Definition of Quality of Service parameters and their computation

Version 8.0

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This is a Non-binding Permanent Reference Document of the GSMA

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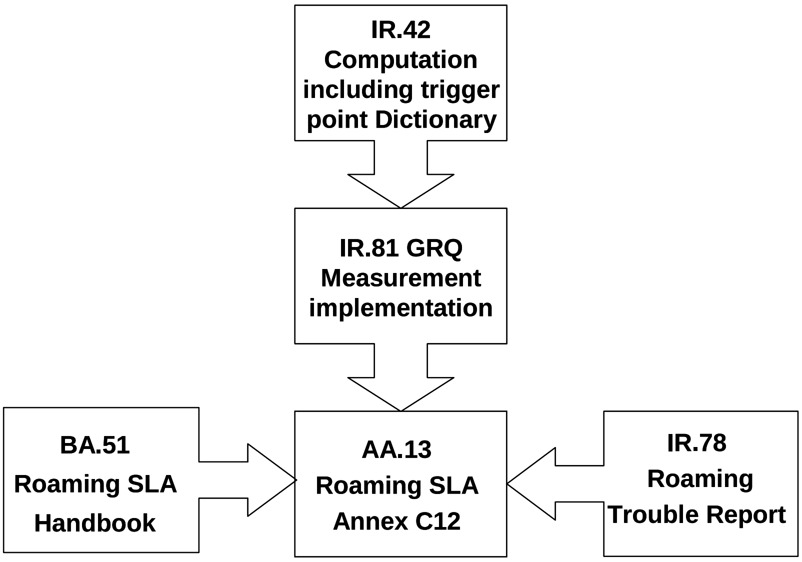
1 Introduction

Scope of document

This document defines quality of service (QoS) parameters and their computation. A QoS parameter is also called quality Key Performance Indicator (KPI). Although the QoS
1.1 definitions themselves are generic the scope of this document underlines the roaming deployment of those QoS parameters.

The parameter definition is split into two parts: the abstract definition and the generic description of the measurement method with the respective trigger points.

Consistent use of the definitions in this document will allow independent parties to compare QoS measurements and results. Figure 1 shows the relationship to the other QoS-related PRDs.



1.2 **Figure 1: Relationship between GSMA QoS documents**

General considerations

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

It is assumed that the mobile and the desired service can be operated correctly by the end user, as operability is not evaluated in this document. 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.

Only voice quality values measured for calls ended successfully must be used for statistical analysis. However, measured values from calls ended unsuccessfully (for example, calls that are dropped) should be available for additional evaluation if required, and therefore must be stored. Further preconditions will apply when reasonable.

Monitoring of services using the parameters defined in this document could impact the traffic and load on the networks involved, including the home public mobile network (HPMN), the visited public mobile network (VPMN), and intermediary networks. Due consideration must therefore be given to the monitoring regime to avoid unnecessary or adverse impacts on these networks.

1.3 Definition of Terms

Term	Meaning
APN	Access Point Name
CS	Circuit Switched
CSFB	Circuit Switched Fall Back
DNS	Domain Name System
EPC	Evolved Packet Core
FTP	File Transfer Protocol
GBR	Guaranteed Bit Rate
HLR	Home Location Register
HPMN	Home Public Mobile Network
HSS	Home Subscriber Server
HTTP	Hypertext Transport Protocol
IMS	Internet Protocol Multimedia Subsystem
IPX	Internet Protocol Exchange
ISUP	ISDN User Part
KPI	Key Performance Indicator
LQO	Listening Quality Objective
LTE	Long Term Evolution (Radio)
MME	Mobility Management Entity
MT	Mobile Terminated
NER	Network Efficiency Ratio
MS	Mobile Station
MSC	Mobile Switching Centre
OCN	Original Called Number
PCEF	Policy & Charging Enforcement Function
PCRF	Policy & Charging Rule Function
P-CSCF	Proxy - Call Session Control Function
PESQ	Perceptual Evaluation of Speech Quality
PEVQ	Perceptual Evaluation of Video Quality
PING	Packet Internet Groper

Term	Meaning
POLQA	Perceptual Objective Listening Quality Analysis
PRD	Permanent Reference Document
QCI	QoS Class Indicator
QoS	Quality of Service
RDN	Redirecting Number
RTCP	RTP Control Protocol
RTP	Real-time Transport Protocol
SGSN	Serving GPRS Support Node
SIP	Session Initiation Protocol
SLA	Service Level Agreement
SMS MO	Mobile Originated SMS
SMS MT	Mobile Terminated SMS
SMSoSGs	SMS over SGs
SS7	Signalling System 7
SQI	Service Quality Indicator
SRVCC	Single Radio Voice Call Continuity
SSI	Single Service Indicator
ViLTE	(conversational) Video over LTE
VoLTE	Voice over Long Term Evolution
VPMN	Visited Public Mobile Network

1.4 Document Cross-References

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

- References are non-specific, i.e. refer to the latest version of the document.

No.	Document	Description
[1]	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".
[2]	ETSI TS 102 250-2	"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 ".
[3]	GSMA PRD IR.88	LTE and EPC Roaming Guidelines
[4]	ITU-T-E.437	“Comparative metrics for network performance Management”
[5]	ETSI EN 300 089	"Integrated Services Digital Network (ISDN); Calling Line Identification Presentation (CLIP) supplementary service; Service description".
[6]	ITU-T-Y.1540	“IP packet transfer and availability performance parameters”; IPLR – IP Packet Loss Ratio

No.	Document	Description
[7]	3GPP TS 29.002	”Mobile Application Part (MAP) specification”
[8]	3GPP TS 29.060	”General Packet Radio Service (GPRS); GPRS Tunnelling Protocol (GTP) across the Gn and Gp interface”
[9]	3GPP TS 24 008	"Mobile radio interface Layer 3 specification; Core network protocols; Stage 3".
[10]	IETF RFC 2647	”Benchmarking Terminology for Firewall Performance; 3.1.7 Goodput”
[11]	GSMA PRD IR.34	GSMA Inter-Service Provider IP Backbone Guidelines
[12]	ITU-T-E.800	“Terms and Definitions Related to Quality of Service and Network Performance Including Dependability - Telephone Network and ISDN Quality of Service, Network Management and Traffic Engineering (Study Group II)”
[13]	ITU-T E.431	“Service Quality Assessment for Connection Set-up and Release Delays (Study Group II)”
[14]	ITU-T E.425	“Internal Automatic Observations Series E: Overall Network Operation, Telephone Service, Service Operation and Human Factors Network Management - Checking the Quality of the International Phone Service”
[15]	GSMA PRD BA.51	“Roaming Service Level Agreement Guidelines”
[16]	ITU-T QE.850	“Usage of cause and location in the digital subscriber signalling system No. 1 and the signalling system No. 7 ISDN user part”
[17]	ITU-T Q.732.2	“Stage 3 Description for Call Offering Supplementary Services Using Signalling System No.7: Call Diversion Services: - Call Forwarding Busy - Call Forwarding No Reply - Call Forwarding Unconditional - Call Deflection - Series Q: Switching and Signalling - Specifications of Signalling System No.7 - ISDN Supplementary Services”
[18]	GSMA PRD IR.25	VoLTE Roaming Testing
[19]	GSMA PRD IN.21	“GSM Association Roaming Database, Structure and Updating Procedures”
[20]	3GPP TS 29.272	MME and SGSN related interfaces based on Diameter protocol
[21]	ITU-T P.862	Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs
[22]	GSMA PRD AA.13	“International Roaming Agreements – Common Annexes
[23]	GSMA PRD IR.78	“Roaming Trouble Report”
[24]	3GPP TS 45.008	Radio subsystem link control
[25]	3GPP TS 25.304	UE procedures in idle mode and procedures for cell reselection in connected mode
[26]	3GPP TS 36.304	EUTRA; UE procedures in idle mode
[27]	ITU-T P.863	Perceptual Objective Listening Quality Assessment (POLQA)
[28]	ETSI TR 102 506	Estimating Speech Quality per Call
[29]	ITU-T P.862.1	Perceptual evaluation of speech quality (PESQ): Mapping function for transforming P.862 raw result scores to MOS-LQO
[30]	GSMA PRD IR.65	IMS Roaming & Interworking Guidelines

No.	Document	Description
[31]	GSMA PRD IR.92	IMS Profile for Voice and SMS
[32]	GSMA PRD IR.94	IMS Profile for Conversational Video Service
[33]	GSMA PRD NG.103	VoLTE-RCS Roaming and Interconnection Guidelines
[34]	ETSI TR 103 219	QoS Aspect of Voice Communication in an LTE Environment
[35]	ITU-T G.107.1	Wideband E-Model
[36]	3GPP TS 29.109	Generic Authentication Architecture (GAA); Zh and Zn Interfaces based on Diameter Protocol
[37]	3GPP TS 29.213	Policy and Charging Control Signalling Flows and Quality of Service (QoS) Parameter Mapping
[38]	3GPP TS 29.214	Policy and Charging Control over Rx Reference Point
[39]	3GPP TS 29.229	Cx and Dx interfaces based on the Diameter protocol; Protocol Details
[40]	3GPP TS 29.329	Sh interfaces based on the Diameter protocol; Protocol Details
[41]	ITU-T J.247	Objective perceptual multimedia video quality measurement in the presence of a full reference
[42]	IETF RFC 3550	RTP: A Transport Protocol for Real-Time Applications
[43]	IETF RFC 4961	Symmetric RTP / RTP Control Protocol (RTCP)
[44]	ITU-T Y.1540	Internet protocol data communication service – IP packet transfer and availability performance parameters
[45]	IETF RFC 5481	Packet Delay Variation Applicability Statement
[46]	3GPP TS 24.301	NAS Protocol for Evolved Packet System

2 QoS Parameters

2.1 QoS Parameters Model

Figure 2 shows a model for QoS parameters. This model has four layers.

- The first layer is the Network Availability, which defines QoS from the viewpoint of the network and service provider.
- The second layer is the Network Access, the basic requirement for all the other QoS aspects and QoS parameters. The outcome of this layer is the QoS parameter Network Accessibility from the viewpoint of the service user.
- The third layer contains the three QoS aspects: Service Access, Service Integrity and Service Retain-ability.
- The fourth layer contains the different services to be provided to the service user in case of roaming, for example Voice (Telephony), SMS, Web browsing, File Transferring and Ping services. The outcomes are the QoS parameters for those services.

For monitoring of roaming QoS, please refer to PRD IR.81 for details of call flows and trigger points.

Note that the QoS is dependent on interconnectivity between operators and carriers or between roaming hub providers. An Internet Protocol Exchange (IPX) QoS monitoring scheme provides more possibilities for measuring QoS across IPX interconnections [11].

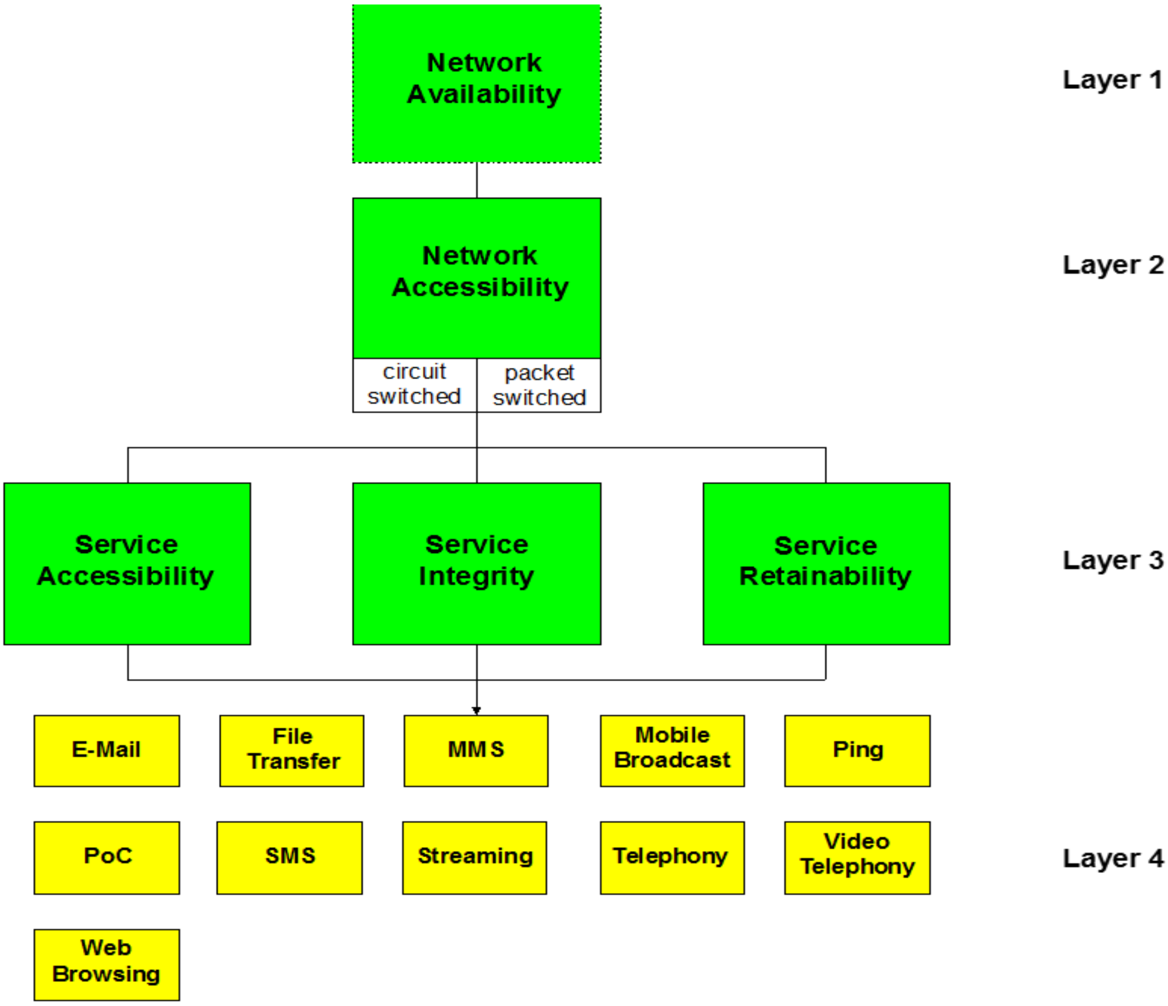


Figure 2: Four Layer model for QoS Parameters.

2.2 Service-independent QoS parameters

The service-independent QoS parameters characterise network availability, network accessibility, IP connectivity and bearer context cut-off. The parameters are therefore considered as technology-dependent. Table 1 provides an overview.

Technologies				
	2G / 3G	**4G**	VoLTE/ViLTE	**Future technology**
Network /service availability	Scan for PLMN: Radio network availability	Scan for PLMN: Radio network availability	EPS Network feature support: IMS voice / video over PS session in S1 mode supported	FFS
Network / service	CS and/or PS attach: Network selection and registration successful ratio /time	LTE attach (EPS or EPS+IMSI): Network selection and registration successful ratio /time	IMS Registration success ratio /time	
IP connectivity	(Primary) PDP context activation: PDP context activation successful ratio /time	Default / Dedicated EPS bearer context activation: Default EPS bearer context activation successful ratio /time	QCI 5 default EPS bearer context activation for the IMS well-known APN connection; QCI 1 dedicated EPS bearer context activation for voice media; Default / dedicated EPS bearer context activation success ratio /time	
Network / service retainability	PDP context dropped: PDP context cut-off ratio	Default EPS bearer context dropped: Default EPS bearer context cut-off ratio	QCI 5 default / QCI 1 dedicated EPS bearer context cut-off ratio	

Table 1: Overview of QoS parameters on technology-dependency

For 2G/ 3G (i.e. GSM/GPRS and UMTS) networks it is necessary to establish a data connection before the possibility of accessing a service. In LTE networks the concept of IP connectivity “always on” has been established. It allows a faster access to the services of a mobile network by state changes on request of a user or application. The impact of this concept on the QoS parameters is that they cannot be triggered explicitly anymore as they require a specific service request. Nevertheless the defined QoS parameters remain valid and can be derived from the trigger points given.

