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TECHNICAL SPECIFICATION

**Speech and multimedia Transmission Quality (STQ);
Reference benchmarking,
background traffic profiles and KPIs;
Part 4: Reference benchmarking for IPTV,
Web TV and RCS-e Video Share**

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Foreword

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

The present document is part 4 of a multi-part deliverable covering the Reference benchmarking, background traffic profiles and KPIs as identified below:

Part 1: "Reference benchmarking, background traffic profiles and KPIs for VoIP and FoIP in fixed networks";

Part 2: "Reference benchmarking and KPIs for High speed internet";

Part 3: "Reference benchmarking, background traffic profiles and KPIs for UMTS and VoLTE";

Part 4: "Reference benchmarking for IPTV, Web TV and RCS-e Video Share".

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Introduction

The present document describes the reference benchmarking, background traffic profiles and key performance Indicators for IPTV, WebTV and Video RCS-e.

1 Scope

The offer of new NGN services requires new:

- KPIs;
- QoS measurement; and
- benchmarking methods;

which are needed to ensure the quality of new services. To ensure the comparability of test results, reference benchmarking methods and background traffic load profiles are needed.

The present document describes key performance indicators and benchmarking methods for the spectrum of potential applications. All access technologies offered by the operator under test are considered.

2 References

2.1 Normative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the reference document (including any amendments) applies.

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

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

The following referenced documents are necessary for the application of the present document.

[1] Recommendation ITU-T E.800 (2008): "Definitions of terms related to quality of service".

[2] Recommendation ITU-T Y.1541 (2011): "Network performance objectives for IP-based services".

[3] IETF RFC 3357: "One-way Loss Pattern Sample Metrics".

[4] Recommendation ITU-T P.1201 (2012): "Parametric non-intrusive assessment of audiovisual media streaming quality".

[5] Recommendation ITU-T P.1201.2 (2012): "Parametric non-intrusive assessment of audiovisual media streaming quality - Higher resolution application area".

[6] Recommendation ITU-T P.1202 (2012): "Parametric non-intrusive bitstream assessment of video media streaming quality".

[7] Recommendation ITU-T P.1202.2 (2013): "Parametric non-intrusive bitstream assessment of video media streaming quality - Higher resolution application area".

[8] Recommendation ITU-T P.1201 Amendment 2 (2013): "New Appendix III - Use of P.1201 for non-adaptive, progressive download type media streaming".

2.2 Informative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the reference document (including any amendments) applies.

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] ETSI TR 101 578 (V1.1.1): "Speech and multimedia Transmission Quality (STQ); QoS aspects of TCP-based video services like YouTube™".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

benchmark: evaluation of performance value/s of a parameter or set of parameters for the purpose of establishing value/s as the norm against which future performance achievements may be compared or assessed

NOTE: The definition is taken from Recommendation ITU-T E.800 [1].

3.2 Abbreviations

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

AAC	Advanced Audio Coding
CS	Circuit Switched
GSMA	GSM Association
HDTV	High-Definition Television
HEVC	High Efficiency Video Coding
HTTP	HyperText Transfer Protocol
IP	Internet Protocol
IPDV	IP Delay Variation
IPLR	IP Packet Loss Ratio
IPTD	IP Packet Transfer Delay
IPTV	IP Television
ITU-T	Telecommunication Standardization Sector of the International Telecommunications Union
MOS	Mean Opinion Score
MPEG	Moving Picture Experts Group
NGN	New Generation Network
PCAP	Packet CAPtures
QoE	Quality of Experience
QoS	Quality of Service
RCS	Rich Communication Services
RFC	Requests for Comments
SDTV	Standard-Definition Television
SIP	Session Initiation Protocol
STB	Set-Top Box
STQ	Speech and multimedia Transmission Quality
TCP	Transmission Control Protocol
TS	Technical Specification
TV	Television
UMTS	Universal Mobile Telecommunications System

UNI	User Network Interface
UNI-to-UNI	User Network Interface to User Network Interface
WAP	Wireless Access-Point
WSP	Wireless Session Protocol

4 Management Summary

The assessment of IPTV QoS and QoE from an end-user perspective is performed through MOS values for the video and audio signals in the transmitted streams while using the set-top box and the associated remote control.

