6 INDICATORS OF QUALITY OF SERVICE

This section defines quality indicators that characterize the performance of services supported on mobile communication systems in their various phases of access and use[6].

6.1 RADIO COVERAGE

6.1.1 RADIO NETWORK AVAILABILITY

Network availability is the probability that mobile services are available to a user, via a radio infrastructure (network radio coverage).

$$Radio\ Network\ Availability\ [\%] = \frac{\sum Measurements\ with\ Available\ Mobile\ Services}{\sum Measurements\ Conducted} \times 100$$

(6.1.1)

Mobile services are considered to be available when radio signal levels show values above the minimum thresholds that allow their use. These thresholds may be adjusted by mobile operators and usually present different values for GSM, UMTS and LTE.

As appropriate measurement equipment - RF Scanner -, combined with a geo-referencing system makes it possible to obtain coverage levels of mobile networks on the routes under study.

Table 2 –GSM, UMTS and LTE Coverage Levels

Coverage	GSM	UMTS	LTE
Very good	$-75\ dBm \leq RxLev$	$-85\ dBm \leq CPICH\ RSCP$	$-95\ dBm \leq RSRP$
Good	$-85\ dBm \leq RxLev < -75\ dBm$	$-95\ dBm \leq CPICH\ RSCP < -85\ dBm$	$-105\ dBm \leq RSRP < -95\ dBm$
Acceptable	$-95\ dBm \leq RxLev < -85\ dBm$	$-105\ dBm \leq CPICH\ RSCP < -95\ dBm$	$-115\ dBm \leq RSRP < -105\ dBm$
Poor	$-105\ dBm \leq RxLev < -95\ dBm$	$-115\ dBm \leq CPICH\ RSCP < -105\ dBm$	$-125\ dBm \leq RSRP < -115\ dBm$
Non-existent	$RxLev < -105\ dBm$	$CPICH\ RSCP < -115\ dBm$	$RSRP < -125\ dBm$

[6] These *Service Quality Indicators* are based on the following technical specifications: ETSI TS 102 250, namely part 2, ETSI TR 101 578, ETSI TR 102 678, ETSI TR 102 505, ETSI TR 102 581, ETSI EG 202 057, namely parts 3 and 4, ETSI TS 100 910, ETSI TS 143 022, ETSI TS 145 008, ETSI TS 125 304, ETSI TS 136 304, ETSI TS 136 133; on the following recommendations ITU-T P.863, ITU-T P.863.1, ITU-T J.343, ITU-T J.343.1, ITU-T P.910 and ITU-T Q.3960; and on the following reports ECC Report 256 and ECC Report 103 [ETSI TS 102 250-2, ETSI TR 101 578, ETSI TR 102 678, ETSI TR 102 505, ETSI TR 102 581, ETSI EG 202 057-3, ETSI EG 202 057-4, ETSI TS 100 910, ETSI TS 143 022, ETSI TS 145 008, ETSI TS 125 304, ETSI TS 136 304, ETSI TS 136 133, ITU-T P.863, ITU-T P.863.1, ITU-T J.343, ITU-T J.343.1, ITU-T P.910, ITU-T Q.3960, ECC Report 256, ECC Report 103].

6.2 VOICE SERVICE

6.2.1 ACCESSIBILITY OF THE VOICE SERVICE

Service accessibility is the probability that the end-user can access the voice service, that is, the probability of being successful when making calls.

A call is considered to be "Set Up with Success" if it reaches the called terminal (a "call signal" is heard on the caller terminal).

$$Service\ accessibility\ [\%] = \frac{\sum Sucessfully\ Set\ Up\ Calls}{\sum Attempts\ to\ Set\ Up\ Calls} \times 100$$

(6.2.1)

6.2.2 VOICE CALL SET UP TIME

Call set up time is the period of time elapsing from the sending of a complete destination address (target telephone number) to the setting up of a call.

$$Call\ Set\ up\ time\ [s] = t_{calling\ signal} - t_{address\ sending}$$

$t_{address\ sending}$ – moment when the user presses the send button.
$t_{calling\ signal}$ – moment when the call is successfully set up (one hears the "call signal" on the caller terminal).

(6.2.2)

6.2.3 VOICE CALL COMPLETION RATE

The voice call completion rate is the probability that a call has, after being successfully set up, to be maintained during a period of time, ending normally, that is, according to the user's will.

$$Call\ Completion\ Rate\ [\%] = \frac{\sum Normally\ ended\ calls}{\sum Sucessfully\ Set\ Up\ Calls} \times 100$$

(6.2.3)

6.2.4 VOICE CALL AUDIO QUALITY

This indicator quantifies the perceptibility of the conversation during a voice call. Both communication directions are assessed and only calls ending normally are considered.