This section contains also DNS KPI. DNS service is an intermediate internet service used by the other specific PS data services such as FTP or HTTP. However, the DNS service is independent from those specific data services. Network Accessibility Circuit Switched (NA – CS) [1] [12].

Please refer to PRD IR.81 for details of call flow and trigger points.

2.2.1 Network Availability [2] [12]

2.2.1.1 Abstract definition

Probability that the Mobile Services are offered to a user.

See ITU-T Rec. E.800: The probability that the user of a service after a request receives the proceed-to-select signal within specified conditions.

See complementary QoS parameter RNU in ETSI TS 102 250-2 (5.1 Radio Network Unavailability).

2.2.1.2 Computation

2.2.1.2.1 Trigger points:

GSM: C1-Criteria > 0 [24]

GPRS: broadcasting GPRS indicator in system information 3 or system information 4

UMTS (WCDMA) [25], LTE [26]: S Criteria fulfilled

Any emergency camping on any other than the target networks is considered as no network.

The target networks could constitute more than one network, for example to cover national or international roaming or via a roaming hub.

When the mobile equipment supports multi-mode (GSM/UMTS/LTE),the judgement on Radio Network Availability is made with respect to the radio access technology under the test aspect.

For CSFB and SMSoSGs, LTE S criteria is applied.

2.2.1.2.2 Abstract formula:

$$\text{Radio Network Availability}\,[\%] = \frac{\text{probing attempts with mobile services available}}{\text{all probing attempts}} \times 100$$

2.2.2 Network Accessibility [2]

2.2.2.1 Abstract definition

Probability that the user performs a successful selection and registration on the desired PLMN (manual selection mode, automatic selection mode with a defined desired PLMN) or on a certain PLMN (automatic selection mode without a defined desired PLMN).

See ETSI 102 250-2 (5.2.1 Network Selection and Registration Failure Ratio).

2.2.2.2 Computation

2.2.2.3 Trigger points:

Initiate manually or automatically PLMN selection, stop measurement after successful registration.

Abstract formula:

2.2.2.3

$$\text{NA[\%]} = \{\text{Manual} \mid \text{Automatic}\}\ \text{Network Selection and Registration Success Ratio [\%]} = \frac{\text{successful selection and registration attemps on PLMN}}{\text{all selection and registration attempts}} \times 100$$

NA-CS[%] is applied to GSM or UMTS-CS. NA-PS[%] is applied to GPRS, UMTS-PS, LTE, respectively.

2.2.3 Circuit Switched LU Success Ratio (CS LU – SR) [2]

2.2.3.1 Abstract definition

The CSLU success ratio describes the probability that a subscriber can successfully attach to the CS network.

See ETSI 102 250-1 (5.2 Network non-accessibility).

2.2.3.2 Computation

See ETSI TS 102 250-2 (5.2 Network non-accessibility).

2.2.3.2.1 **Abstract formula:**

$$CSLUSuccessRatio = \frac{NrSuccessfulCSAttachAttempts}{TotalNrCSAttachAttempts}\ x100\,\%$$

2.2.3.2.2

Trigger points:

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

2.2.3.2.3

Remarks

Success ratio measurements will depend on whether the LU is the very first LU attempt or one of subsequent attempts. (See remarks for CSLU-D below).

2.2.4 Circuit Switched Location Update Delay (CS LU - D)

2.2.4.1 Abstract definition

This CSLU delay describes the time period needed to attach to the CS network.

2.2.4.2 Computation

2.2.4.2.1 **Abstract formula:**

$$CSLUDelay = \frac{Sum(t_{CSAttachComplete} - t_{CSAttachStart})}{NrSuccessfulCSAttachAttempts}$$

2.2.4.2.2 **Trigger points:**

(for the computation of the unit CSLU delay):

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

2.2.4.2.3 **Remarks:**

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 Mobile Switching Centre (MSC) has changed since the detach, or if it is the very first attach of the mobile to the network), the MSC contacts the Home Location Register (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.

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

2.2.5 Packet Switched LU Success Ratio (PS LU - SR)[1] [2] [7]

2.2.5.1 Abstract definition

The PSLU success ratio describes the probability of a subscriber to successfully attach to the PS network.

See ETSI TS 102 250-2 (5.3 Attach Failure Ratio)

[1] The KPI Packet-Switched Location Update was named originally from the MAP_Update_GPRS_Location procedure, as described in 3GPP TS 29.002 [7]. It is used for the PS attach, i.e. for GPRS attach. The same term Location is also used in the Location Management procedure in 3GPP 29.272 [20] for Update Location and Cancel Location at S6a, i.e. EPS attach.

2.2.5.2 Computation 4

2.2.5.2.1 **Abstract formula:**

$$\boxed{PSLUSuccessRatio = \frac{NrSuccessfulPSAttachAttempts}{TotalNrPSAttachAttempts} x100\,\%}$$

2.2.5.2.2 **Trigger points:**

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

2.2.5.2.3 **Remarks**

Depending upon the technologies, PS attach is understood as either a GPRS attach, or an EPS attach.

A combined EPS + IMSI attach is applied to CSFB or SMSoSGs.

2.2.6 Packet Switched Location Update Delay (PS LU - D) [2]

2.2.6.1 Abstract definition

This PSLU delay describes the time period needed to attach to the PS network.

See ETSI TS 102 250-2 (5.4 Attach Setup Time)

2.2.6.2 Computation

2.2.6.2.1 **Abstract formula:**

2.2.6.2.2

$$\boxed{PSLUDelay = \frac{Sum(t_{PSAttachComplete} - t_{PSAttachStart})}{NrSuccessfulPSAttachAttempts}}$$

Trigger points:

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

2.2.6.2.3 **Remarks:**

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, S4 SGSN or MME has been changed since the detach, or if it is the very first attach of the mobile to the network), the SGSN contacts the HLR, the S4 SGSN contacts HSS or the MME contacts HSS respectively, 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.

2.2.7 PDP Context Activation Success Ratio (SA PSD) [2] [19]

2.2.7.1 Abstract definition

A packet-switch data session will be considered set-up successfully if a PDP Context can be successfully activated.

See ETSI TS 102 250-2 (5.5 PDP Context Activation Failure Ratio)

2.2.7.2 Computation

2.2.7.2.1 Abstract formula:

$$\boxed{PDPContextActivationSuccessRate = 100\,\% \times \frac{NrOfPDPContextActivationsSuccessful}{NrOfAllPDPContextActivationAttempts}}$$

Any PDP Context Activation request is considered as an attempt.

2.2.7.2.2 Trigger points:

PDP Context Activations are considered successful upon the reception of notifications of successful PDP context activation (Activate PDP Context Accept).

2.2.7.2.3 Remarks:

It is recommended to use a limited and defined list of APNs for consistent measurements. Operators are encouraged to maintain a list of standard APNs for measurements in PRD IR.21 (or alternatively defined in roaming service level agreements (SLAs)). The list could include APNs for MMS, WAP and/or internet, both pre-paid and post-paid. The APNs should be defined as part of test conditions.

LTE: see Default EPS Bearer Context Activation Success Ratio

2.2.8 PDP Context Activation Time [2] [19]

2.2.8.1 Abstract definition

Is time between sending the PDP Context Activation request and receiving the notification of successful completion of that activation.

See ETSI TS 102 250-2 (5.6 PDP Context Activation Time)

2.2.8.2 Computation

2.2.8.2.1 Abstract formula:

$$\boxed{\mathrm{PDPContextActivationTime[s]} = \left(t_{\mathrm{PDPcontextactivationaccept}} - t_{\mathrm{PDPcontextactivationrequest}}\right)[s]}$$

Trigger points:

Start:	Sending of the PDP Context Activation request
2.2.8.2.2 End:	Reception of the notification of successful PDP context activation (Activate PDP Context Accept)

Remarks: 5
When averaging the PDP Context Activation Time, only the successful activations should be considered.

It is recommended to use a limited and defined list of APNs for consistent measurements. Operators are encouraged to maintain a list of standard APNs for measurements in PRD IR.21 (or alternatively defined in Roaming SLAs). The list could include APNs for MMS, WAP and/or internet, both pre-paid and post-paid. The APNs should be defined as part of test conditions.

User activation (GPRS PDP context creation procedure) is recommended as network activation tends to be for local services (home usage).

LTE: See Default EPS Bearer Context Activation Time or additional Default EPS Bearer Context Activation Time

2.2.9 PDP Context Cut-Off Ratio [2] [8]

2.2.9.1 Abstract definition

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

ETSI TS 102 250-2 (5.7 PDP Context Cut-off Ratio).

See also 3GPP TS 29.060.

2.2.9.2 Computation

2.2.9.2.1

Abstract formula:

2.2.9.2.2

$$PDPContextCutOffRatio = 100\,\% \times \frac{NrOfPDPContextLossesNotInitiatedByTheUser}{NrOfAllSuccessfullyActivatedPDPContexts}$$

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
2.2.9.2.3 the SGSN or GGSN. The UE receives from the network the message DEACTIVATE PDP CONTEXT REQUEST, MODIFY PDP CONTEXT REQUEST or DETACH REQUEST.

Remarks:

When analysing how to practically measure this parameter, a key point will be to identify how to assess whether a PDP context loss has been initiated by the user or not. Active and passive monitoring methods might adopt different approaches for that. In the active monitoring case, the test case specifies when the PDP context should be deactivated and any loss prior to that can thus (generally) be considered as ‘not initiated by the user’. In the passive monitoring case, the use of PDP context failure codes can be used.

A precondition for measuring this parameter is that a PDP context is successfully established first.

2.2.10 Default EPS Bearer Context Activation Success Ratio [2] [3]

2.2.10.1 Abstract definition

Default EPS bearer context for the default APN: success of the EPS attach procedure (PS LU-SR)
In case of an additional default EPS bearer context: success of activation of the additional default EPS bearer context

See ETSI TS 102 250-2 (5.12.1 Default EPS Bearer Context Activation Failure Ratio)

2.2.10.2 Computation

2.2.10.2.1 Abstract formula:

Default EPS bearer context activation success ratio for the default APN: same as EPS attach success ratio (PS LU-SR).

Additional default EPS bearer context, referring to an APN different from the default one:

$$\text{Additional Default EPS Bearer Context Activation Success Ratio [\%]} = \frac{\text{additional PDN connection establishment successes}}{\text{additional PDN connection initiations}} \times 100$$

Any additional Default Context Activation request is considered as an initiation.

2.2.10.2.2 Trigger points:

The trigger points for the default EPS bearer context referring to the default APN is same as EPS attach procedure.

PDN Connectivity Request for additional default EPS bearer Context Activations is sent by the UE. The additional PDN connectivity is considered successful upon the reception of notifications of additional default EPS bearer context activation (Activate default EPS bearer Context Request)

2.2.10.2.3 Remarks:

It is recommended to use a limited and defined list of APNs for consistent measurements. Operators are encouraged to maintain a list of standard APNs for measurements in PRD IR.21 (or alternatively defined in roaming service level agreements (SLAs)). The list could include APNs for MMS, WAP and/or internet, both pre-paid and post-paid. The APNs should be defined as part of test conditions.

When the UE is powered on it will perform an EPS attach which includes the registration with the network and the setup of a default EPS bearer context for the default APN. A default EPS bearer context is successfully established when (EPS) ATTACH COMPLETE is sent by the UE. This context replaces the primary PDP context defined for 2G/3G networks.

2.2.11 Default EPS Bearer Context Activation Time [2]

2.2.11.1 Abstract definition

The default EPS bearer context activation time is the time period needed to establish the initial default EPS bearer context for the default APN or any additional PDN connection (i.e. any additional default EPS bearer context), respectively.

See ETSI TS 102 250-2 (5.12.2 Default EPS Bearer Context Activation Time)

2.2.11.2 Computation

2.2.11.2.1 Abstract formula:

Default EPS bearer context for the default APN[2]:

$$\boxed{\text{Default EPS Bearer Context Activation Time } [s] = \left(t_{attachcomplete} - t_{attachrequest}\right)[s]}$$

Additional default EPS bearer context:

$$\boxed{\begin{array}{l}\text{Additional Default EPS Bearer Context Activation Time } [s] = \\ \left(t_{Additional\,PDN connection establishment} - t_{Additional\,PDN connection initiation}\right)[s]\end{array}}$$

2.2.11.2.2 Trigger points:

PDN CONNECTIVITY REQUEST for additional default EPS bearer Context Activations is sent by the UE. The ACTIVATE DEFAULT EPS BEARER CONTEXT ACCEPT is sent by the UE.

2.2.11.2.3 Remarks:

Precondition: Successful default PDN connection for the default APN is prior to the additional default PDN connections.

[2] The PDN CONNECTIVITY REQUEST message is piggybacked in ATTACH REQUEST. The ACTIVATE DEFAULT EPS BEARER CONTEXT REQUEST message is piggybacked in ATTACH ACCEPT. The ACTIVATE DEFAULT EPS BEARER CONTEXT ACCEPT message is piggybacked in ATTACH COMPLETE.

2.2.12 Dedicated EPS Bearer Context Activation Success Ratio [2]

2.2.12.1 Abstract definition

The dedicated EPS bearer context activation success ratio measures the probability that a dedicated bearer can be activated. It is the proportion of successful dedicated bearer context activation attempts and the total number of dedicated bearer activation attempts.

See ETSI TS 102 250-2 (5.13.1 Dedicated EPS Bearer Context Activation Failure Ratio)

2.2.12.2 Computation

2.2.12.2.1 **Abstract formula:**

$$\text{Dedicated EPS Bearer Context Activation success Ratio }[\%] = \frac{\text{dedicated EPS bearer activation successes}}{\text{dedicated EPS bearer activation initiations}} \times 100$$

2.2.12.2.2 **Trigger points:**

The UE requests bearer resource allocation (or modification) procedure (BEARER RESOURCE ALLOCATION REQUEST or BEARER RESOURCE MODIFICATION REQUEST). The activation of the dedicated EPS bearer is considered as initiated when the ACTIVATE DEDICATED EPS BEARER CONTEXT REQUEST is received by the UE.

2.2.12.2.3 **Remarks:**

Precondition: Successful PDN connection for the APN, for which a Dedicated EPS Bearer shall be established, already exists

2.2.13 Dedicated EPS Bearer Context Activation Time [2]

2.2.13.1 Abstract definition

The Dedicated EPS bearer context activation time is the time that is needed to establish a dedicated bearer for user data transfer.

2.2.13.1.1 See ETSI TS 102 250-2 (5.13.2 Dedicated EPS Bearer Context Activation Time)

Abstract formula:

2.2.13.1.2

$$\text{Dedicated EPS Bearer Context Activation Time }[s] = \left(t_{\text{Dedicated EPS Bearer context activation success}} - t_{\text{Dedicated EPS Bearer context activation initiation}}\right)[s]$$

Trigger points:

BEARER RESOURCE ALLOCATION (or MODIFICATION) REQUEST is sent by the UE.
The ACTIVATE DEDICATED EPS BEARER CONTEXT ACCEPT is sent by the UE.

Remarks:

The dedicated EPS bearer context activation procedure is triggered by the UE for bearer resource allocation and completed when the allocated dedicated EPS bearer context is accepted by the UE.

2.2.13.1.3

2.2.14 EPS Bearer Context Cut-off Ratio [%] [2]

2.2.14.1 Abstract Definition

The default or dedicated EPS bearer context cut-off ratio measures whether a default or a dedicated EPS bearer context is deactivated without being initiated intentionally by the user[3].