The video quality is estimated based on several parameters extracted from the transmitted bitstream.

The measurement of the so-called "zapping time" provides information about the time it takes to switch between TV channels.

5 Reference Benchmarking

5.1 Introduction

To determine the quality of IPTV, WebTV and RCS-e the following measurement values are determined.

Table 1: Overview of quality benchmarks for IPTV measurements

IPTV	
1.	IPTV Quality (P.1201, P.1202)
2.	IPTV STB Startup Response Time
3.	IPTV STB Startup Time
4.	IPTV STB Startup Failure Ratio
5.	IPTV Zapping Response Time
6.	IPTV Zapping Time
7.	IPTV Packet Loss
8.	IPTV Bitrate Video/Other/Total

Table 2: Overview of quality benchmarks for WebTV measurements

WebTV	
1.	Progressive Download
2.	WebTV Quality
3.	WebTV Initial Loading Time
4.	WebTV Average Duration of Stalling Events
5.	WebTV QoS parameters • Player IP Service Access Failure Ratio • Player IP Service Access Time • Player Download Cut-off Ratio • Player Download Time • Player Session Failure Ratio • Player Session Time • Video IP Service Access Failure Ratio • Video IP Service Access Time • Video Reproduction Start Failure Ratio • Video Reproduction Start Delay • Video Play Start Failure Ratio • Video Play Start Time • IP Service Access Failure Ratio • IP Service Access Time • Video Session Cut-off Ratio • Video Session Time • Impairment Free Video Session Ratio • Expected Size • Downloaded Size • Compression Ratio • Transfer Cut-off Ratio • Transfer Time • Mean User Data Rate • Playout Cut-off Ratio • Playout Cut-off Time • Expected Duration • Playout Duration • Freeze Occurrences Count • Accumulated Video Freezing Duration • Video Skip Occurrences Count • Accumulated Video Skips Duration • Video Maximum Freezing Duration • Video Freezing Impairment Ratio Failure Ratio • Video Freezing Time Proportion • End-to-End Session Failure Ratio

Table 3: Overview of quality benchmarks for RCS-e measurements

RCS-e Video	
1.	RCS-e Video Share Setup Time
2.	RCS-e Video Quality Score
3.	RCS-e Video Share Setup Failure Ratio
4.	RCS-e Bitrate Video/Total
5.	RCS-e I-/P-/B-Slices
6.	RCS-e Packet Loss
7.	RCS-e Video Share Failure Ratio
8.	RCS-e Video Share Drop Ratio

5.2 IPTV Quality

5.2.1 Audiovisual quality

The IPTV audiovisual quality is estimated by bitstream/parametric models, which have no access to a reference signal (see note) and are therefore called non-intrusive. An audiovisual, packet-header-based quality model is described in Recommendation ITU-T P.1201.2 [5] and two differently complex bitstream-based video quality models in Recommendation ITU-T P.1202.2 [7]. These are suited for the high resolution application areas (i.e. HDTV, including 720p25, 1080i50, 1080p25, 1080p50, 1080p60) as well as SDTV, that is, the video resolutions that are relevant for IPTV.

NOTE: A full-reference approach, in which a reference video signal is compared to the signal at the receiver side, is neither practically feasible due to the amount of data processing needed, nor guaranteeing a fair estimation of the perceived quality, as the reference content could be specifically prepared so as to deliver appealing (D)MOS results.

The models receive as input the media stream as packet captures (PCAP) and may also use out-of-band information (such as video resolution) which may not be part of the packet information. P.1201 only operates on the packet header information to provide audiovisual quality estimation, whereas P.1202 may also decode part of the video bitstream for a more precise assessment. Therefore, P.1202 is computationally more complex, and also requires a non-encrypted video payload.

The test sequences are 10 seconds long. The selected video sequences should cover different types of content. In particular, it is proposed to use at least three different types of content ranging from mostly stable scenes to fast moving scenes. Naturally, a type of the content chosen for a test should be carefully described in a test report. For the analysis, also running live IPTV sessions can be used, comparing, for example, the resulting audio, video and audiovisual MOS results across different channels and programs.