The assessment of this QoS indicator consists in the comparison between the original audio sample that is sent, *X(t),* and the corresponding received degraded sample, *Y(t)*, on the other end of the call, by applying the POLQA algorithm[7]. The objective audio quality index obtained through this algorithm is close to what would be obtained if sample *Y(t)* were submitted to the subjective appreciation of a panel of service users.

$$Call\ Audio\ Quality_{side\ A}\ [MOS_{LQO}] = f\{X_B(t)\ ;\ Y_A(t)\}$$

$$Call\ Audio\ Quality_{side\ B}\ [MOS_{LQO}] = f\{X_A(t)\ ;\ Y_B(t)\}$$

side A; side B – designation of both ends of a voice call.
MOS_{LQO} – scale that measures the perceived audio quality (Mean Opinion Score – Listening-only Quality Objective).
f – function corresponding to the application of a calculation algorithm and conversion function of results in MOS_{LQO} values.
$X_A(t); X_B(t)$ – original audio sample sent by side A (B).
$Y_A(t); Y_B(t)$ – – degraded audio sample received by side A (B), resulting from the transmission of the original sample $X_B(t)$ $(X_A(t))$.

(6.2.4)

Results obtained by the application of the algorithm are shown on a MOS (Mean Opinion Score) type scale ranging from 1 to 5 called MOS_LQO (Mean Opinion Score – Listening-only Quality Objective), such as shown on *Table 3*. The MOS scale quantifies the effort that it takes to perceive a communication.

Table 3 - MOS_{LQO} Scale

MOS	Quality
5	Excellent
4	Good
3	Acceptable
2	Poor
1	Bad

In situations where each direction of the same call sends and receives several ("*n*") audio samples $\{X_1(t), X_2(t), \ldots, X_n(t); Y_1(t), Y_2(t), \ldots, Y_n(t)\}$, the Call Audio Quality indicator is calculated through the arithmetic average of values obtained by applying the formula shown above to each pair of audio samples, that is:

[7] POLQA – *Perceptual Objective Listening Quality Assessment* [ITU-T P.863, ITU-T P.863.1].

$$Call\ Audio\ Quality_{side\ A}\,[MOS_{LQO}] = \frac{\sum_{i=1}^{n} f\{X_{i_B}(t)\ ;\ Y_{i_A}(t)\}}{n}$$

$$Call\ Audio\ Quality_{side\ B}\,[MOS_{LQO}] = \frac{\sum_{i=1}^{n} f\{X_{i_A}(t)\ ;\ Y_{i_B}(t)\}}{n}$$

(6.2.5)

6.3 SMS – Short Message Service

6.3.1 Accessibility of the SMS Service

Service accessibility is the probability that the user can access the Short Message Service, that is, the probability of being successful when sending SMS.

$$Accessibility of\ the\ SMS\ Service\ [\%] = \frac{\sum Successfully\ Sent\ SMS}{\sum Attempts\ to\ Send\ SMS} \times 100$$

(6.3.1)

6.3.2 SMS Delivery Time

The SMS delivery time corresponds to the time between the beginning of the sending of the message to a Short Message Centre (SMSC) and the end of its reception at the destination terminal equipment.

$$SMS\ Delivery\ Time\ [s] = t_{end\_reception} - t_{start\_sending}$$

$t_{start\_sending}$ – moment when the user starts sending the SMS.

$t_{end\_reception}$ – moment when the destination terminal equipment ends reception of the SMS sent by the origination terminal equipment

(6.3.2)

Corrupted messages, messages delivered outside the predefined time frame and duplicate received messages are not taken into consideration in the calculation of this indicator. An SMS is considered to be corrupted where there is at least one-bit error.

6.3.3 SMS Delivery Rate

The SMS delivery rate is the probability that the message is delivered successfully to the destination, that is, the ratio between the number of SMS received successfully by the destination terminal equipment and the number of SMS sent by the origination terminal equipment.

$$SMS\ delivery\ rate\ [\%] = \frac{\sum Successfully\ Received\ SMS}{\sum Attempts\ to\ Send\ SMS} \times 100$$

(6.3.3)

Corrupted messages or messages delivered outside the predefined time frame are considered to have failed for the purpose of the calculation of this indicator. An SMS is considered to be corrupted where there is at least one-bit error.

Duplicate received messages are not accounted for in this indicator.

6.4 DATA SERVICES

6.4.1 COMPLETED DATA SESSION RATE (*HTTP, HTTP WEB BROWSING AND YOUTUBE VIDEO STREAMING*)

Probability that a data session (File Transfer - HTTP upload/download –, Internet browsing – HTTP web browsing – or YouTube Video Streaming) is set up and takes place successfully, that is, where it remains active during the full period of time predefined for the file transfer (HTTP upload/download), where it allows the full transfer of the webpage (HTTP web browsing) or where it allows the transfer and full playout of multimedia content (YouTube Video Streaming).

$$Completed\ Data\ Session\ Ratio\ [\%] = \frac{\sum Successfully\ Completed\ Sessions}{\sum Attempts\ to\ Set\ Up\ Sessions} \times 100$$

(6.4.1)

6.4.2 DATA TRANSFER RATE (*HTTP*)

This indicator quantifies the average data transfer rate during a File Transfer session (HTTP upload/download).