See ETSI TS 102 250-2 (5.12.3 Default EPS Bearer Context Cut-off ratio, 5.13.3 Dedicated EPS Bearer Context Cut-off ratio)

2.2.14.2 Computation

2.2.14.2.1 **Abstract Formula**

$$\text{Default} \mid \text{Dedicated EPS Bearer Context Cut-off Ratio [\%]} = \frac{\text{default} \mid \text{dedicated EPS bearer context losses not initiated by the user}}{\text{successfully activated default} \mid \text{dedicated EPS bearer contexts}} \times 100$$

2.2.14.2.2 **Trigger Points**

Different trigger points for a EPS bearer context deactivation not initiated intentionally by the user are possible. The EPS bearer context will be deactivated by the network. The UE receives from the network the message DEACTIVATE EPS BEARER CONTEXT REQUEST, MODIFY EPS BEARER CONTEXT REQUEST or DETACH REQUEST.

2.2.14.2.3 **Remarks**

Default EPS bearer context (for the default APN) loss means UE EPS detach.

2.2.15 DNS Host Name Resolution Success Ratio [%] [2]

2.2.15.1 Abstract Definition

The DNS host name resolution success ratio is the probability for a host name to host address translation of a DNS resolver is successful.

See ETSI TS 102 250-2 (5.10 DNS Host Name Resolution Failure Ratio)

[3] The default EPS bearer context for the default APN is indispensable to be maintained to use any service over LTE networks or any additional default EPS bearer context. Deactivating the last default EPS bearer will cause the UE detach.

2.2.15.2 Computation

2.2.15.2.1 Abstract Formula

$$\text{DNS Host Name Resolution Success Ratio [\%]} = \frac{\text{successful DNS host name resolution requests}}{\text{DNS host name resolution requests}} \times 100$$

2.2.15.2.2 Trigger points

Start: Request to resolve a host address from DNS server, or DNS protocol data packet containing DNS type A (host address) “Standard query” query for the desired host name.

Stop: Host address received from DNS server, or DNS protocol data packet received containing a type A (host address) “Standard query response, No error” response, the respective type A “Standard query” query and an answer including the desired host name to host address translation.

2.2.15.2.3 Remarks

Precondition for measurement:

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

Since messages carried by UDP are restricted to 512 bytes. UDP is the recommended method for standard queries on the Internet.

The KPI is relevant only for PS services.

2.2.16 DNS Host Name Resolution Time [s] [2]

2.2.16.1 Abstract Definition

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

See ETSI TS 102 250-2 (5.11 DNS Host Name Resolution Time)

2.2.16.2 Computation

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

2.2.16.2.1 Abstract Formula

$$\text{DNS Host Name Resolution Time [s]} = \left(t_{\text{StandardQuery Response}} - t_{\text{StandardQuery}}\right)[s]$$

2.2.16.2.2 Trigger points

Start: Request to resolve a host address from DNS server, or DNS protocol data packet containing DNS type A (host address) “Standard query” query for the desired host name

Stop: Host address received from DNS server, or DNS protocol data packet received containing a type A (host address) “Standard query response, No error” response, the respective type A “Standard query” query and an answer including the desired host name to host address translation.

2.2.16.2.3 Remarks

Precondition for measurement:

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

For static measurement methodologies, as defined in TS 102 250-3 [1], the queried DNS name server shall have any data related to the host name to be resolved available as authoritative data in one of the name server’s zones, so that no recursive lookups have to be performed and no use of cached information will be required.

If the related data is not stored locally in the name server’s zone, the resolution time would vary due to DNS caching strategies

The KPI is relevant only for PS services.

2.3 Telephony Service

To simplify the description of the voice call KPIs, R-party and H-party are used in the context where R-party is the roaming side in VPMN and H-party is in HPMN.

A precondition for the applicability of this section to the voice CSFB:

The LTE networks (VPMN) support voice CSFB. The UE (represented by R-party) are configured as voice centric and CS voice only or CS voice preferred, IMS PS Voice as secondary.

2.3.1 Service Accessibility Telephony - MO (SA-T-MO) [2] [12] [14]

2.3.1.1 Abstract definition

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

See ITU-T Rec. E.800: The probability that a service can be obtained within specified tolerances and other given operating conditions when requested by the user. The term NER (Network Effectiveness Ratio defined in ITU-E 425) is understood as Service Accessibility Telephony.

See complementary Service Non-Accessibility Telephony (SNAT) in ETSI TS 102 250-2 (6.6.1 Service non-accessibility).

2.3.1.2 Computation

For a successful call attempt:

CS, CSFB

- The user hears the alerting tone, or R-party receives ALERTING

- H-party rings, or sends ALERTING

2.3.1.2.1 **Abstract formula:**

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

2.3.1.2.2 **Trigger points:**

Beginning of the call attempt::	Successful pressing send button (it is important to check, if coverage has been given when the send button is pressed, otherwise this Call Attempt counts to Network Non Accessibility (NNA)).
Successful call attempt:	Measurement (alerting[4] tone heard or ALERTING (CC message) received by R-party), and H-party rings or sends ALERTING (CC message).

2.3.1.2.3 **Remarks:**

SAT/NER is not catching the voice carriers that are sending fake ring tones like sending fake ACM/CPG messages.

2.3.2 Service Accessibility Telephony - MT (SA-T-MT) [2] [14]

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

See ITU-T Rec. E.800: The probability that a service can be obtained within specified tolerances and other given operating conditions when requested by the user. The term NER (Network Effectiveness Ratio defined in ITU-E 425) can be understood as Service Accessibility Telephony.

See complementary Service Non-Accessibility Telephony (SNAT) in ETSI TS 102 250-2 (6.6.1 Service non-accessibility).

2.3.3.1 Computation

For a successful call attempt:
CS, CSFB
The user hears the ringing or R-party sends ALERTING (CC message)

H-party hears the alerting tone or receives ALERTING (CC message)

[4] Due to network problems and despite H-party being not busy, it may even be possible for the R-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.

Abstract formula:

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

2.3.3.1.1

Trigger points:

2.3.3.1.2

Beginning of the call attempt of H-party:	Successful pressing send button (it is important to check, if coverage has been given when the send button is pressed, otherwise this Call Attempt counts to Network Non Accessibility (NNA))
CS, CSFB	
Successful call attempt:	Measurement (R-party rings or sends ALERTING (CC message)
	Alerting tone heard or ALERTING (CC message) received by H-party).

2.3.4 Setup Time Telephony – MO (ST-T-MO) [1] [12] [13]Abstract definition

Time between sending of complete address information and receipt of call setup notification as defined in ETSI TS 102 250-2 (6.6.2 Setup Time).

See ITU-T Rec. E.800: The expectation of the time duration between an initial bid by the user for the acquisition of a service and the instant of time the user has access to the service, the service being obtained within specified tolerances and other given operating conditions.

The term ST-T can also be understood as PDD (Post Dialling Delay); see ITU-E 431.

2.3.4.2 Computation

For a successful call attempt:

CS, CSFB

- The user hears the alerting tone, or R-party receives ALERTING
- H-party rings or sends ALERTING (CC message).

2.3.4.2.1

Abstract formula:

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

t_2: point of time where signalling connect is established (alerting tone is heard or ALERTING (CC message) is received by test equipment))

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

Trigger points:

2.3.4.2.2

Beginning of Setup Time measurement:	Successful pressing send button at R-party (it is important to check, if coverage has been given, otherwise this Call Attempt counts to Network Non Accessibility (NNA))
CS, CSFB	
Successful signalling connection:	Measurement (alerting tone heard or ALERTING (CC message) received by R-party).

2.3.5 Setup Time Telephony – MT (ST-T-MT) [2] [12] [13]Abstract definition

Time between sending of complete address information and receipt of call setup notification as defined in ETSI TS 102 250-2 (6.6.2 Setup Time).

ITU-T Rec. E.800: The expectation of the time duration between an initial bid by the user for the acquisition of a service and the instant of time the user has access to the service, the service being obtained within specified tolerances and other given operating conditions.

The term ST-T can also be understood as PDD (Post Dialling Delay); see ITU-E 431.

2.3.5.2 Computation

2.3.5.2.1 Abstract formula:

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

t_2: point of time where connect is established (for example alerting[5] or subscriber busy is detected by test equipment))

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

t_2: point of time where signalling connect is established (for example ringing[6] or ALERTING (CC message) sent by test equipment))

2.3.5.2.2 t_1: point of time where the user presses the send button on mobile equipment

Trigger points:

Beginning of Setup Time measurement:	Successful pressing send button at H-party (it is important to check, if coverage has been given, otherwise this Call Attempt counts to Network Non Accessibility (NNA))

[5] If an end to end connection is not established, this measurement must be ignored. It is assumed that early traffic channel assignment is used.

[6] If an end to end signalling connection is not established, this measurement must be ignored. It is assumed that early traffic channel assignment for GSM is used.

CS, CSFB

Successful signalling connection:	Measurement (ringing or ALERTING (CC message) sent by R-party).

2.3.6 CSSR - MO (Call Setup Success Ratio) [7]Abstract definition

CSSR expresses the relationship between the number of seizures and the sum of the number of seizures resulting in a successful established call.

Call Setup Success Ratio is defined in 3GPP TS 29 002.

2.3.6.2 Computation

2.3.6.2.1 Abstract formula:

$$\boxed{CSSR\ [\%] = \frac{\text{Number of successfulcall establishments}}{\text{Number of call attempts}} * 100\%}$$

2.3.6.2.2 Trigger points:

Beginning of the call attempt of R-party:	Successful pressing send button (it is important to check, if coverage has been given when the send button is pressed, otherwise this Call Attempt counts to Network Non Accessibility (NNA)).
Successful call establishment:	Open connection between R-party and H-party, where both parties can hear each other.
CS, CSFB	
	R-party receives CONNECT and H-party receives CONNECT ACKNOWLEGE (CC messages)

2.3.7 CSSR - MT (Call Setup Success Ratio) [7]Abstract definition

CSSR expresses the relationship between the number of seizures and the sum of the number of seizures resulting in a successful established call.

CSSR is defined in 3GPP 29 002.

2.3.7.2 Computation

2.3.7.2.1 Abstract formula:

It is assumed that the A Party is in the home network in order to avoid an international tromboning of the call between the VPMN and the HPMN.

$$\text{CSSR [\%]} = \frac{\text{Number of successful call establishments}}{\text{Number of call attempts}} * 100\%$$

2.3.7.2.2 **Trigger points:**

Beginning of the call attempt of H-party:	Successful pressing send button (it is important to check, if coverage has been given when the send button is pressed, otherwise this Call Attempt counts to Network Non Accessibility (NNA)).
Successful call establishment:	Open connection between R-party and H-party, where both parties can hear each other.
CS, CSFB	
	R-party receives CONNECT ACKNOWLEGE and H-party receives CONNECT (CC messages)

2.3.8 REL (ISUPv2 signalling transparency) [16]Abstract definition

Effective uncorrupted transmission by the VPMN of the Cause Value in the Release (REL) ISDN User Part (ISUP) messages, as defined in ITU-T Q.850.

2.3.8.1.1 **Testing protocol:**

- The HPMN sends to the VPMN a REL ISUP message with a valid populated 'Cause Value' field
- The VPMN must send back to the HPMN the REL ISUP message with the same 'Cause Value'
- The HPMN uses SS7 monitoring tool to measure the key performance indicator (KPI)

2.3.8.2 Computation

2.3.8.2.1 **Abstract formula:**

$$\text{REL} - \text{CV [\%]} = \frac{\text{Number of uncorrupted Cause Value in REL ISUP messages}}{\text{Number of sent REL ISUP messages with populated Cause Value}} \cdot 100\,\%$$

2.3.9 OCN & RDN (ISUPv2 signalling transparency) [17]Abstract definition

Effective uncorrupted transmission by the VPMN of Original Called Number, (OCN) and Redirecting Number (RDN), as defined in ITU-T Q.732.2.

Using a testing tool:

- R, H and C all belong to the HPMN
- R is roaming on the VPMN network while H and C are located in the HPMN
- A late call forward takes places and is not optimally routed (that is, the calls is effectively routed towards the R destination and returned to the HPMN).

2.3.9.2 Computation

Abstract formula:

$$\text{OCN \& RDN}[\%] = \frac{\text{Number of Call Forwards including OCN/RDN}}{\text{Number of Call Forward tests}} \cdot 100\%$$

2.3.9.2.1

Trigger points:

OCN & RDN are correctly transmitted.

2.3.9.2.2

2.3.10 Call Completion Ratio Circuit Switched Telephony (CCR-CS-T) [2]

2.3.10.1 Abstract definition

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

See ETSI TS 102 250-2 (6.6.5 Cut-off Call Ratio).

See also complementary QoS Indicator: Call Non-Completion Rate circuit switched (CNCR-CS).

2.3.10.2 Computation

2.3.10.2.1 **Abstract formula:**

$$\text{CCR - CS - T}[\%] = \frac{\text{Number of intentionally terminated telephony calls}}{\text{Number of successful telephony call attempts}} \cdot 100\,\%$$

2.3.10.2.2 **Trigger points:**

Successful call attempt:	Connect measurement (alerting' tone or ALERTING message detected by R-party)
Terminated call:	Release of connection directly by R- or H-party

2.3.11 ALOC (Average Length of a Call) [4]

2.3.11.1 Abstract Definition

As defined in ITU E-437: average duration of calls. The advice is to measure this for MOC and MTC separately, as there could be a significant natural difference between these 2 call types.

2.3.11.2 Computation

Using traffic report:

$$\boxed{ALOC\ MOC[seconds] = \frac{\text{Total network usage of MOC seconds in a month}}{\text{Total number of MOC calls in a month}}}$$

$$\boxed{ALOC\ MTC[seconds] = \frac{\text{Total network usage of MTC seconds in a month}}{\text{Total number of MTC calls in a month}}}$$

2.3.12 CLI Transparency [5]

Call Line Identification (CLI) between countries is often not transmitted, the display indicating “PRIVATE”, "UNAVAILABLE" or "INTERNATIONAL". This is usually the case with different network types and international roaming scenarios.

Overseas number may be compressed into a "domestic" format and thus possibly not be recognizable: e.g. a US number <**1** 555 555 7878> may be displayed in the UK as <555 555 7878>, instead of <**001** 555 555 7878> (or as **+1** 555 555 7878), where the "+" represents the access code to dial international numbers).

2.3.12.1 Abstract Definition

CLI needs to be delivered correctly and complete, in a way it can be used to dial back the original called party.

2.3.12.2 Computation

$$\boxed{\text{CLI transparency}[\%] = \frac{\text{Number of complete and correct CLI's}}{\text{Number of calls}} * 100\%}$$

2.3.13 Speech Quality on Call Basis (SpQ) [2] [21] [27] [28] [29]Abstract definition

Telephony speech quality on call basis is an indicator representing the quantification of the end-to-end speech transmission quality of the Mobile Telephony Service. This parameter computes the speech quality on the basis of completed calls[7].

2.3.13.2 Computation

The validation of the end-to-end quality is made using MOS-LQO scales. These scales describe the opinion of users with speech transmission and its troubles (noise, robot voice, echo, dropouts and so on), according to ITU-T Recommendation P.862 PESQ in conjunction with ITU-T Recommendation P.862.1, or according to ITU-T Recommendation P.863 POLQA. The algorithm used should be reported. The speech quality measurement is taken per call. An aggregation should be made on one value for speech quality per call.

2.3.13.2.1 Abstract formula:

CS, CSFB

[7] The acoustic behaviour of mobile terminals is not part of this speech quality measurement.

$$\text{Telephony Speech Quality on Call Basis (received R - party)} = \text{f}\left(\text{MOS - LQO}\right)$$
$$\text{Telephony Speech Quality on Call Basis (received H - party)} = \text{f}\left(\text{MOS - LQO}\right)$$

Optionally it might be useful to aggregate both speech quality values into one. In this case the worst of both shall be used. This aggregated speech quality value shall be called SpQ (min).

Trigger points:

Beginning of connection:	Interchange speech samples between R-party and H-party
End of connection:	Release of connection

2.3.13.2.2

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

2.3.14 CSFB Return to LTE Success Ratio

2.3.14.1 Abstract definition

This parameter measures the probability that a UE has to re-join the LTE network after a CSFB call within a pre-determined time interval.