The following MOS values on a 5-point scale can be predicted by the respective models and shall be reported:

- P.1201: Audiovisual MOS, Video MOS, Audio MOS
- P.1202: Video MOS

To provide a more representative estimation of IPTV service quality, tests should be performed two times per test signal (at least three different video sequences from a content perspective involved in the measurement). This approach results in a total of at least 4 MOS values for each of the 6 measurements, totaling in at least 24 MOS values.

For describing the overall *audiovisual* quality of the service, the audiovisual MOS shall be averaged across all measurements. For describing the overall *video* quality of the service, the video MOS shall be averaged across all measurements. In order to assess variations in quality across programs and channels, it is recommended to further report the individual audio, video and audiovisual MOS values obtained for a given time-interval $T_{IPTV} \geq 10$ s.

5.2.2 Functional QoS parameters of the video channel

One functional QoS parameter of the video channel is recommended in this case, namely streaming service access time. It is defined as follows:

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

where $t_{\text{stream request}}$ is the time when a stream is requested and $t_{\text{reception of first data packet}}$ is the time when a first data packet is received. Table 4 shows trigger points, technical description and protocol parts for this parameter.

Table 4: Trigger Points for streaming service access time

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

The exact derivation of the trigger points from the signalling protocol under test in the plug test has to be determined by the test house and shall be stated in the test report.

5.2.3 Network performance parameters in the video channel

The following parameters between UNI and UNI in both directions of transmission shall be monitored during the period when the end-to-end audiovisual quality and QoS parameters are being tested:

- UNI-to-UNI one way packet delay over time.
- UNI-to-UNI one way packet delay variation over time.
- UNI-to-UNI one way packet loss over time, and additionally the following performance metrics:
    - Loss noticeable rate - a loss of packet is considered "noticeable" if the distance between the lost packet and the previously lost packet is no greater than delta, a positive integer, where delta is the "loss constraint". More information about this metric is available in IETF RFC 3357 [3].

NOTE 1: As "noticeable" means noticeable at a network layer and not an application layer. If an impact of loss on the user perception is of interest, a measurement should be done at an application layer.

- Loss period lengths - represents the number of packets lost in each loss period, it is an indicator of burstiness of each loss period. More information about this metric is available in IETF RFC 3357 [3].
- Inter loss period lengths - measures a distance between two loss periods. More information about this metric is available in IETF RFC 3357 [3].

NOTE 2: All above mentioned parameters are measured at a network layer (UNI-UNI) and should not be confused with those measured at an application layer including also impact of corresponding application (e.g. impact of codec, jitter buffer, etc.).

In case these parameters are measured in separately set-up connections, it is not possible to relate their values to the end-to-end QoS or audiovisual quality results.

As outlined in the general notes to Table 1 of Recommendation ITU-T Y.1541 [2], an evaluation interval of 1 minute is suggested for IPTD, IPDV, and IPLR and, in all cases, the interval shall be recorded with the observed value.

5.3 IPTV benchmark measurements

5.3.1 IPTV STB Startup Response Time

This measurement is used to show the time in milliseconds needed for the first communication between the set-top box (STB) and the IPTV headend.

IPTV STB Startup Response Time is defined in the context of the present document as the time that elapses from the sending of the first packet of the STB to the receipt of the first packet of the IPTV headend.

5.3.2 IPTV STB Startup Time

This measured value shows the time in seconds which a set-top box requires to start from the standby mode.

IPTV STB Startup Time is defined in the context of the present document as the time from the sending of the infrared code for the switch-on signal to the set-top box until the video on the first channel shown after the startup is fluently played (this event is called "Full Screen" in Figure 1).

5.3.3 IPTV STB Startup Failure Ratio

If the trigger points defined in Figure 1 are not detected during the test, the STB Startup is regarded as unsuccessful.

The ratio of failed starts of the STB to the total initiated starts of STB is represented by the STB failure ratio as a percentage.

