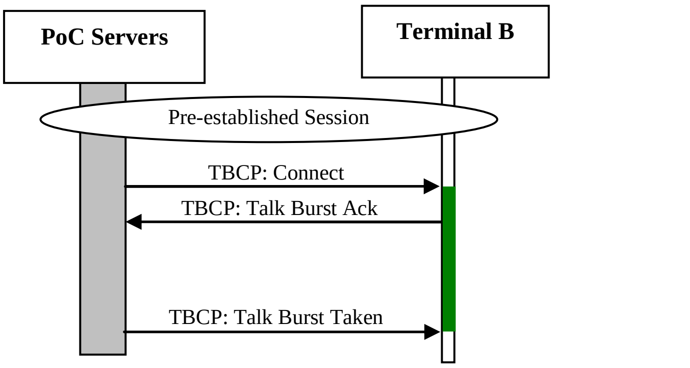
a) PoC On-demand Session, automatic answer



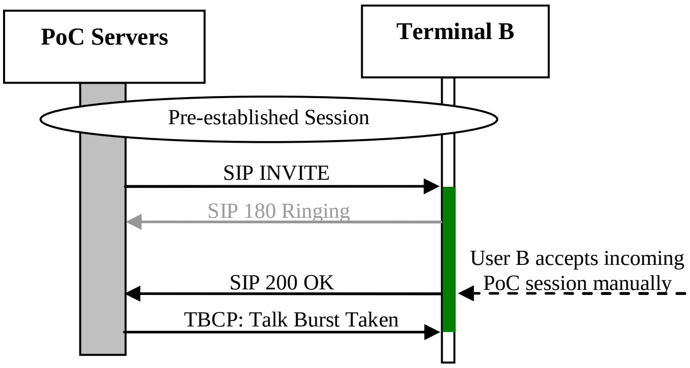
b) PoC On-demand Session, manual answer



c) PoC Pre-established Session, automatic answer

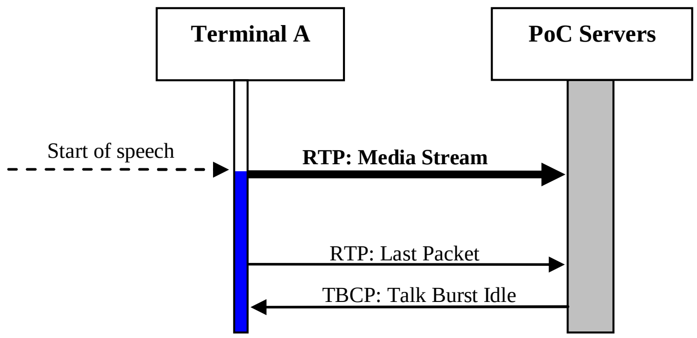


d) PoC Pre-established Session, manual answer

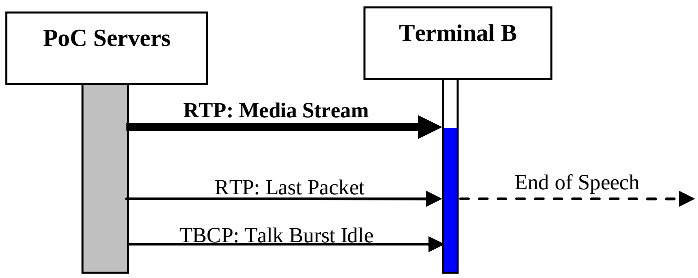


C.6 Media Streaming

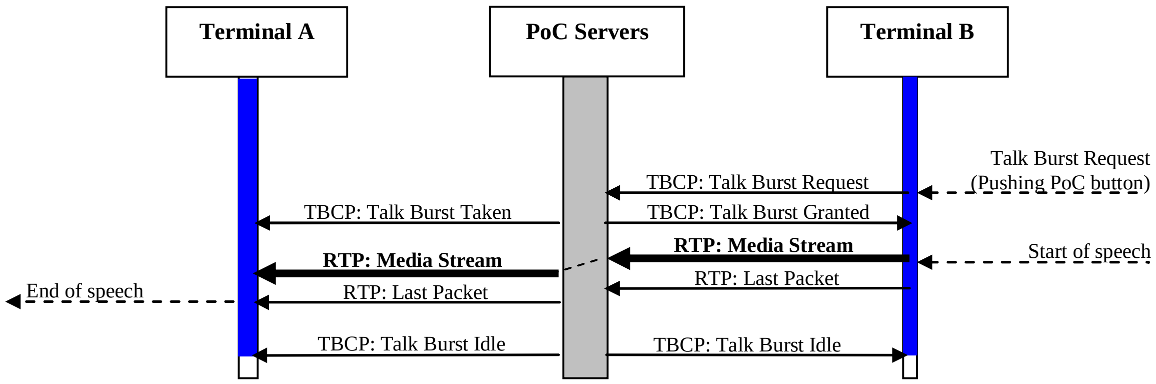
a) First Media Stream from User A to PoC Server



b) First Media Stream from PoC Server to User B (without Media Buffering)