For a valid calculation the following preconditions must be met:

- LTE coverage is present at the end of the CSFB call
- A call is successfully established (with or without CSFB at H-Party)
- The call is regularly disconnected (i.e. no drop)

2.3.14.2 Computation

2.3.14.2.1

Abstract formula

2.3.14.2.2

$$\text{Return to LTE Success Ratio [\%]} = \frac{\text{successful Return to LTE attempts}}{\text{all Return to LTE attempts}} \times 100$$

Trigger points

Start: Hang up the call

Stop: network type indicator on the UE’s display switches to LTE[8]Remarks

This KPI shall be separately tested for CSFB voice MO call and MT call.

[8] In a measurement system with automatic call dialling this KPI is influenced by the call interval. In particular the call interval will implicitly set the timeout for Return to LTE calculation.

2.3.15 CSFB Return to LTE TimeAbstract definition

This parameter measures the time needed by the UE to re-join the LTE network after a CSFB call.

2.3.15.2 Computation

2.3.15.2.1 Abstract formula

$$\text{Return to LTE Time [s]} = \left(t_{\text{first SIB message in LTE received}} - t_{\text{call disconnected}}\right)[\text{s}]$$

2.3.15.2.2 Trigger points

$t_{\text{call disconnected}}$ time when the call is disconnected

$t_{\text{first SIB message in LTE received}}$ time when the first SIB message in LTE is received

2.3.15.3 Remarks

This KPI shall be separately tested for CSFB voice MO call and MT call.

2.4 Short Message Service

The SMS KPIs are applied to SMS over GSM, UMTS CS or SMSoSGs.

2.4.1 Service Accessibility SMS MO (SA SMS MO) [2]

2.4.1.1 Abstract definition

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

See ETSI TS 102 250-2 (7.4.2 SMS Service non-accessibility).

2.4.1.2 Computation

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

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

2.4.1.2.1 Abstract formula:

2.4.1.2.2

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

Trigger points [for example Layer-3 messages]:

Start SMS service attempt:	Initiate sending an SMS at Roaming side
Successful SMS service attempt:	Receiving acknowledgement of the SMSC

2.4.2 Service Accessibility SMS MT (SA SMS MT) [2]Abstract definition

Probability that the end-user can receive a Short Message from its Home Network SMS-C while it is offered by display of the network indicator on the Mobile Equipment. In this case the user wants to receive a Short Message.

See ETSI TS 102 250-2 (7.4.2 SMS Service non-accessibility).

2.4.2.2 Computation

Only the valid attempts have to be measured. Errors due to user mistake (for example memory full) should be excluded.

Abstract formula:

2.4.2.2.1

$$\boxed{\text{Service Accessibility SMS MT [\%]} = \frac{\text{Number of successful SMS - MT service attempts}}{\text{Number of all SMS - MT service attempts}}}$$

Trigger points [for example Layer-3 messages]:

2.4.2.2.2

Start SMS service attempt:	Initiate sending a SMS from Home SMS-C
Successful SMS service attempt:	Receiving Short Message at Roaming side

2.4.3 Access Delay SMS MO (AD SMS-MO) [2]Abstract definition

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

See ETSI TS 102 250-2 (7.4.3 SMS Access Delay).

2.4.3.2 Computation

2.4.3.2.1

Abstract formula:

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

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

2.4.3.2.2

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

Trigger points [for example Layer-3 messages]:

Start SMS service attempt:	Initiate sending an SMS at Roaming side
Successful SMS service attempt:	Receiving acknowledgement of the SMSC

2.4.4 Access Delay SMS MT (AD SMS-MT) [2]Abstract definition

Time between sending a Short Message from the Home Short Message Centre and receiving the notification at the Short Message Centre.

See ETSI TS 102 250-2 (7.4.3 SMS Access Delay).

2.4.4.2 Computation

2.4.4.2.1 **Abstract formula:**

$$\boxed{\text{Access Delay SMS MT}\,[s] = \left(t_{receive} - t_{send SMS}\right)[s]}$$

$t_{receive}$: point of time the SMS Centre receives confirmation that the Short Message was correctly delivered.

$t_{send\ SMS}$: point of time the Short Message leaves the SMS Centre

2.4.4.2.2 **Trigger points [for example Layer-3 messages]:**

Start SMS service attempt:	Initiate sending a SMS from Home SMS-C
Successful SMS service attempt:	Receiving Short Message Confirmation of Delivery at roaming side

2.4.5 End-to-End Delivery Time for SMS MO (E2E DT SMS-MO) [2]Abstract definition

The SMS end-to-end delivery time is the time between sending a short message from R-party in VPMN to a Short Message Centre and H-party in HPMN receiving the very same short message from the Short Message Centre.

See ETSI TS 102 250-2 (7.4.5 End-to-End Delivery Time).

2.4.5.2 Computation

2.4.5.2.1 **Abstract formula:**

$$\boxed{\text{SMS MO End-to-End Delivery Time}\,[s] = \left(t_{H,receive} - t_{R,send}\right)[s]}$$

$t_{H,receive}$: point of time the H-party in the HPMN receives the short message from the SMS Centre.

2.4.5.2.2 $t_{R,send}$: point of time the user sends the SMS to the SMS Centre.

Remarks:

Not relevant for QoS Roaming SLA since time measured is dependent on the performance of the HPMN SMS-C.

2.4.6 End-to-End Delivery Time for SMS MT (E2E DT SMS-MT) [2]Abstract definition

The SMS end-to-end delivery time is the time between H-party in HPMN sending a short message to a Short Message Centre and R-party in VPMN receiving the very same short message from the Short Message Centre.

See ETSI TS 102 250-2 (7.4.5 End-to-End Delivery Time).

2.4.6.2 Computation

2.4.6.2.1 **Abstract formula:**

$$\boxed{\text{SMS MT End-to-End Delivery Time [s]} = \left(t_{R,receive} - t_{H,send}\right)[s]}$$

$t_{R,receive}$: point of time the R-party in the VPMN receives the new short message from the SMS Centre.

$t_{H,send}$: point of time the H-party sends a short message to the SMS Centre in the HPMN.

2.4.6.2.2 **Remarks:**

Not relevant for QoS Roaming SLA since time measured is dependent on the performance of the HPMN SMS-C.

2.4.7 Completion Ratio SMS (CR SMS) [2] [15]Abstract definition

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

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

For test and measurement purposes a message is considered valid if it is delivered successfully within a time window defined.

See ETSI TS 102 250-2 (7.4.4 SMS Completion Failure Ratio)

2.4.7.2 Computation

2.4.7.2.1 **Abstract formula:**

$$\boxed{\text{CR SMS CS [\%]} = \frac{\text{successful received Test SMS - duplicate received Test SMS - corrupted Test SMS}}{\text{Number of all send Test SMS}}}$$

2.4.7.2.2 **Trigger points:**

Successfully send and received SMS via SMSC.

Time window of measurements according to user profile.

Circuit Switched Data Service

2.5.1 Service Accessibility, Circuit Switched Data (SA –CSD) [15]

2.5.1.1 Abstract definition

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

Probability that the end-user's DTE can access the Mobile Data Service when requested.

There are 2 layers of accessibility for CSD

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

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

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

- ATDT command including target number.
- Receive Connect from target network DCE
- Send relevant command to target Data Server.
- Receive valid response from Data Server

The specific commands and responses from data servers will be detailed in 'Typical procedures for quality of service measurement equipment'.

2.5.1.2 Computation

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

2.5.1.2.1

Abstract formula:

2.5.1.2.2

$$\text{Service Accessibility CSD} = \frac{\text{Number of successful call attempts}}{\text{Number of call attempts}}$$

Trigger points:

Beginning of the call attempt:	ATDT command with dialled number sent by A-party DTE.
Successful call attempt:	Valid response received from Data Server.

2.5.2 Set-up Time (ST – CSD)Abstract definition

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

2.5.2.2 Computation

Abstract formula:

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

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

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

Trigger points:

2.5.2.2.2

Beginning of the Set-up time measurement:	Sending of ATDT command by A-party
Successful connection:	Valid response received from Data Server.

2.5.3 Data Quality (DQ-CSD)

For definitions of Data Quality Parameters refer to section 2.7.

2.5.3.1 Completion Ratio Circuit Switched Data (CR-CSD)

2.5.3.1 Abstract definition

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

2.5.3.2 Computation

2.5.3.2.1

Abstract formula:

$$\boxed{\text{Call completion Ratio CSD } = \frac{\text{Number of calls terminated by end users}}{\text{Number of successful data call attempts}}}$$

2.5.3.2.2

Trigger points:

Successful call attempt:	Valid response received by A-party DTE.
Completed call:	DTE ‘ready’ only when call ended by either party intentionally.

2.6

Packet Switched Data Service (General Packet Radio Service)

For test purposes it will be necessary to have the mobile test equipment in a stable state before testing. For each test the mobile should begin by being powered on and attached but not PDP context activated. Specific details are to be found in ‘Typical procedures for quality of service measurements’.

Note: The bearer technology will affect the monitoring results for many of the Packet Switched Data measurements.

2.6.1 Service Accessibility Ratio – Packet Switched Data (SA – PSD) [2] [7]

2.6.1.1 Abstract definition

Probability that a subscriber can successfully attach to the PS network. As defined in ETSI TS 129 002. See also ETSI TS 102 250-2 (5.3 Attach Failure ratio).

There are 2 layers of accessibility for GPRS:

- Access to the mobile network GPRS core infrastructure.
- Access to the required data service provided by a data server.

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

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

- Sending of valid command (for example ATD*99# (with IP address of target server)) from A party DTE to obtain IP connection.
- Receive valid response from GGSN
- Send valid command to target Data server.
- Receive valid response from target Data server.

The specific commands and responses from data servers will be detailed in 'Typical procedures for quality of service measurements'.

If multiple Access Point Names (APNs) are used, the measures should be performed on a per-APN basis for consistency in measurements.

2.6.1.2 Computation

A session will be considered set-up successfully if a valid response is received from the target data server

2.6.1.2.1 Abstract formula:

$$\text{Service Accessibility PSD} = \frac{\text{Number of successful session attempts}}{\text{Number of session attempts}}$$

2.6.1.2.2 Trigger points:

Beginning of the session attempt:	Send valid command request (for example ATD*99# (with IP address of target server))
Successful session attempt:	Valid response received from target data server

2.6.2 Set-up Time – Packet Switched Data (ST – PSD) [2]Abstract definition

Time between sending of valid command (for example ATD*99# (with IP address of target server) message and receipt of valid response message from target data server.

2.6.2.2 Computation

A session will be considered set-up successfully if a valid response is received from the target data server

Abstract formula:

$$\boxed{\text{Set-up Time Packet Switched Data [s]} = t_2 - t_1}$$

2.6.2.2.1 t1: point of time where A-party valid session request command

t2: point of time where connect is established (valid response received by A-party from data server)

Trigger points:

2.6.2.2.2	Beginning of the session attempt:	Send valid command request (for example ATD*99# (with IP address of target server))
	Successful session attempt:	Valid response received from target data server

Note for all data quality testing it is assumed that for each test, PDP Context is activated and at the end of the individual test PDP Context is de-activated.

For definitions of Data Quality Parameters refer to section 2.7.

2.6.3 Service Integrity - Throughput (Kbit/sec) [2]

2.6.3.1 Abstract definition

This parameter describes the average data transfer rate at the network transport level (and not at the User Application level), based on the Mean Data Rate as defined by ETSI TS 102 250-2 (6.1.7 Mean Data Rate).

The prerequisite for this parameter is network and service access.

2.6.3.2 Computation

Abstract formula:

2.6.3.2.2

$$\boxed{BitPipeThroughput = \frac{VolumeOfDataTransferred}{TransferTime}}$$

Remarks:

The measurement of Throughput will be influenced by Packet Loss and Roundtrip Time (Delay). Throughput measurements may also be influenced by service-side factors such as radio cell reservation and network usage. Mobile Station ()

2.6.4 Service Integrity - Goodput (Kbit/sec) [10]Abstract definition

This parameter describes the average data transfer rate at the User Application level (and not at the network transport level).

The prerequisite for this parameter is network and service access.

Goodput is defined in IETF RFC2647.

2.6.4.2 Computation

Abstract formula:

Goodput may be calculated as:

2.6.4.2.1

$$BitPipeGoodput = \frac{VolumeOfUsefulDataTransferred}{TransferTime}$$

Please note that the definition of “useful data” depends on the user applications used for the measurement.

Remarks:

2.6.4.2.2 The measurement of Goodput will be influenced by Packet Loss and Roundtrip Time (Delay). Goodput measurements may also be influenced by service-side factors such as packet size and the User Application.

2.6.5 Service Integrity - Roundtrip Time [11]Abstract definition

Roundtrip Time (Roundtrip Delay) is the total time that it takes to transmit an IP packet from the source to the destination and receive the reply packet from the destination at the source.

The prerequisite for this parameter is network and service access.

See ‘Delay’ in section 8 of PRD IR.34 ‘Inter-Service Provider IP Backbone Guidelines’.

2.6.5.2 Computation

2.6.5.2.1

Abstract formula:

$$Round\ Trip\ Time\ (ms) = (timestamp\ Packet\ received) - (timestamp\ Packet\ sent)$$

2.6.6 Service Integrity – Packet Loss [6] [11]

2.6.6.1 Abstract definition

Packet Loss is the ratio of dropped packets to all packets sent from the source to Destination over a given period of time.

The prerequisite for this parameter is network and service access.

See ‘Packet Loss Ratio’ in section 8 of PRD IR.34 ‘Inter-Service Provider IP Backbone Guidelines’.

See also ‘Packet Loss’ in ITU-T Y.1540.

2.6.6.2 Computation

Abstract formula:

2.6.6.2.1

$$Packet\_loss = 100\,\% \times \left(1 - \frac{PacketSent}{Packet\mathrm{Re}\,ceived}\right)$$

- The IP address should be one of the HPMN network (GGSN, WAP GW, tests server, …)
- The firewalls are not allowed to block these ICMP echo requests for the tested IP address and Port

2.6.7 Completed Session Ratio (CoSeR – PSD)Abstract definition

Probability that a successful session attempt is not released for a reason other than intentional by any of the parties involved in the session.

2.6.7.2 Computation

2.6.7.2.1

Abstract formula:

2.6.7.2.2

$$\text{Completed Session Ratio PSD} = \frac{\text{Number of sessions not released other than by end user}}{\text{Number of successful data session attempts}}.$$

Trigger points:

Successful session attempt:	Valid response received from target data server.
Completed session:	Session released intentionally by either end-user.

2.6.8 Service Retainability – Average PDP Context Average Session Time [8]Abstract definition

The average PDP context session time is the average duration of the PDP context sessions successfully completed.

PDP Context is defined in 3GPP 29.060.

2.6.8.2 Computation

Abstract formula:

$$PDPContextAverageSessionTime = Average(t_{PDPContextDeactivation} - t_{PDPContextActivation})$$

2.6.8.2.1 Trigger points:

Start:	Notification of successful PDP context activation (Activate PDP Context Accept)
End:	PDP context deactivation request initiated by the user (Deactivate PDP Context Request)

2.6.8.2.2

2.6.8.2.3 Remarks:

The PS bearer has to be active in the cell used by a subscriber (cf. Unavailability) and the mobile station has to be attached as well as the respective PDP context has to be activated.

2.7 Data Service Class Definitions and Measurements

The following definitions for data services and data quality DQ are relevant for both circuit switched and packet switched data as, the different classes of data service will be applied identically irrespective of the data bearer system.

Note that data quality will be a result of an overall call or session. For test purposes it may be desirable to break this down into geographically distinct measurements but for QoS reporting should be kept to call or session lengths.

Data classes are defined in 3rd Generation Partnership Project Technical Specification Group Services and System Aspects; QoS Concept and Architecture (3G TR 23.907) - see Table 1.