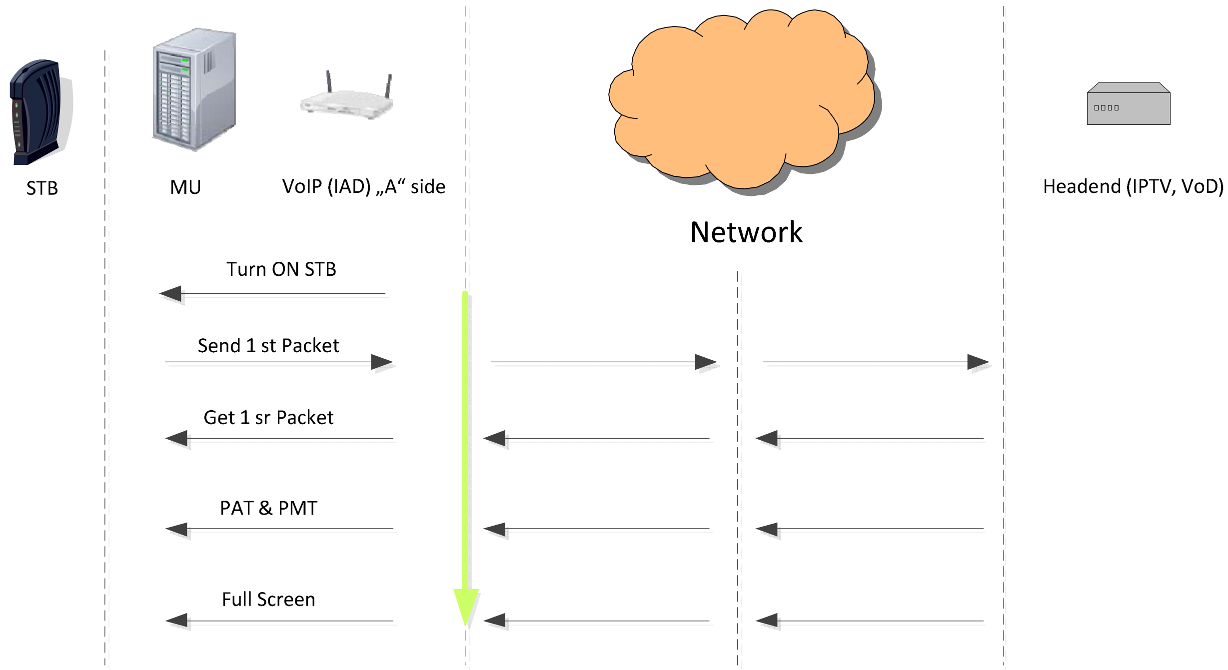

Figure 1: Measurement of the IPTV STB Startup Time

5.3.4 IPTV Zapping Response Time

This measured value shows the time in milliseconds which is needed for the first communication between the set-top box (STB) and the IPTV headend during a channel change.

STB IPTV zapping response time is defined in the context of the present document as the time which elapses from the sending of the first packet of the STB to the receipt of the first packet of the IPTV headend during a channel change.

5.3.5 IPTV Zapping Time

This measuring value shows the time in milliseconds that is required for a channel change.

IPTV Zapping Time is defined as the time that elapses from the sending of the infrared code for a channel change to the set-top box until the video on the requested channel is fluently played (this event is called "Full Screen" in Figure 2).