This indicator only takes account of successfully completed sessions (sessions that remained active during the full period of time predefined to transfer the file).

$$Data\ Transfer\ Rate\ [kbps] = \frac{Volume\ of\ Data\ Sent\ or\ Received\ [kbit]}{Sending\ or\ Reception\ Time\ [s]}$$

$Sending\ or\ Reception\ Time$ – Predefined period of time required to send or receive the information. Does not include the period of time required to set up data sessions (phases of packet switched network registration, PDP context activation (for GSM/UMTS) or Dedicated EPS Bearer Setup (for LTE) and remote server authentication).

(6.4.2)

6.4.3 WEBPAGE TRANSFER TIME (HTTP WEB BROWSING)

This indicator quantifies the average time required for the transfer of a webpage (reference page or another).

This indicator only takes account of successfully completed sessions (sessions that allowed the full transfer of the webpage).

$$Webpage\ Transfer\ Time\ [s] = t_{end\_reception} - t_{webpage\_request}$$

$T_{webpage\_request}$ – moment when the user equipment makes the webpage reception request.
$T_{end\_reception}$ – moment when the full webpage is received by the user equipment.

(6.4.3)

6.4.4 CONTENT DISPLAY DELAY (YOUTUBE VIDEO STREAMING)

The Content Display Delay within a YouTube Video Streaming session is the period of time between the request for multimedia contents (the user presses “play”) and the start of video playout (display of the first frame) in the terminal equipment of the user.

$$Content\ Display\ Delay\ [s] = t_{start\ of\ playout} - t_{content\ request}$$

$T_{content\ request}$ – moment when the user makes the request for contents.
$T_{start\ of\ playout}$ – moment when the playout of contents requested start in the user equipment.

(6.4.4)

6.4.5 FREEZE DURATION (YOUTUBE VIDEO STREAMING)

This indicator aggregates all interruptions or freezing events during a YouTube Video Streaming session that ends normally. Freezes are only considered where they are recognized by the user (where they exceed 120 ms [ETSI TR 101 578]).

$$Freeze\ Duration\ [s] = \sum_{i=0}^{n} (Freeze\ Duration)_i\ [s]$$

n – total number of freezes during a session

(6.4.5)

6.4.6 VIDEO QUALITY (YOUTUBE VIDEO STREAMING)

This indicator quantifies the visual quality of the communication during the YouTube Video Streaming session. Only sessions that are normally completed are considered.

The video quality is estimated through the algorithm defined by ITU in its Recommendation J.343.1 [ITU-T J.343.1]. This algorithm is based on a no-reference[8] model (Hybrid-NRe – hybrid no-reference encrypted), that is, the video quality is estimated via analysis of the received video, the video originally transmitted not being known.

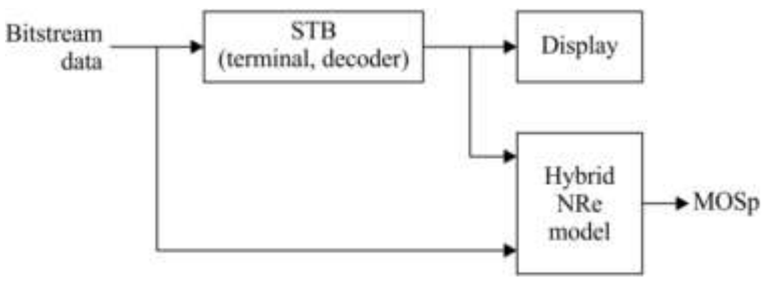

Figure 3 – Block-diagram of the Hybrid-NRe model [ITU-T J.343.1].

The Hybrid-NRe model measures the visual effect of spatial and temporal degradations as a result of video coding, erroneous transmission or video rescaling.

The visual quality estimated by this model is presented on a MOS (Mean Opinion Score) scale, ranging from 1 to 5, derived from the ACR (absolute category rating) scale defined by ITU in Recommendation

[8] The use of a "no-reference" model allows for tests with any public content available on YouTube to be conducted.

P.910 [ITU-T P910], presented in Table 4.

Table 4 - MOS Scale [ITU-T P.910]

MOS	Quality
5	Excellent
4	Good
3	Acceptable
2	Poor
1	Bad

6.4.7 VIDEO STREAMING RESOLUTION (YOUTUBE VIDEO STREAMING)

In a YouTube Video Streaming session, the YouTube allows the dynamic adjustment of the video resolution of transmitted contents, optimizing it to the available bandwidth and characteristics of the mobile terminal, and improving the viewing experience. Only sessions that are normally completed are considered.

$$Video\ streaming\ resolution\ [p] = \frac{\sum_{i=1}^{n}(Video\ clip\ resolution)_i\ [p]}{n}$$

n – total number of video clips that make up the contents received in the mobile terminal equipment

(6.4.6)

6.4.8 DATA TRANSFER LATENCY

This indicator quantifies the time required for an information packet to be delivered from the user equipment to the *Dedicated Server* or *vice-versa*. This delay corresponds to half the Round Trip Time (RTT) obtained by the Ping tool (ICMP echo).

$$Data\ Transfer\ Latency\ [ms] = \frac{Ping_{RTT}[ms]}{2}$$

(6.4.7)