Traffic class	Conversational class conversational RT	Streaming class streaming RT	Interactive class Interactive best effort	Background Background best effort
Fundamental characteristics	• Preserve time relation (variation) between information entities of the stream • Conversational pattern (stringent and low delay)	• Preserve time relation (variation) between information entities of the stream	• Request response pattern • Preserve payload content	• Destination is not expecting the data within a certain time • Preserve payload content
Example of the application	• Voice	• Streaming video	• Web browsing	• Background download of emails

Table 2: UMTS QoS classes

2.7.1 Conversational Class Data

2.7.1.1 Abstract definition

Indicator representing the end-to-end data transmission quality of the Conversational Class Data Service. This represents full duplex transfer of data in near real time.

2.7.1.2 Computation

The end-to-end data quality is validated by measuring the average data throughput in both up-link and down link direction on a best effort basis. The data throughput measurement will be computed and averaged over the duration of the session/call and reported in bits per second. Additionally the minimum throughput averaged over 10% of the overall call/session length, the maximum throughput over 10% of the overall call/session length and worst. The worst delay time for the call/session should also be reported

2.7.1.2.1 Abstract formula:

$$\boxed{\begin{array}{l}\text{DQ(received A-side)} = \text{X bits/sec} \\ \text{DQ(received B-side)} = \text{X bits/sec}\end{array}}$$

2.7.1.2.2 Trigger points:

Beginning of call/session data sample:	Interchange data frames of predefined data between A and B-party DTE
End of call/session data sample:	Calculation of average data throughput for Call/session data sample

2.7.2 Streaming ClassAbstract definition

Indicator representing the end-to-end data transmission quality of the Mobile, Circuit Switched, Streaming Class Data Service. This measure represents a delivery of data in one direction (up-link or down-link) in near real time for example video broadcast.

Additionally the minimum throughput averaged over 10% of the call/session duration, the maximum throughput averaged over 10% of the call/session duration and the worst block error rate. The worst delay time for the call/session should also be reported

Note for streaming class service only the down link direction is considered, but if service applications are introduced for uplink streaming then this can be added for calculation for data received by B-Party

2.7.2.2 Computation

The end-to-end data quality is validated by measuring the data throughput in down link direction on a best effort basis. The data throughput measurement will be computed and averaged over the duration of the call/session and be reported in bits/sec.

2.7.2.2.1 **Abstract formula:**

$$\boxed{DQ(\text{received A-side}) = X \text{ bits/sec}}$$

2.7.2.2.2 **Trigger points:**

Beginning of Call/session data sample:	Transmission of data frames of indexed predefined data B-party to A-party
End of Call/session data sample:	Calculation of average data throughput for call/session data sample

2.7.3 Interactive ClassAbstract definition

Indicator representing the end-to-end data transmission quality of the Mobile Circuit Switched Interactive Class Data Service. This represents duplex transfer of data in non real-time.

2.7.3.2 Computation

The validation of the end-to-end data quality is made by the time taken to download specified files of fixed data size to the A-party DTE when, requested by the A-party sending a request to the data server.

Assumption: The A-party DTE has already been connected to the data server as part of the call set-up process.

2.7.3.2.1 **Abstract formula:**

$$\boxed{DQ \text{ download time } [s] = t_2 - t_1}$$

t_1: point of time where A-party DTE sends data request.

t_2: point of time where A-party receives complete uncorrupted requested file/s

2.7.3.2.2 **Trigger points:**

Beginning of request for download:	Data request sent by A-party DTE
Download of file/s complete:	Uncorrupted file/s received by A-party DTE.

2.7.4 Background classAbstract definition

Indicator representing the end-to-end data transmission quality of the Mobile Circuit Switched Background Class Data Service. This represents data transfer with no real-time dependency (although for QoS testing, data transfer time is measured).

2.7.4.2 Computation

The validation of the end-to-end data quality is made by the time taken to download a file/s of fixed data size to the A-party DTE when, requested by the A-party sending a request to the target server.

Assumption: The A-party DTE has already been connected to the data server as part of call set-up process.

Abstract formula:

$$\boxed{\text{DQ File download time } [s] = t_2 - t_1}$$

2.7.4.2.1 t_1: point of time where A-party DTE sends data transfer request

t_2: point of time where A-party receives complete uncorrupted file/s

Trigger points:

2.7.4.2.2

Beginning of request for download:	Request sent by A-party DTE
Download of file/s complete:	Uncorrupted file/s received by A-party DTE

2.8 FTP QoS Parameters

Precondition: PS attached and the respective PDP context / default EPS bearer context activated.

Assumption: a single TCP/IP connection is applied.

2.8.1 FTP {Download|Upload} IP Service Access Success Ratio [%] [2]

2.8.1.1 Abstract definition

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

See ETSI TS 102 250-2 (6.1.3 FTP IP-Service Failure Ratio)

2.8.1.2 Computation

2.8.1.2.1

Abstract Formula

2.8.1.2.2

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

Trigger Points

Download:
Start: Initiate file download, or the first [SYN] sent on the data socket.
Stop: File download starts, or
Method A[9]: Reception of the first data packet containing content

[9] Method A is used for payload throughput, method B is used for transaction throughput, ref. to ETSI TS 102 250-2, 4.2 FTP, HTTP and E-mail issues.

Method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.
Upload:
Start: Initiate file upload, or the first [SYN] sent on the data socket.
Stop: File upload starts, or
Method A: Sending the first data packet containing content
Method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.

2.8.2 FTP {Download|Upload} IP Service Access Setup Time [s] [2]Abstract definition

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

See ETSI TS 102 250-2 (6.1.4 FTP IP-Service Setup Time)

2.8.2.2 Computation

2.8.2.2.1 Abstract Formula

$$\text{FTP}\{\text{Download}\,|\,\text{Upload}\}\ \text{IP-Service Setup Time}\,[s] = \left(t_{\text{IP-Serviceaccesssuccessful}} - t_{\text{IP-Serviceaccessstart}}\right)[s]$$

2.8.2.2.2 Trigger Points

Download:
Start: Initiate file download, or the first [SYN] sent on the data socket.
Stop: File download starts, or
Method A: Reception of the first data packet containing content
Method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.
Upload:
Start: Initiate file upload, or the first [SYN] sent on the data socket.
Stop: File upload starts, or
Method A: Sending the first data packet containing content
Method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.

2.8.3 FTP {Download|Upload} Session Success Ratio [%] [2]Abstract definition

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

See ETSI TS 102 250-2 (6.1.5 FTP Session Failure Ratio)

2.8.3.2 Computation

Abstract Formula

2.8

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

2.8.3.2.2 **Trigger Points**

Download:
Start: Initiate file download, or the first [SYN] sent on the control socket.
Stop: File download successfully completed, or Reception of the last data packet containing content
Upload:
Start: Initiate file upload, or the first [SYN] sent on the control socket.
Stop: File upload successfully completed, or Reception of the [FIN, ACK] for the last data packet containing content.

2.8.4 FTP {Download|Upload} Session Time [s] [2]Abstract definition

The session time is the time period needed to successfully complete a file transfer session.

See ETSI TS 102 250-2 (6.1.6 FTP Session Failure Ratio)

2.8.4.2 Computation

2.8.4.2.1 **Abstract Formula**

$$\text{FTP \{Download | Upload\} Session Time [s]} = \left(t_{\text{session end}} - t_{\text{session start}}\right)[\text{s}]$$

2.8.4.2.2 **Trigger Points**

Download:
Start: Initiate file download, or the first [SYN] sent on the control socket.
Stop: File download successfully completed, or Reception of the last data packet containing content
Upload:
Start: Initiate file upload, or the first [SYN] sent on the control socket.
Stop: File upload successfully completed, or Reception of the [FIN, ACK] for the last data packet containing content.

2.8.5 FTP {Download|Upload} Mean Data Rate [kbit/s] [2]Abstract Formula

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

See ETSI TS 102 250-2 (6.1.7 FTP Mean Data Rate)

2.8.5.2 Computation

Abstract Formula

2.8

$$\text{FTP}\{\text{Download}\,|\,\text{Upload}\}\ \text{Mean Data Rate [kbit/s]} = \frac{\text{user data transferred [kbit]}}{\left(t_{\text{data transfercomplete}} - t_{\text{data transferstart}}\right)[\text{s}]}$$

Trigger Points

2.8.5.2.2

The average throughput is measured from opening the data connection to the end of the successful transfer of the content (file).
Download:
Start: File download starts, or
Method A: Reception of the first data packet containing content
Method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket
Stop: File download successfully completed, or Reception of the last data packet containing content.
Upload:
Start: Fie upload starts, or
Method A: Sending of the first data packet containing content
Method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.
Stop: File upload successfully completed, or Reception of the [FIN, ACK] for the last data packet containing content.

2.8.6 FTP {Download|Upload} Data Transfer Success Ratio [%] [2]Abstract definition

The data transfer success ratio is the proportion of completed data transfers and data transfers that were started successfully.

See ETSI TS 102 250-2 (6.1.8 FTP Data Transfer Cut-off Ratio)

2.8.6.2.1

2.8.6.2 Computation

Abstract Formula

2.8.6.2.2

$$\text{FTP}\ \{\text{Download}\,|\,\text{Upload}\}\ \text{Data Transfer Success Ratio [\%]} = \frac{\text{completed data transfers}}{\text{successfuly started data transfers}} \times 100$$

Trigger Points

Download:
Start: File download starts, or
Method A: Reception of the first data packet containing content

Method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.
Stop: File download successfully completed, or Reception of the last data packet containing content
Upload:
Start: File upload starts, or
Method A: Sending of the first data packet containing content
Method B: Reception of the [ACK] from the [SYN, ACK] for active mode connections, sending of the [ACK] for the [SYN, ACK] for passive mode connections on the data socket.

Stop: File upload successfully completed, or Reception of the [FIN, ACK] for the last data packet containing content.

2.8.7 FTP {Download|Upload} Data CapacityAbstract definition

FTP {Download|Upload} denotes the maximum FTP download or upload Bandwidth of data capacity.

2.8.7.2 Computation

2.8.7.2.1 Abstract Formula

$$\text{FTP \{Download | Upload\} Bandwidth [mbit/s]} = \sum_{1}^{n} DataRateOfEachTCPconnection$$

2.8.7.2.2 Trigger points

Start: multiple TCP/IP connections are established and download (or upload) one or multiple FTP files from (or to) one or multiple different FTP servers in parallel.

Stop: if all FTP files are downloaded (or uploaded) or a fixed time duration is reached.

2.8.7.2.3 Remarks

The test purpose is to measure the available download or upload Bandwidth of the network in case of PS data roaming. The FTP data contents and the establish time of the multiple TCP/IP connections are irrelevant.

2.9 WEB Browsing (HTTP / HTTPS) QoS Parameters

Preconditions: PS attached and the respective PDP context / default EPS bearer context activated.

The KPIs defined in this section are generic and are applicable to HTTP or HTTPS.

Assumption: a single TCP/IP connection.

2.9.1 HTTP IP-Service Access Success Ratio [%] [2]

2.9.1.1 Abstract definition

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

See ETSI TS 102 250-2 (6.8.3 HTTP IP-Service Access Failure Ratio)

2.9.1.2 Computation

Abstract Formula

2.9

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

Trigger points

2.9.1.2.2 Start: enter the URL and hit “Return” or First [SYN] sent.
Stop: Web page download starts, or
Method A: Reception of the first data packet containing content.
Method B: Sending of the first GET command.

2.9.2 HTTP IP-Service Setup Time [s] [2]Abstract definition

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

See ETSI TS 102 250-2 (6.8.4 HTTP IP-Service Access Time)

2.9.2.2 Computation

2.9.2.2.1

Abstract Formula

2.9.2.2.2

$$\text{HTTP IP-Service Setup Time [s]} = \left(t_{\text{IP-Service access successful}} - t_{\text{IP-Service access start}}\right)[s]$$

Trigger points

Start: enter the URL and hit “Return” or First [SYN] sent.
Stop: Web page download starts, or
Method A: Reception of the first data packet containing content.
Method B: Sending of the first GET command.

2.9.3 HTTP Session Success Ratio [%] [2]Abstract definition

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

See ETSI TS 102 250-2 (6.8.5 HTTP Session Success Failure Ratio)

2.9.3.2 Computation

Abstract Formula

2.9

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

Trigger points

2.9.3.2.2

Start: enter the URL and hit "Return" or first [SYN] sent.
Stop: The complete Web page appears in the browser window, or Reception of the last data packet containing content.

2.9.4 HTTP Session Time [s] [2]Abstract definition

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

See ETSI TS 102 250-2 (6.8.6 HTTP Session Time)

2.9.4.2 Computation

Abstract Formula

2.9.4.2.1

$$\text{HTTP Session Time [s]} = \left(t_{\text{session end}} - t_{\text{session start}}\right)[s]$$

2.9.4.2.2 **Trigger points**

Start: enter the URL and hit "Return" or first [SYN] sent.
Stop: The complete Web page appears in the browser window, or Reception of the last data packet containing content

2.9.5 HTTP Mean Data Rate [kbit/s] [2]Abstract definition

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

See ETSI TS 102 250-2 (6.8.7 HTTP Mean Data Rate)

2.9.5.2.1

2.9.5.2 Computation

Abstract Formula

2.9.5.2.2

$$\text{HTTP Mean Data Rate [kbit/s]} = \frac{\text{user data transferred [kbit]}}{\left(t_{\text{data transfer complete}} - t_{\text{data transfer start}}\right)[s]}$$

Trigger points

Start: Web page download starts or First [SYN] sent, or
Method A: Reception of the first data packet containing content.
Method B: Sending of the first GET command

Stop: Web page download successfully completed, or reception of the last data packet containing content

2.9.6 HTTP Data Transfer Success Ratio [%] [2]Abstract definition

The data transfer success ratio is the proportion of completed data transfers and data transfers that were started successfully.

See ETSI TS 102 250-2 (6.8.8 HTTP Data Transfer Cut-off Ratio)

2.9.6.2 Computation

Abstract Formula

2.9

$$\text{HTTP Data Transfer Success Ratio [\%]} = \frac{\text{completed data transfers}}{\text{successfully started data transfers}} \times 100$$

Trigger points

2.9.6.2.2 Start: Web page download starts or First [SYN] sent, or
Stop: Web page download successfully completed, or reception of the last data packet containing content.

2.9.7 HTTP Content Compression Ratio [%] [2]Abstract definition

The HTTP content compression ratio denotes the compression level of the received data accessible by the user agent in relation to the data sent by the origin server by using HTTP. It takes into account the overall effects of loss and lossless compression and non-reversible modifications of the original stored content during transmission.

See ETSI TS 102 250-2 (6.8.9 HTTP Content Compression Ratio)

2.9.7.2 Computation

2.9.7.2.1

Abstract Formula

$$\text{HTTP Content Compression Ratio [\%]} = \left(1 - \frac{\text{received HTTP content size}}{\text{sent HTTP content size}}\right) \times 100$$

2.9.7.2.2

Trigger points

Start: Web page download starts or First [SYN] sent, or
Method A: Reception of the first data packet containing content.
Method B: Sending of the first GET command
2.9.7.2.3 Stop: Web page download successfully completed, or reception of the last data packet containing content.

Remarks

Regarding the download of images the HTTP content compression ratio gives no indication on the quality of the compressed images as perceived by the user. The explanations on the influence of performance enhancement proxies should be taken into account. The current definition is applied to transferring HTTP content that consists of multiple objects (e.g. a web page) or content that consists of a single object. The “sent HTTP content” is an external input parameter for the calculation. It may be a constant (e.g. reference web page) or it may be measured directly at the HTTP server during the test execution, in a different network.

2.9.8 HTTP {Download|Upload} Data CapacityAbstract definition

HTTP {Download|Upload} denotes the maximum HTTP download or upload Bandwidth of data capacity.