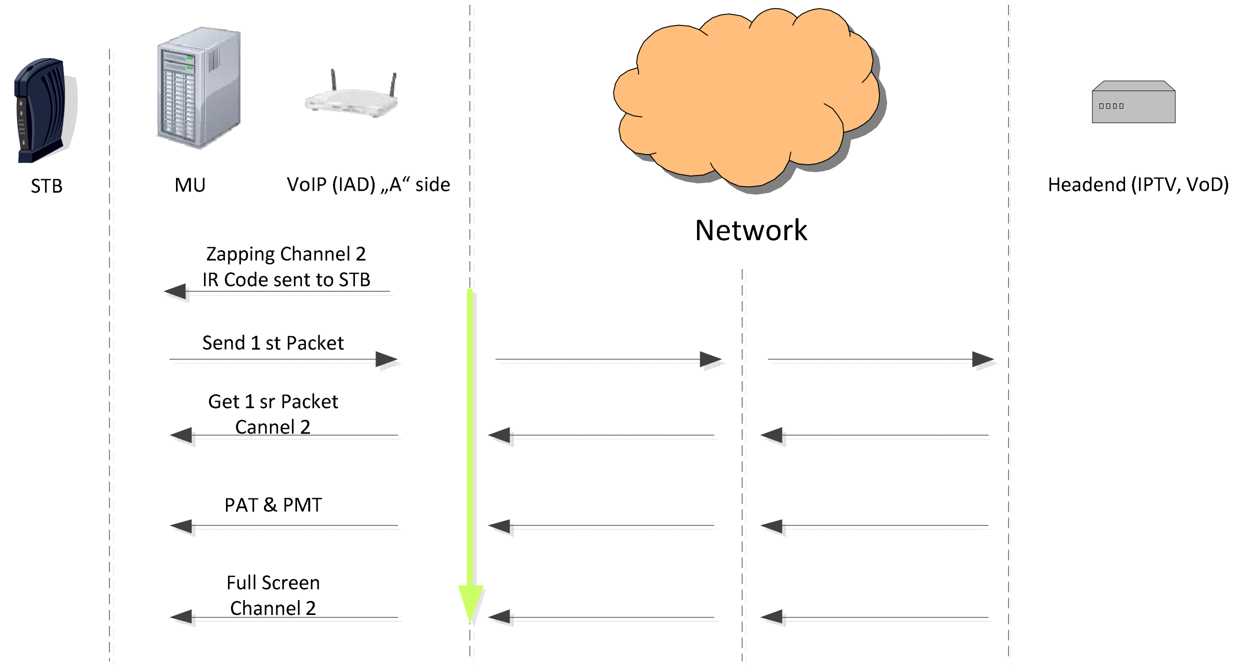

Figure 2: Measurement of the IPTV Zapping Time

5.3.6 IPTV Zapping Failure Ratio

The Zapping Failure Ratio depicts the ratio of failed channel changes to the total initiated channel changes, as a percentage.

5.3.7 IPTV Packet Loss

The IPTV Packet Loss defines the number of lost IPTV packets of an IPTV stream in percent. Correction mechanisms are taken into account and only the packets are reported that are actually lost.

IPTV Continuity Error Video/Audio: The IPTV Continuity Error is calculated from the Continuity Counter of the MPEG-TS Header and established per detected stream.

A set of further intermediate parameters provided by the Recommendation ITU-T P.1202.2 [7] are reported so as to enable diagnostic assessment of causes for low audio, video or audiovisual MOS (see Recommendation ITU-T P.1201.2 [5], Section 4, p.20 ff):

- Audio:
    - *AudioFrameLoss*, the average percentage of lost audio frames
    - *AudioBurstiness*, the average number of audio frames lost in a row
- Video:
    - *VideoBitPerPixel*, the average bits per pixel
    - *VideoContentComplexity*, a measure extracted from frame header information reflecting video content complexity
    - *VideoFreezingRatio*, the ratio of frozen frames over total video frames
    - *VideoLossMagnitude*, a parameter characterizing the spatio-temporal loss magnitude in case of slicing employed for packet-loss handling

5.3.8 IPTV Bitrate Video/Other/Total

The IPTV Bitrate in Mbit/s is divided into:

1) video bitrate
2) other
3) total

The video bitrate and the other bitrate are detected by the IPTV measuring unit. The total bitrate is calculated from the sum of the video bitrate and the other bitrate: The other bitrate includes both the audio and the data streams.

5.4 WebTV

5.4.1 Introduction

There are two major delivery technologies for Web TV video, which result in two measurement scenarios that are similar in setup, but require different analysis.

For Web TV streaming, the audiovisual content is typically encapsulated in the HTTP protocol and can include different media containers and codecs. For example, containers and codecs to be considered can include:

- MPEG-2 TS or MP4 container with H.264/HEVC video and AAC audio;
- WebM container with VP8/VP9 video and Vorbis audio.

The HTTP data is transmitted over TCP. There are two major transmission techniques that can be distinguished: progressive download and adaptive streaming.

5.4.2 Progressive Download

5.4.2.1 Introduction

The client requests a single resource from the server, which is downloaded progressively through HTTP until it is completely loaded. Client playback may begin when a playout buffer threshold is reached. Playback will stop when the playout buffer is depleted, e.g. due to bad network performance and buffer-emptying during playout. This results in visible rebuffering, that is, stalling events for the user, which impact the experienced quality.

The specific phases of Web TV playout are described in [i.1] and shown in Figure 3.