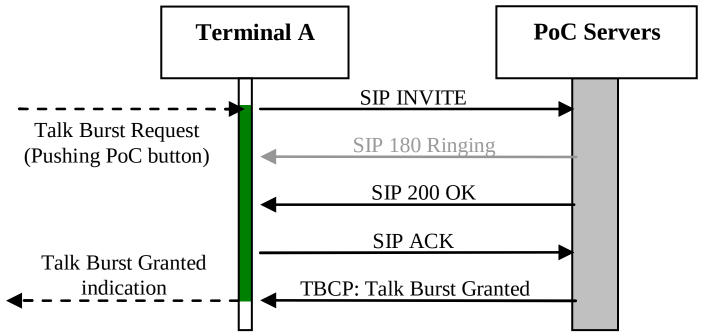


c) Last Media Stream from User B to User A via PoC Network (without Media Buffering), including Talk Burst Request of User B.

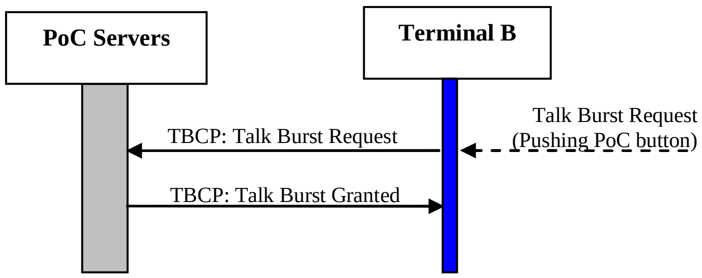


C.7 Talk Burst Request

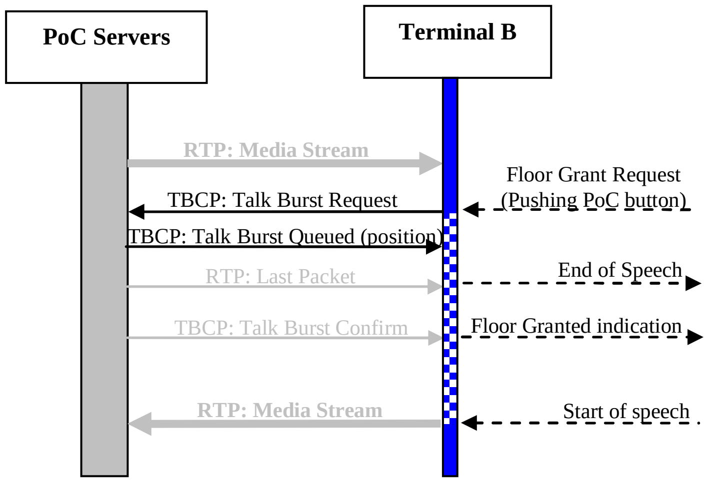
a) Implicit Talk Burst Request (On-demand Session Initiation)



b) Explicit Talk Burst Request

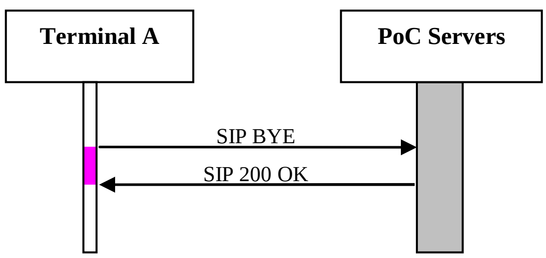


c) Queued Talk Burst Request

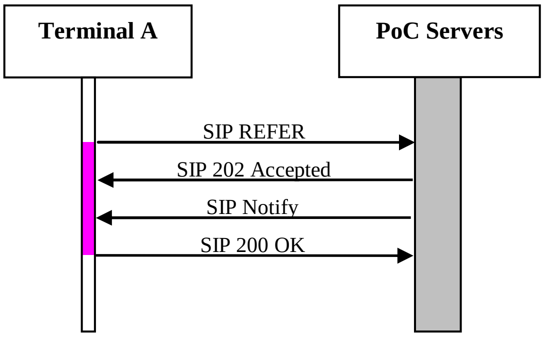


C.8 Leaving PoC Session

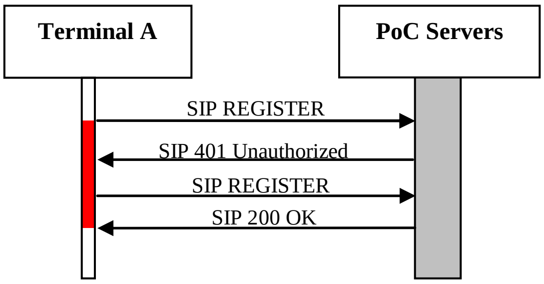
a) Leaving On-demand PoC Session



b) Leaving Pre-established PoC Session



C.9 Deregistration



Annex D (informative): Bibliography

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History

Document history		
V1.1.1	October 2003	Publication
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