2.9.8.2 Computation

Abstract Formula

2.9

$$\text{HTTP \{Download | Upload\} Bandwidth [mbit/s]} = \sum_{1}^{n} DataRateOfEachTCPconnection$$

Trigger points

2.9.8.2.2 Start: multiple TCP/IP connections are established and download (or upload) one or multiple HTTP files from (or to) one or multiple different HTTP servers in parallel.

Stop: if all HTTP files are downloaded (or uploaded) or a fixed duration is reached.

2.9.8.2.3 Remarks

The test purpose is to measure the available download or upload Bandwidth of the network in case of PS data roaming. The HTTP data contents and the establish time of the multiple TCP/IP connections are irrelevant.

2.10 PING QoS Parameters

2.10.1 PING Packet Loss Ratio

2.10.1.1 Abstract definition

PING packet Loss ratio is the proportion of dropped PING packets vs. the total PING packets sent.

See also the section: Service integrity - Packet Loss

2.10.1.2.1

2.10.1.2 Computation

Abstract Formula

2.10.1.2.2

$$\text{PING Packet Loss Ratio [\%]} = (1 - \frac{\text{Succssful PING packets received}}{\text{Total number of PING packets sent}}) \times 100$$

Trigger Points

Start: ICMP echo request sent,

Stop: ICMP echo reply received.

Remarks

An optional parameter is the number of PING packets sent.

2.10.2 PING Round Trip Time [ms] [2]Abstract definition

2.10.1.2.3

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 IP connection must already be established.

See ETSI TS 102 250-2 (6.3.1 PING roundtrip time)

2.10.2.2 Computation

Abstract Formula

2.1

$$\text{Ping Round Trip Time [ms]} = \left(t_{\text{packet received}} - t_{\text{packet sent}}\right)[\text{ms}]$$

Trigger Points

2.10.2.2.2 Start: ICMP echo request sent,

Stop: ICMP echo reply received by the sender.

2.10.2.2.3 Remarks

The size of the PING data packet is an optional parameter which will impact on the result of the measurement.

3 QoS Parameters for Interconnection

The QoS parameters (KPIs) specified in this section are mainly used for TDM Interconnection between a Mobile Operator and an International Carrier for International Transit Voice.

3.1 ASR = (Answer Seizure Ratio)

ITU E.425 [14] ASR represents the relationship between the number of seizures that result in an answer signal and the total number of seizures.

ASR is measured by the mobile operator on outgoing traffic on their gateway switch.

3.2 ABR = (Answer BID Ratio)

ITU E.425 [14] ABR represents the relationship between the number of bids that result in an answer signal and the total number of bids.

ABR is to be measured by the carrier on incoming traffic on their gateway switch.

3.3 CLI = (Calling Line Identification)

Calling Line Identification, also known as “A number” is basic information contained in the signalling system that identifies the calling party.

CLIP – CLI Presentation provides for the calling user number to be displayed to the called user.

CLIR – CLI Restriction provides a means for the calling user to restrict presentation of its MSISDN to the called user.

A more detailed definition is referred to in section 2.3.11.

3.4 NER = (Network Efficiency Ratio)

ITU E.425 [14] NER expresses the relationship between the number of seizures and the sum of the seizures resulting in an answer signal OR busy signal OR no answer. It excludes the effects of customer behaviour and terminal behaviour. It represents the ability of the network to deliver calls to the far end terminal.

A more detailed definition is referred to in sections 2.3.1 and 2.3.2.

3.5 PGAD = (Post Gateway Delay)

ITU E.437 [4] PGAD is the time interval between the seizure of the international circuit and the receipt of the answer supervision.

3.6 PDD = (Post Dial Delay)

ITU E.431 [13] PDD is the time interval between dialling completion and the call connection (ringing tone).

A more detailed definition is provided in sections 2.3.3 and 2.3.4.

3.7 ALOC = (Average Length of Conversation)

ITU E.437 [4] ALOC measures the average duration of calls. A statistically significant difference in ALOC to the same destination on different routes may be investigated.

A more detailed definition is referred to in section 2.3.10.

3.8 Speech Quality on Sample Basis.

Reference: ITU-T Recommendation P.862 [21] in conjunction with ITU-T Recommendation P.862.1 [29].

A more detailed definition is referred to in the section 2.3.12.

4 VoLTE / ViLTE Quality Parameters

This section contains VoLTE [31] and ViLTE [32] quality KPIs for interconnect and roaming [18] [30] [33]. The quality parameters (KPIs) are divided into seven groups:

- Accessibility (IMS Registration quality)
- Voice / video service integrity and retainability
- SRVCC (PS-CS) quality
- EPS bearer quality

- Media transport quality
- Diameter quality[10]

The first three groups contribute to the service quality over LTE. The last three groups characterize the quality of networks / transport.

4.1 VoLTE / ViLTE Service Quality Parameters

VoLTE / ViLTE service QoS parameters (KPIs) contains the service accessibility, integrity mobility and preservation.

4.1.1 VoLTE / ViLTE Accessibility Parameters

4.1.1.1 IMS Registration Success Ratio

The SIP Registration procedure in IR.92 [31] and IR.94 [32] shall be followed. See also 5.3.2 of ETSI TR 103 219 [34].

4.1.1.1.1 Abstract definition

This parameter denotes the probability that UE successfully registers to IMS.

$$\text{IMS Registration Success Ratio}\,[\%] = \frac{\sum \text{successful IMS Registration attempts}}{\sum \text{all IMS Registration attempts}} \times 100$$

4.1.1.2 IMS Registration Time

The SIP Registration procedure in IR.92 [31] and IR.94 [32] shall be followed. See ETSI TR 103 219 [34] section 5.3.3.

4.1.1.2.1 Abstract formula

$$\text{IMS Registration Time }[s] = \left(t_{\text{Network confirms IMS registration}} - t_{\text{UE requests IMS Registration}}\right)[s]$$

4.1.1.3 IMS Third-party Registration Success Ratio

4.1.1.3.1 Abstract definition

This parameter denotes the probability for the 3rd-party IMS registrations to be successfully performed.

$$\text{IMS 3rd-party registration success ratio}\,[\%] = \frac{\sum \text{successful IMS 3rd-party Registration attempts}}{\sum \text{all IMS 3rd-party Registration attempts}} \times 100$$

[10] The Diameter KPIs, denoting the quality of PCC and DRA quality, are not applied to e2e testing.

4.1.2 VoLTE / ViLTE Service Quality Parameters

The entire quality parameters in this section characterize the QoS aspect of the VoLTE / ViLTE service integrity.

For the simplification of the technical description in a roaming scenario, H-party denotes a party at HPMN, R-party denotes a Roaming party at the VPMN. In an interconnect scenario, A-party denotes a party at HPMN A and B-party denotes a party at HPMN B where PMN A and PMN B are interconnected.

For a valid KPI test, the following preconditions need to be met [34]:

- Default QCI5 EPS bearer context for the IMS well-known APN shall be established and the EPS ATTACH message shall contain "IMS PS Voice" in the voice domain preference list.
- IMS shall be present.
- LTE coverage shall be present.
- IMS registration shall be successfully accomplished.

4.1.2.1 VoLTE / ViLTE service access success ratio – NER MO (%)

4.1.2.1.1 Abstract definition

The KPI denotes the probability for the end-user to access the VoLTE / ViLTE service and initiate a voice or video call. See also 2.3.1.1.

4.1.2.1.2 Abstract formula

$$\text{VoLTE MO Accessibility [\%]} = \frac{\sum \text{Number of successful call attempts}}{\sum \text{Number of total call attempts}} \times 100$$

$$\text{ViLTE MO Accessibility [\%]} = \frac{\sum \text{Number of successful call attempts}}{\sum \text{Number of total call attempts}} \times 100$$

Trigger points from an end user’s viewpoint, see 2.3.1.2..

Technical description

Start test: SIP "INVITE" sent by the R- / A-party.

Successfully stop test: ”180 Ringing” is received by the R- / A-party.

Unsuccessfully stop test: either the R- / A-party receives a 4xx error message as response to the "INVITE", or timeout.

4.1.2.2 VoLTE / ViLTE service access success ratio – NER MT (%)

4.1.2.2.1 Abstract definition

The KPI denotes the probability that the end-user can access the VoLTE / ViLTE service and receive a VoLTE / ViLTE call. See also 2.3.2.

Abstract formula

$$\text{VoLTE MT Accessibility [\%]} = \frac{\sum \text{Number of successful call attempts}}{\sum \text{Number of total call attempts}} \times 100$$

4.1

$$\text{ViLTE MT Accessibility [\%]} = \frac{\sum \text{Number of successful call attempts}}{\sum \text{Number of total call attempts}} \times 100$$

Trigger points from an end user's viewpoint, see also 2.3.2.1.

Test start: Beginning of a call attempt of H-party / B-party, SIP "INVITE" received by the R-Party / A-party.

Successfully Test stop : ”180 Ringing” is sent by the R-party / A-party and ”180 Ringing” is received by H-party / B-party.

Unsuccessfully test stop : either the H-/ B-party receives a 4xx error message as response to the "INVITE", or timeout.

4.1.2.3 VoLTE / ViLTE session setup time – PDD-MO (s)

4.1.2.3.1 Abstract definition

Time needed to setup an MO VoLTE / ViLTE call. See also 2.3.3.1.

Note: The KPI VoLTE / ViLTE session setup in this definition is referred to a complete VoLTE / ViLTE session establishment, therefore is not applied to SRVCC when voice session is handed over legacy technologies during the voice session setup.

4.1.2.3.2 Abstract formula

$$\text{VoLTE MO Session Setup Time [s]} = \left(t_{\text{Calling party receives notification}} - t_{\text{Calling party initiates call session}}\right)[\text{s}]$$

$$\text{ViLTE MO Session Setup Time [s]} = \left(t_{\text{Calling party receives notification}} - t_{\text{Calling party initiates call session}}\right)[\text{s}]$$

Test start: R- / A- party initiates VoLTE session and sends SIP "INVITE"

Successful test stop : SIP: "200 OK (INVITE)" sent by H- / B-party and SIP: "200 OK (INVITE)" received by R- / A-party [34]. In manual test, the time for the user accepting the incoming call is excluded in the calculation.

Unsuccessful Test stop : R- / A-party receives a 4XX error message - the session set-up is cancelled or does not receive any notification within a pre-determined time.

4.1.2.4 VoLTE / ViLTE session setup time – PDD-MT (s)

4.1.2.4.1 Abstract definition

Time needed to setup an MT VoLTE / ViLTE call. See also 2.3.4.1.

Note: The KPI VoLTE / ViLTE session setup in this definition is referred to a complete VoLTE session establishment, therefore is not applied to SRVCC when voice session is handed over legacy technologies during the voice session setup.

4.1

Abstract formula

$$\text{VoLTE MT Session Setup Time [s]} = \left(t_{\text{Calling party receives notification}} - t_{\text{Calling party initiates call session}}\right)[\text{s}]$$

$$\text{ViLTE MT Session Setup Time [s]} = \left(t_{\text{Calling party receives notification}} - t_{\text{Calling party initiates call session}}\right)[\text{s}]$$

Test start: H- / B-party initiates VoLTE / ViLTE session and sends SIP "INVITE" .

Successful test stop : SIP: "200 OK (INVITE)" sent by R- / A-party and "200 OK (INVITE)" received by H- / B-party [34]. In manual test, the time for the user accepting the incoming call is excluded in the calculation.

Unsuccessful test stop : H- / B-party receives a 4XX error message - the session set-up is cancelled or does not receive any notification within a pre-determined time.

4.1.2.5 VoLTE / ViLTE session setup success ratio – CSSR-MO (%)

4.1.2.5.1 Abstract definition

VoLTE / ViLTE session setup success ratio denotes the probability that a successful MO VoLTE / ViLTE call attempt results in a successfully established MO VoLTE / ViLTE call. See also 3.2.5.

4.1.2.5.2 Abstract formula

$$\text{VoLTE MO session setup ratio [\%]} = \frac{\sum \text{successful MO VoLTE calls}}{\sum \text{all successful MO VoLTE call attempts}} \times 100$$

$$\text{ViLTE MO session setup ratio [\%]} = \frac{\sum \text{successful MO VoLTE calls}}{\sum \text{all successful MO VoLTE call attempts}} \times 100$$

Test start: R- / A-party initiates VoLTE / ViLTE session and sends SIP "INVITE"

Successful test stop : SIP "200 OK (INVITE)" sent by H- / B-party and SIP "200 OK (INVITE)" received by R- / A-party [34]. In manual test, the time for the user accepting the incoming call is excluded in the calculation.

Unsuccessful test stop : R- / A-party receives a 4XX error message - the session set-up is cancelled or does not receive any notification within a pre-determined time. Any intentional call terminations at R- / A-party or H- / B-party are excluded.

4.1.2.6 VoLTE / ViLTE session setup success ratio – CSSR-MT (%)

4.1.2.6.1 Abstract definition

VoLTE session setup success ratio denotes the probability that a successful MT VoLTE / ViLTE call attempt results in a successfully established MT VoLTE / ViLTE call. See also 3.2.6.

4.1 Abstract formula

$$\text{VoLTE MT session setup ratio [\%]} = \frac{\sum \text{successful MT VoLTE calls}}{\sum \text{all successful MT VoLTE call attempts}} \times 100$$

$$\text{ViLTE MT session setup ratio [\%]} = \frac{\sum \text{successful MT VoLTE calls}}{\sum \text{all successful MT VoLTE call attempts}} \times 100$$

Test start: H- / B-party initiates VoLTE / ViLTE session and sends SIP "INVITE" .

Successful test stop : SIP: "200 OK (INVITE)" sent by R- / A-party and "200 OK (INVITE)" received by H- / B-party [34]. In manual test, the time for the user accepting the incoming call is excluded in the calculation.

Unsuccessful test stop: H- / B-party receives a 4XX error message that the session set-up is cancelled or does not receive any notification within a pre-determined time. Any intentional call terminations at R- / A-party or H- / B-party are excluded.

4.1.2.7 VoLTE / ViLTE session duration – ALOC-MO (s)

4.1.2.7.1 Abstract definition

MO VoLTE / ViLTE call duration, see also 2.3.10.

4.1.2.7.2 Abstract formula

$$\text{VoLTE MO session duration [s]} = \left(t_{\text{BYE}} - t_{\text{200 OK (INVITE)}}\right)\text{[s]}$$

$$\text{ViLTE MO session duration [s]} = \left(t_{\text{BYE}} - t_{\text{200 OK (INVITE)}}\right)\text{[s]}$$

An MO VoLTE / ViLTE session duration is the time measured at R- / A-party between receiving "200 OK (INVITE)" and sending “BYE”. The call release is initiated by R- / A-party.

4.1.2.8 VoLTE / ViLTE session duration – ALOC-MT (s)

4.1.2.8.1 Abstract definition

MT VoLTE call duration, see also 2.3.10.

Abstract formula

$$\text{VoLTE MT session duration [s]} = \left(t_{\text{BYE}} - t_{\text{200 OK (INVITE)}}\right)\text{[s]}$$

4.1

$$\text{ViLTE MT session duration [s]} = \left(t_{\text{BYE}} - t_{\text{200 OK (INVITE)}}\right)\text{[s]}$$

An MT VoLTE / ViLTE session duration is the time measured at R- / A-party between sending "200 OK (INVITE)" and receiving “BYE”. The call release is initiated by H- / B-party.

4.1.2.9 OIP transparency – CLI transparency

4.1.2.9.1 Abstract definition

VoLTE call originating identification presentation (OIP) needs to be delivered and presented correctly and complete. It can be used to call back to the original called party. See also 2.3.11.

Abstract formula

4.1

$$\text{OIP transparency ratio [\%]} = \frac{\sum \text{Number of complete and correct OIPs}}{\sum \text{Number of successful VoLTE calls}} \times 100$$

4.1.2.10 VoLTE speech quality (SpQ MOS-LQO)

The VoLTE Speech Quality represents the end-to-end speech quality of the VoLTE service.