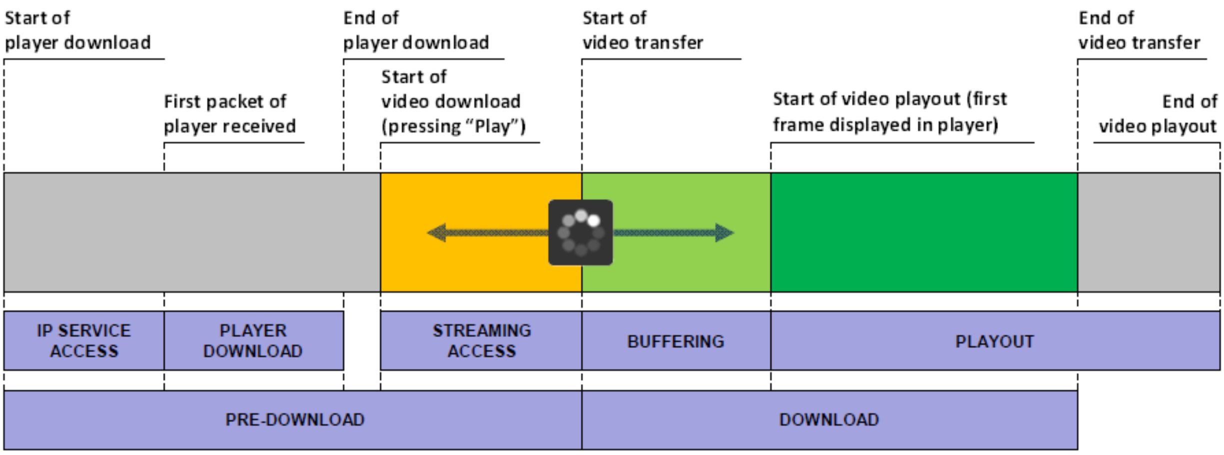

Figure 3: Phases of Progressive Download video services [i.1]

The Web TV test sequences should be of 30 s up to 60 s length. The selected video sequences should cover different types of content. In particular, it is proposed to use at least three different types of content ranging from mostly stable scenes to fast moving scenes. Naturally, a type of the content chosen for a test should be carefully described in a test report.

To provide a more representative estimation of Web TV service quality, tests should be performed two times per test sequence (at least three different video sequences from a content perspective involved in the measurement).

In addition to prepared and self-hosted content, an existing online video service may serve as content provider, but care needs to be taken that the same resource is used when comparing test results.

5.4.2.2 WebTV Quality

The overall audiovisual quality is estimated using the approach described in Recommendation ITU-T P.1201 Amd. 2 [8], Appendix III which aims at estimating the overall quality impairments due to audio and video coding as well as initial loading time and rebuffering/stalling events.

The model receives as input meta-information about the video and audio, such as codecs and bitrates, as well as video frame sizes and types (i.e. I/P/B). No decoding of the video is performed, which results in lower computational complexity.

The model outputs a MOS value on a 5-point scale, corresponding to the audiovisual quality of the session, including degradations from initial loading and stalling.

The degradation due to initial loading and stalling events during the session are estimated based on the initial loading time, the number of stalling events and their average duration. These indicators should be determined by monitoring the playback client's behaviour (e.g. through a programming interface), and should not be estimated by assuming a certain playout buffer behaviour. In the case that client behaviour cannot be monitored precisely, a theoretical model of a playout buffer may be used to determine initial loading and stalling event times. The assumed playout buffer thresholds shall be reported in such a case.

A total of at least 6 MOS values shall be generated from at least 3 sequences with 2 measurements each. The overall service quality is described by the average of these MOS values.

Alongside the overall media session MOS score, as diagnostic information, the P.1201 Appendix III model delivers additional outputs:

- multimedia (audiovisual) MOS
- video MOS
- audio MOS
- stalling and initial buffering degradation score

Reporting this information enables the identification of differences between different streamed contents complementary to the initial buffering and stalling degradation.

5.4.2.3 WebTV Initial Loading Time

This value corresponds to the time in milliseconds from the playback intent to the first start of the media presentation. Delays due to page loading and player initialization are not included. In other words, this corresponds to the time the user would see a "loading" indicator shown in the player window.

5.4.2.4 WebTV Number of Stalling Events

The number of stalling events in the session, excluding the initial loading time.