ETSI TS 102 250-2 [2] and defines two variants of a "Telephony Speech Quality" KPI. Both are based on the same single measurement. A single measurement consist of the MOS-LQO value determined for a single transferred speech sample, according to ITU-T P.863 POLQA [27]. A speech call usually consists of multiple speech samples transferred in both directions.

The first variant, called "Telephony Speech Quality on Call Basis" (ETSI TS 102 250-2 [2] clause 6.6.3) does a pre-aggregation of the measured MOS-LQO values to *one value per call* (in either direction). ETSI TS 102 250-2 does not specify the aggregation method for this pre-aggregation. A suitable method can be found in ETSI TR 102 506 [28].

The second variant, called "Telephony Speech Quality on Sample Basis" (ETSI TS 102 250-2 clause 6.6.4) does *not* pre-aggregate the measured MOS-LQO values. In other words, one call will result in *multiple* values for this KPI.

Averaging the results obtained in a measurement campaign with repeated calls will in general yield different statistics for the two variants. Due to its pre-aggregation, the "Telephony Speech Quality on Call Basis" focusses more on how users judge entire calls, while the "Telephony Speech Quality on Sample Basis" gives insight into the overall speech transmission quality for a particular connection or interconnection.

4.1.2.10.1 Abstract definition

VoLTE speech quality in roaming is measured on Call Basis, See also 2.3.12.

Abstract formula

$$\boxed{\begin{array}{l}\text{VoLTE Speech Quality on Call Basis (received R - party)} = f\left(\text{MOS - LQO}\right)\\ \text{VoLTE Speech Quality on Call Basis (received H - party)} = f\left(\text{MOS - LQO}\right)\end{array}}$$

4.1

The validation of the end-to-end quality is made using MOS-LQO scales [2]. These scales describe the opinion of users with speech transmission and its troubles (noise, robot voice, echo, dropouts, time scaling introduced by the jitter buffer, etc.) according to ITU-T P.863 [27]. The scale used has to be reported. An aggregation for measurement campaigns in VoLTE roaming is made on call basis (the 1st variant).

The KPI shall be tested at the call terminating side and at the call originating side. The two streams shall be successfully tested at the same time.

4.1.2.11 VoLTE speech quality (SpQ R-Factor-LQ)

4.1.2.11.1 Abstract definition

The R-Factor is an *estimated* speech quality rating defined by the E-Model (ITU-T G.107.1 [35]). As an alternative method of assessing call quality, R-Factor is scaling from 0 to 120. It can also be *calculated* from MOS-LQO value according to ITU-T P.863 – POLQA [27].

4.1.2.11.2 Abstract formula

$$\boxed{\begin{array}{l}\text{VoLTE Speech Quality R - Factor (received R - party)} = f\left(\text{G.107 - LQ}\right)\\ \text{VoLTE Speech Quality R - Factor (received H - party)} = f\left(\text{G.107 - LQ}\right)\end{array}}$$

The KPI shall be tested at the call terminating side and at the call originating side. The two streams shall be successfully tested at the same time. The table below shows a proposed mapping range between the user’s satisfaction, MOS and R-Factor scales.

User satisfaction level	MOS scale	R-Factor scale
Very satisfied	**4.3 - 5.0**	**90 - 100**
Satisfied	**4.0 - 4.3**	**80 - 90**
Some users satisfied / dissatisfied	**3.6 - 4.0**	**70 - 80**
Many users dissatisfied	**3.1 - 3.6**	**60 - 70**
Nearly all users dissatisfied	**2.6 - 3.1**	**50 - 60**
Not recommended	**1.0 - 2.6**	**< 50**

Table 3: Proposed mapping MOS and R-Factor scales

4.1.2.12 ViLTE Audio Quality (SpQ MOS-LQO) on sample basis

4.1.2.12.1 Abstract definition

The KPI of ViLTE audio speech quality on sample basis is an indicator representing the quantification of the end-to-end voice transmission quality as perceived by the user. This parameter computes the speech quality on a sample basis. See also 6.7.9 of [2].

4.1.2.12.2 Abstract formula

$$\text{ViLTE speech quality on sample basis (received R - party)} = f(\text{MOS - LQO})$$
$$\text{ViLTE speech quality on sample basis (received H - party)} = f(\text{MOS - LQO})$$

The KPI shall be tested at the call terminating side and at the call originating side. The two audio streams shall be successfully tested at the same time together with a successful test on the two video streams.

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).

4.1.2.13 ViLTE Video quality (MOS)

ViLTE video quality KPI is characterized by the Mean Opinion Score according to PEVQ MOS [41].

4.1.2.13.1 Abstract definition

End-to-end quality of the video signal as perceived by the end user during a ViLTE session. This parameter computes the video quality on a sample basis. See also 6.7.10 of [2].

4.1.2.13.2 Abstract formula

$$\text{ViLTE video quality on sample basis (received R - party)} = f(\text{PEVQ - MOS})$$
$$\text{ViLTE video quality on sample basis (received H - party)} = f(\text{PEVQ - MOS})$$

The KPI shall be tested at the video call terminating side and at the video call originating side. The two video streams shall be successfully tested at the same time together with a successful test on the two audio streams.

4.1.3 SRVCC (PS-CS) Quality Parameters [34]

SRVCC is the quality of voice mobility between 4G and the legacy technologies. For a valid KPI calculation the following preconditions need to be met:

- EPS bearer should be established and the EPS ATTACH message should contain "IMS PS Voice" as well as "CS Voice" in the voice domain preference list.

SRVCC can have different variants depending on the supporting mid-call, alerting or pre-alerting state of a VoLTE call when the call is handed over to UMTS or GSM.

4.1.3.1 SRVCC Success Ratio [34]

4.1.3.1.1 Abstract Definition

The KPI is the probability that UE successfully handover a VoLTE call to UMTS or GSM.

4.1.3.2 Abstract Formula

$$\text{SRVCC SuccessRatio}\,[\%] = \frac{\sum \text{successful SRVCC handovers}}{\sum \text{all SRVCC invocations}} \times 100$$

4.1.3.3 SRVCC Time [34]

For a valid KPI calculation the following preconditions need to be met:

- EPS bearer should be established and the EPS ATTACH message should contain "IMS PS Voice" as well as "CS Voice" in the voice domain preference list.
- An SRVCC handover to UMTS or GSM is successful.

4.1.3.3.1 Abstract Definition

The KPI specifies the time taken to successfully handover a VoLTE call to UMTS or GSM.

4.1.3.3.2 Abstract Formula

$$\text{SRVCC Time}\,[s] = \left(t_{\text{SRVCC handover successfuly completed}} - t_{\text{SRVCC handover invoked}}\right)[s]$$

4.2 VoLTE / ViLTE Networks & Transport QoS Parameters

The VoLTE / ViLTE networks and transport KPIs contains EPS bearer quality, RTP transport quality and DIAMETER KPIs.

4.2.1 VoLTE / ViLTE EPS Bearer Quality Parameters

This section contains the EPS bearer KPIs for SIP signalling, voice and video media in the VoLTE / ViLTE context.

4.2.1.1 Default EPS Bearer Context Activation Success Ratio

Default EPS bearer context is established and connected to the IMS well-known APN. If a VoLTE / ViLTE capable UE is involved in the KPI test, then a single or dual default EPS bearer contexts can be established in the EPS attach procedure. Only the default bearer context for the IMS well-known APN connection is concerned for this KPI.

Abstract formula

4.2

$$\text{QCI 5 Default EPS Bearer Context Activation SuccessRatio}\,[\%] = \frac{(\text{To IMS well-known APN}) \sum \text{PDN connection establishment successes}}{(\text{To IMS well-known APN}) \sum \text{PDN connection initiations}} \times 100$$

See section 2.2.10 for more details.

4.2.1.2 Default EPS Bearer Context Activation Time

Default EPS bearer context is established and connected to the IMS well-known APN.

If a VoLTE / ViLTE capable UE is involved in the KPI test; a single or dual default EPS bearer contexts can be established in the EPS attach procedure. Only the default bearer context for the IMS well-known APN connection is concerned for this KPI. See also 2.2.11.

4.2.1.2.1 Abstract formula

$$\text{QCI 5 Default EPS Bearer Context Activation Time}\,[s] = \left(t_{\text{IMS PDN connection establishment}} - t_{\text{IMS PDN connection initiation}}\right)[s]$$

The default QCI 5 EPS bearer context is used in AM for SIP Signalling and SMSoIP.

4.2.1.3 Default EPS Bearer Downlink GBR

The KPI indicates the Guaranteed Bit Rate by the network in Downlink; the value taken from EPS Quality of Service information element assigned by EPC [46].

4.2.1.4 Default EPS Bearer Uplink GBR

The KPI indicates the Guaranteed Bit Rate by the network in Uplink; the value taken from EPS Quality of Service information element assigned by EPC [46].

4.2.1.5 Default EPS Bearer QCI

The KPI indicates the Quality of Service Class Identifier (QCI) for the default EPS bearer; the value taken from EPS Quality of Service information element assigned by EPC [46]. QCI=5 is expected.

4.2.1.6 Dedicated EPS Bearer Context Activation Success Ratio

Dedicated EPS bearer context is established to transport the IMS-based voice media. The dedicated EPS bearer context for VoLTE is a Persistent EPS bearer context. See also 2.2.12.

Abstract formula

4.2

$$\text{Dedicated EPS Bearer Context Activation successRatio}\,[\%] = \frac{\sum \text{dedicated EPS bearer activation successes}}{\sum \text{dedicated EPS bearer activation initiations}} \times 100$$

The dedicated QCI1 GBR bearer is used in UM for the conversational voice service of VoLTE and ViLTE audio.

The dedicated QCI2 GBR bearer or QCI 8/9 non-GBR bearer is used in AM for the ViLTE conversational video service.

4.2.1.7 Dedicated EPS Bearer Context Activation Time

Dedicated EPS bearer context is established to transport the IMS-based voice media. The dedicated EPS bearer context for VoLTE is a Persistent EPS bearer context. See also 2.2.13.

Abstract formula

4.2.1.7.1

$$\text{Dedicated EPS Bearer Context Activation Time [s]} = \left(t_{\text{DedicatedEPSBearercontextactivationsuccess}} - t_{\text{DedicatedEPSBearercontextactivationinitiation}}\right)[\text{s}]$$

The dedicated QCI1 GBR bearer is used in UM for the conversational voice service of VoLTE and ViLTE audio.

The dedicated QCI2 GBR bearer or QCI 8/9 non-GBR bearer is used in AM for the ViLTE conversational video service.

4.2.1.8 Dedicated EPS Bearer Downlink GBR

The KPI indicates the Guaranteed Bit Rate by the network in Downlink; the value taken from EPS Quality of Service information element assigned by EPC [46].

4.2.1.9 Dedicated EPS Bearer Uplink GBR

The KPI indicates the Guaranteed Bit Rate by the network in Uplink; the value taken from EPS Quality of Service information element assigned by EPC [46].

4.2.1.10 Dedicated EPS Bearer QCI

The KPI indicates the Quality of Service Class Identifier (QCI) for the Dedicated EPS bearer; he value taken from EPS Quality of Service information element assigned by EPC [46].

4.2.1.11.1

4.2.1.11 IP data volume Rx

Abstract definition

IP data volume Rx denotes the total volume (in Kbytes or Mbytes) of IP data received at Rx on an EPS bearer. The IP header is included in the calculation.

Abstract formula

$$\text{IP data volume received} = \sum_{1}^{Nr} \text{IP\_PacketLength}\,[\text{KB}]$$

Nr is the total number of RTP packets received at Rx.

If QCI5, the KPI measures the IP data volume of SIP signalling and SMSoIP within a determined period.

If QCI1, the KPI measures the voice IP data volume within a VoLTE or ViLTE call duration.

If QCI2 or QCI 8/9, the KPI measures the video IP data volume within a ViLTE call duration.

4.2.1.12 IP data volume Tx

4.2.1.12.1 Abstract definition

IP data volume Tx denotes the total volume (in KB or MB) of IP data transmitted from Tx on an EPS bearer. The IP header is included in the calculation.

4.2.1.12.2 Abstract formula

$$\text{IP data volume transmitted} = \sum_{1}^{\text{Nt}} \text{IP\_PacketLength}\,[\text{KB}]$$

Nt is the total number of RTP packets transmitted from Tx.

If QCI5, the KPI measures the IP data volume of SIP signalling and SMSoIP within a determined period.

If QCI1, the KPI measures the voice IP data volume within a VoLTE or ViLTE call duration.

If QCI2 or QCI 8/9, the KPI measures the video IP data volume within a ViLTE call duration.

4.2.2 VoLTE / ViLTE Media Transport Quality Parameters

The VoLTE media uses symmetric RTP over UDP to transport voice [42]. The RTP quality is the key factor for the media transport. Because of the symmetry of the RTP protocol [43] the RTP KPIs in this section are relevant for the receiver Rx and transmitter Tx of the both ends (call originator and terminator).

The ViLTE media uses two separate RTP connections to transport audio and video streams. The audio and video RTP quality KPI are separately measured when two RTP connections are present during the test.

The media transport quality KPIs defined in this section are end-to-end quality parameters. It is assumed that for a typical VoLTE / ViLTE call between 2 UEs (particularly in an active test between two probes), the RTP/RTCP is sent end-to-end between the UEs. The RTP packets can traverse a bunch of RTP-“translators”, e.g. IMS-AGW and TrGW, if network-to-network interface (NNI) is encountered. The translators do not change the RTP packet payload contents apart from the Network Address and port translation (NAPT) function (i.e. only IP header and probably also UDP header are changed).

4.2.2.1 RTP Max & Mean Packet Delay Variation Rx

The RTP Packet Delay Variation is measured at the e2e Tx/Rx pair on the incoming stream at Rx [44].

4.2.2.1.1 Abstract definition

An RTP packet transit time D is the transfer time of a voice / video RTP stream from Tx to Rx individually identifiable IP packet, observed at Tx and Rx. D = (R – T) where T is the RTP timestamp of the packet at Tx, and R is the arrival time of the packet at Rx in the voice / video RTP stream (of interest within a measurement time interval).
D is also called RTP packet one-way-delay, which is relying on the clock (Time-Stamp) synchronization between Tx and Rx – a kind of two points time measurement.

The mean RTP packet one-way-delay (OWD) is the average of delay of a voice / video RTP stream from Tx to Rx. The test accuracy of OWD relies on the clock synchronization between Tx and Rx.

Taking the minimum transit time in the voice / video RTP stream as the reference, a packet delay variation PDVi is the difference between the transit time and the reference in the voice / video RTP stream . PDVi = Di - Dmin [45], where Di = (Ri – Ti) is the ith packet transit time, Ti is the RTP timestamp of packet i at Tx, and Ri is the arrival time of the packet i at Rx in the voice / video RTP stream within a measurement time interval.

Dmin is the transit time of the packet with the lowest value for delay (minimum) over the current test interval. Values of PDV may be zero or positive, and quantiles of the PDV distribution are direct indications of delay variation. PDV is a version of the one-way-delay distribution, shifted to the origin by normalizing to the minimum delay.

An RTP max PDV is the difference between the maximum transit time and the minimum transit time in the voice / video RTP stream within a measurement time interval, measured at Rx.

An RTP mean PDV is the difference between the average transit time and the minimum transit time in the voice / video RTP stream within a measurement time interval, measured at Rx.

4.2.2.1.2 Abstract formula

$$\boxed{\text{RTP maxPDV [ms]} = (D_{\max} - D_{\min})\,\text{[ms]}}$$

$$\boxed{\text{RTP mean PDV [ms]} = \left(\frac{\sum_i D_i}{\text{nr}} - D_{\min}\right)\text{[ms]}}$$

$$\boxed{\text{RTP One Way Delay [ms]} = \frac{\sum_i D_i}{\text{nr}} - (t\_offset\_Rx - t\_offset\_Tx)\,\text{[ms]}}$$

Where n_r is the number of received packets in the voice / video RTP stream of interest; D_{min} is the minimum delay of RTP packets in transit time in the voice / video RTP stream of interest within a measurement time interval.

4.2.2.1.3 Assumption

The calculation of the KPI RTP average PDV assumes that the clock offset to the universal time at Tx (t_{offset_Tx}) and at Rx (t_{offset_Rx}) are stable within the KPI measurement time interval.

4.2.2.2 RTP means interarrival jitter of incoming stream Rx

The RTP mean inter-arrival jitter is measured at the e2e Tx/Rx pair on the incoming voice / video RTP streams at Rx.

4.2.2.2.1 Abstract definition

The interarrival jitter J is the mean deviation (smoothed absolute value) of the difference in transit time of two consecutive RTP packets from Tx to Rx in the voice / video RTP stream of interest as defined in clause 6.4.1 in [42].

4.2.2.2.2 Abstract formula

$$\mathrm{J}_{\mathrm{i+1}}\ [\mathrm{ms}] = \mathrm{J}_{\mathrm{i}} + \frac{|D_{\mathrm{i,i\text{-}1}}| - \mathrm{J}_{\mathrm{i}}}{16}\ [\mathrm{ms}]$$

Where the i[th] difference $D_{i,i-1}$ in transit time of two consecutive RTP packets $D_{i,i-1} = (R_i - T_i) - (R_{i-1} - T_{i-1})$, T_i is the RTP timestamp of packet i at Tx, and R_i is the arrival time of packet i at Rx in the voice / video RTP stream of interest within a measurement time interval.

4.2.2.2.3 Assumption

The calculation of the KPI RTP mean that interarrival jitter assumes that the clock offset to the universal time at Tx (t_{offset_Tx}) and at Rx (t_{offset_Rx}) are stable within the KPI measurement time interval.

4.2.2.3 RTP Mean Data Rate Tx / Rx

4.2.2.3.1 Abstract definition

This parameter describes an average data transfer rate measured over the entire voice / video call at Tx / Rx. The voice / video call shall be successfully terminated. The prerequisite for this parameter is LTE network and IP / UDP service access.

Note: The KPI is measured at RTP level (RTP header + voice / video payload). IP and UDP headers are not taken into account.

4.2.2.3.2 Abstract formula

$$\mathrm{RTP\{Tx|Rx\}\ Data\ Rate[kbit/s]} = \frac{\sum \mathrm{RTP\ data\ transferred[kbit]}}{\left(t_{\mathrm{packet\ data\ transfer\ complete}} - t_{\mathrm{packet\ data\ transfer\ start}}\right)[\mathrm{s}]}$$

4.2.2.4 RTP packets lost Rx

4.2.2.4.1 Abstract definition

The KPI is defined as the number of lost packets at Rx side (downlink) in the RTP stream of interest within the voice / video call duration. This value is equivalent to the number of missing RTP sequence numbers at Rx.

Abstract formula

$$RTP\_NumPacketsLost = \sum lostPackets$$

4.2.2.4.2

4.2.2.5 RTP packet loss ratio Rx

Abstract definition

The KPI denotes the probability that an RTP packet is lost and not received at Rx.

4.2.2.5.1 **Abstract formula**

4.2

$$RTP\_PacketLossRatio[\%] = \frac{\sum RTP\_NumPacketsLost}{\sum RTP\_NumPacketsReceived + \sum RTP\_NumPacketsLost} \times 100$$

RTP_NumPacketsReceived is the total number of RTP packets received at Rx in the RTP stream of interest within the VoLTE / ViLTE call duration.

4.2.2.6 RTP round trip delay (RTD)

Abstract definition

4.2.2.6.1

An e2e (endpoint A and B) RTP round-trip-delay (A-B-A) is the sum of the packet transit time D_{a2b} and the packet transit time D_{b2a} of the same RTP packet looped-back. The delay time to process and loop the packet back at B side is not counted in the RTD.

4.2.2.6.2 **Abstract formula**

$$RTPmeanRTD_{a2b2a}\,[\mathrm{ms}] = \frac{\sum D_{ia2b} + \sum D_{ib2a}}{N_{ra}}\quad[\mathrm{ms}]$$

Where RTD_{a2b2a} denotes a packet transit time on a round trip path from endpoint A to B and from B loopback to A. D_{ia2b} is the transit time of the i^{th} packet from Tx at endpoint A to Rx of endpoint B. D_{ib2a} is the transit time of the same i^{th} packet looped back from Tx at endpoint B to Rx of endpoint A. N_{ra} is the total number of looped back packets received at Rx of A side.

4.2.2.6.3 Note: Lost RTP packets at looping back are not counted in the calculation.

Assumption

The calculation of the KPI RTP means that RTD assumes the clock offset to the universal time at Tx (t_{offset_Tx}) and at Rx (t_{offset_Rx}) of the A and B endpoints are stable within the KPI measurement time interval.

4.2.3 DIAMETER Quality Parameters

4.2.3.1 Policy & Charging Control

The Diameter PCC KPIs are measured over Rx interface for the quality of Diameter signaling flows [38] between P-CSCFs and PCRFs. The Rx messages consist of four pairs of Diameter Request – Answer commands which contribute to a number of PCC procedures for the initial provision, modification, termination of a Diameter Rx session and session binding (a Diameter Rx session associated with a VoLTE IMS session) [38].

4.2.3.1.1 AA-Request (AAR) Success Ratio

4.2.3.1.1.1 Abstract Definition

The KPI is the probability that an AA-Request command (AAR) sent from P-CSCF via Rx interface to PCRF results in a successful AA-Answer command (AAA) from PCRF received by P-CSCF.

4.2.3.1.1.2 Abstract Formula

$$\text{AAR SuccessRatio}[\%]=\frac{\sum \text{Number of AAA with Result-Code as Diameter\_Success}}{\sum \text{Number of AAR}}\times 100$$

4.2.3.1.2 RA-Request (RAR) Success Ratio

4.2.3.1.2.1 Abstract Definition

The KPI is the probability that a Re-Auth-Request command (RAR) sent from PCRF via Rx interface to P-CSCF results in a successful RA-Answer command (RAA) from P-CSCF received by PCRF.

4.2.3.1.2.2 Abstract Formula

$$\text{RAR SuccessRatio}[\%]=\frac{\sum \text{Number of RAA with Result-Code as Diameter\_Success}}{\sum \text{Number of RAR}}\times 100$$

4.2.3.1.3 ST-Request (STR) Success Ratio

4.2.3.1.3.1 Abstract Definition

The KPI is the probability that a Session-Termination-Request command (STR) sent from P-CSCF via Rx interface to PCRF results in a successful ST-Answer command (STA) from PCRF received by P-CSCF.

4.2.3.1.3.2 Abstract Formula

$$\text{STR SuccessRatio}[\%]=\frac{\sum \text{Number of STA with Result-Code as Diameter\_Success}}{\sum \text{Number of STR}}\times 100$$

4.2.3.1.4 AS-Request (ASR) Success Ratio

4.2.3.1.4.1 Abstract Definition

The KPI is the probability that an Abort-Session-Request command (ASR) sent from PCRF via Rx interface to P-CSCF results in a successful AS-Answer command (ASA) from P-CSCF received by PCRF.

4.2.3.1.4.2 Abstract Formula

$$\text{RAR SuccessRatio}[\%] = \frac{\sum \text{Number of RAA with Result-Code as Diameter\_Success}}{\sum \text{Number of RAR}} \times 100$$

4.2.3.2 DIAMETER Routing Agent

The DRA KPI is to measure the DRA routing quality, in order to reach PCRF or HSS (routing to and redirect / relayed from), when more than one PCRF or HSS has been deployed in a Diameter realm.

4.2.3.2.1 Success Ratio of IMS Diameter Messages Routing

The KPI tests the success ratio of all Diameter sessions established over the Cx [39], Sh [40], Zh interfaces that reach the corresponding HSS when multiple and separately addressable HSSs have been deployed in a Diameter realm.

- Redirect / route the diameter messages from the HSS over the corresponding reference points to Cx, Sh, Zh to the corresponding IMS node I/S-CSCF, VoLTE / ViLTE Application Server, IP Short Message Gateway and Bootstrap Service Function.

4.2.3.2.1.1 Abstract definition

The KPI denotes the probability that Diameter messages (requests and responses) transmitted over Cx- ,Sh- , and Zh-interfaces are successfully routed or redirected / relayed by the DRA. The service failure messages initiated by the DRA over the Cx, Sh, and Zh interfaces are excluded in the DRA forwarding failures.

4.2.3.2.1.2 Abstract formula

$$\text{IMS Diameter Routed Messages SuccessRatio}[\%] = \frac{\sum \text{Requests and Responses successfully transmitted over Cx, Sh, Zh}}{\sum \text{Total Requests and Responses received by DRA} \mid \text{Cx, Sh, Zh}} \times 100$$

4.2.3.2.2 Success Ratio of PCC Diameter Messages Routing

This KPI measures the success ratio of all Diameter sessions established over the Gx [37], Rx [38] reference points for a certain IMS session.

- All diameter messages reach the corresponding PCRF when multiple and separately addressable PCRFs have been deployed in a Diameter realm
- Redirect / route the diameter messages from the PCRF over the corresponding reference point Gx or Rx to the corresponding IMS node PCEF/PDN-GW or P-CSCF/IMS-AGW respectively.

4.2.3.2.2.1 Abstract definition

The KPI denotes the probability that of Diameter messages (requests and responses) transmitted over Gx- and Rx-interface are successfully routed or redirected / routed by the

DRA. The service failure messages initiated by the DRA over the Gx and Rx interfaces are excluded in the DRA forwarding failures.

4.2.3.2.2.2 Abstract formula

$$\text{PCC Diameter Messages Routed Success Ratio}\,[\%] = \frac{\sum \text{Requests and Responses successfully transmitted} \mid \text{Gx, Rx}}{\sum \text{Total Requests and Responses received by DRA} \mid \text{Gx, Rx}} \times 100$$

5 Single Service Indicator and Single Quality Indicator

5.1 Single Service Indicator

Single service indicator (SSI) is a unique quality performance indicator which composes a set of selected KPIs of a particular service. Each KPI in the set has a weight according to the relevance contributing to the single service indicator.

5.1.1 Purpose of SSI

The main purpose of using SSI is to provide a single unique QoS score for the executive management, instead of many individual KPIs.

5.1.2 SSI Calculation

Calculation of an SSI consists of two steps, KPI normalisation and weighting.

5.1.2.1 KPI normalisation

Depending on the KPI definition, different KPIs can have different units, for example, percentage (%) or data rate kbit/s, etc. KPI normalisation maps the KPI measured value into a range between 0 and 100, by using two scoring limits, a higher limit and a lower limit for each KPI (higher limit > lower limit). After the KPI value normalisation, the higher the KPI score is, the better the quality is implied.

Despite of divergence of KPI definitions, there are two types of KPIs which have different calculations before the normalisation.

- Higher is better, for example, success ratio, data rate
- Lower is better, for example, loss ratio, latency delay or session establish time

For scoring "**higher is better**":

- KPI value less than the lower limit is interpreted as 0
- KPI value greater than the higher limit is interpreted as 100
- KPI value between the two limits is linear interpolated using formula:

$$\text{normalised value} = \frac{\text{KPI value - lower limit}}{\text{higher limit - lower limit}} \times 100$$

For scoring "**lower is better**":

- KPI value greater than the higher limit is interpreted as 0
- KPI value less than the lower limit is interpreted as 100
- KPI value between the two limits is linear interpolated using formula:

$$\text{normalised value} = \frac{\text{higher limit - KPI value}}{\text{higher limit - lower limit}} \times 100$$

5.1.2.2 KPI weighting

The weight of a KPI defines how important the role of this particular KPI is in the contribution of the SSI calculation. Assigning a bigger or heavier weight to a KPI will give the KPI more influence on SSI. Assigning smaller or lighter weights to KPIs will make those KPIs not so important. Negative weights are not allowed in the SSI calculation, but using zero weight can temporarily disable the KPI without removing it from the SSI definition. There has been at least one non-zero weighted KPI in the SSI definition.

5.1.2.3 SSI calculation formula

The following formula is used for the SSI calculation:

$$SSI = \frac{\sum_{1}^{n} \text{normalised KPIi} \times \text{weighti}}{\sum_{1}^{n} \text{weighti (all weights)}}$$

where "normalised KPI_i" is a normalised value of the i^{th} particular KPI_i and "$weight_i$" is the value of this KPI_i's "scoring weight". A total n KPIs are included in the SSI calculation.

Some examples helps for better understanding:

- If all KPIs contained in a SSI have the same weight, the value of SSI will be simply arithmetic average of all KPIs. It means all KPIs play the same role in the calculation of the SSI.
- If the weight of a KPI is set to 0, this particular KPI is ignored or masked at the SSI calculation.
- An SSI with two KPIs weighted with 1 and 3, will be calculated using 25% of the first KPI's value and 75% of the second KPI.

5.2 Single Quality Indicator

Similar to SSI combining the weighted KPI and resulting in a single quality score value for a particular service, single quality indicator (SQI) composes weighted SSIs of different services and provides a unique single score value for the quality.

5.2.1 SQI calculation

A SQI value can be calculated by using the following formula

$$SQI = \frac{\sum_{1}^{m} \text{SSIj} \times \text{weightj}}{\sum_{1}^{m} \text{weightj (all weights)}}$$

where "SSI_j" is a value of the j^{th} particular SSI_j and "weigh t_j" is the value of this SSI_j's "scoring weight". A total of m services, i.e. SSIs, are included in the SQI calculation.

Annex A Examples for measuring trigger points

A.1 SMS-Service:

A.1.1 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 Document Management

B.1 Document History

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
0.1.0	5/1/2001	First draft of document for IREG QoS WP discussion		
0.2.0	30/3/2001	Second draft of document for IREG QoS WP e-mail discussion		
0.3.0	27/4/2001	Third draft for IREG QoS workshop including Data Services		
1.0.0	10/6/2001	First stable Version for chapters Telephony, SMS		
2.0.0	11/9/2001	Document for approval at IREG #41 and GSMA with a document classification of “Unrestricted – Public”		
3.0.0	21/9/2001	Document approved by GSMA		
3.0.1	30/4/2002	Parameter Completion Rate SMS circuit switched added		
3.0.2	14/5/2002	Parameter for CSD and PSD Data Services added		
3.1.0	17/6/2002	Document for approval at IREG #43 and GSMA with a document classification of "Unrestricted – Public"		
3.2.0	10/10/2002	Changes in chapters 2.4, 2.5-2.7 approved by QoSWP Meeting #8, 02.10.2002		
3.2.1	10/2/2003	Editorial changes in chapters 2.5-2.7		
3.3	16/4/2007	IREG doc 52_037 incorporated “Addition of the new QoS parameter definitions”		
3.4	10/8/2009	Signal doc 43_009 incorporated “Changing computation method for speech quality from MOS to PESQ ”		Marko Onikki, TeliaSonera
3.5	9/12/2009	Incorporate outputs of Global roaming Quality project	IREG#55, EMC#79	Marko Onikki, TeliaSonera
3.6	4/1/2011	Signal Docs 49_16 and 50_009 incorporated	IREG Signal	Marko Onikki, TeliaSonera
4.0	6/6/2011	Signal Docs 53_009 incorporated	IREG Signal	Marko Onikki, TeliaSonera
5.0	22/5/2015	Incorporated CR1001 and CR1002	IREG	Marko Onikki TeliaSonera
6.0	5/11/2015	Incorporated CR1003	NG	Marko Onikki TeliaSonera
7.0	18/10/2016	CR1004 Add VoLTE and ViLTE roaming quality parameters	NG	Marko Onikki Telia Company
8.0	29/8/2017	CR1005 (Introduction of SSI and SQI calculations	NG	Marko Onikki Telia Company

B.2 Other Information

Type	Description
Document Owner	GSMA NG
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