5.4.2.5 WebTV Average Duration of Stalling Events

The average duration of stalling events in the session, excluding the initial loading time.

5.4.2.6 WebTV QoS parameters

The relevant QoS parameters for a Progressive Download WebTV service are described in [i.1], Table 1. The following parameters are identified:

- Player IP Service Access Failure Ratio
- Player IP Service Access Time

- Player Download Cut-off Ratio
- Player Download Time
- Player Session Failure Ratio
- Player Session Time
- Video IP Service Access Failure Ratio
- Video IP Service Access Time
- Video Reproduction Start Failure Ratio
- Video Reproduction Start Delay
- Video Play Start Failure Ratio
- Video Play Start Time
- IP Service Access Failure Ratio
- IP Service Access Time
- Video Session Cut-off Ratio
- Video Session Time
- Impairment Free Video Session Ratio
- Expected Size
- Downloaded Size
- Compression Ratio
- Transfer Cut-off Ratio
- Transfer Time
- Mean User Data Rate
- Playout Cut-off Ratio
- Playout Cut-off Time
- Expected Duration
- Playout Duration
- Freeze Occurrences Count
- Accumulated Video Freezing Duration
- Video Skip Occurrences Count
- Accumulated Video Skips Duration
- Video Maximum Freezing Duration
- Video Freezing Impairment Ratio Failure Ratio
- Video Freezing Time Proportion
- End-to-End Session Failure Ratio

5.4.3 Adaptive Streaming

5.4.3.1 Introduction

Adaptive streaming is increasingly used in the context of web-based video delivery. Here, the client requests the media in several chunks from the server. Each chunk corresponds to a specific segment of the original media, encoded at a specific representation. Typically, individual segments are from 2 to 10 seconds long.

Media may be coded at different representations of varying quality (e.g. different bitrates or compression coefficients), and the client can choose which representation to request for the next segment based on the available network bandwidth. In case of bad network performance or missing playback capabilities, the client may switch to a lower quality representation to avoid stalling due to re-buffering of content. The selection of different representations may result in visible quality fluctuations, which impact the quality experienced by the user. Additionally, if network conditions do not allow complete segments to be downloaded in time, stalling events still may occur.

5.4.3.2 WebTV Quality

For adaptive streaming, the audiovisual quality model algorithm needs to factor in the possibility of fluctuating quality in addition to initial loading and stalling events. Hence, models targeted at progressive download services may not be accurate enough in their predictions.

NOTE: For estimating the audiovisual MOS, the model under study in ITU-T Study Group 12, Question 14, work item P.NATS (Parametric Non-intrusive Assessment of TCP-based multimedia Streaming quality, considering adaptive streaming) should be used. It will be specified in an updated version of the present document, as soon as the respective P.NATS recommendation will be available (targeted early 2016). Until the P.NATS standard for measuring adaptive streaming quality is available, the audiovisual quality should be estimated as follows: for each played out segment S_n in a video session, where n in $[1, 2, \ldots, N]$ and N is the total number of played out segments, the audiovisual MOS MOS_n should be calculated with the quality model in clause 5.4.2.2. The individual MOS scores MOS_n should then be mathematically averaged to form the audiovisual MOS of the session. In addition, the degradation due to initial loading and stalling should be reported according to clause 5.4.2.2 and should be subtracted from the audiovisual MOS to form the session MOS.

5.4.3.3 WebTVQoS parameters

The QoS parameters 1.1, the metrics defined in clause 5.4.2.2 are applicable to adaptive streaming services.

5.5 RCS-e Video Share

5.5.1 Introduction

Rich Communications Suite enhanced, abbreviated RCS-e and branded under the name "joyn" is a technical standard for communication services published by the industry association of international mobile operators (GSMA). Among other things, the standard enables the transmission of text messages, files, voice, and video. joyn may benefit from network-side QoS mechanisms. A main feature of joyn is the feature "Video Share". Video Share is a unidirectional link whereby a video signal is transmitted in addition to an existing CS call.

5.5.2 RCS-e Video Share Setup Time

To determine the Video Share Setup Time, the time in seconds is measured from the SIP INVITE message of the Video Share on the smartphone on the "A" side until arrival of the SIP 200 OK message on the "A" side.

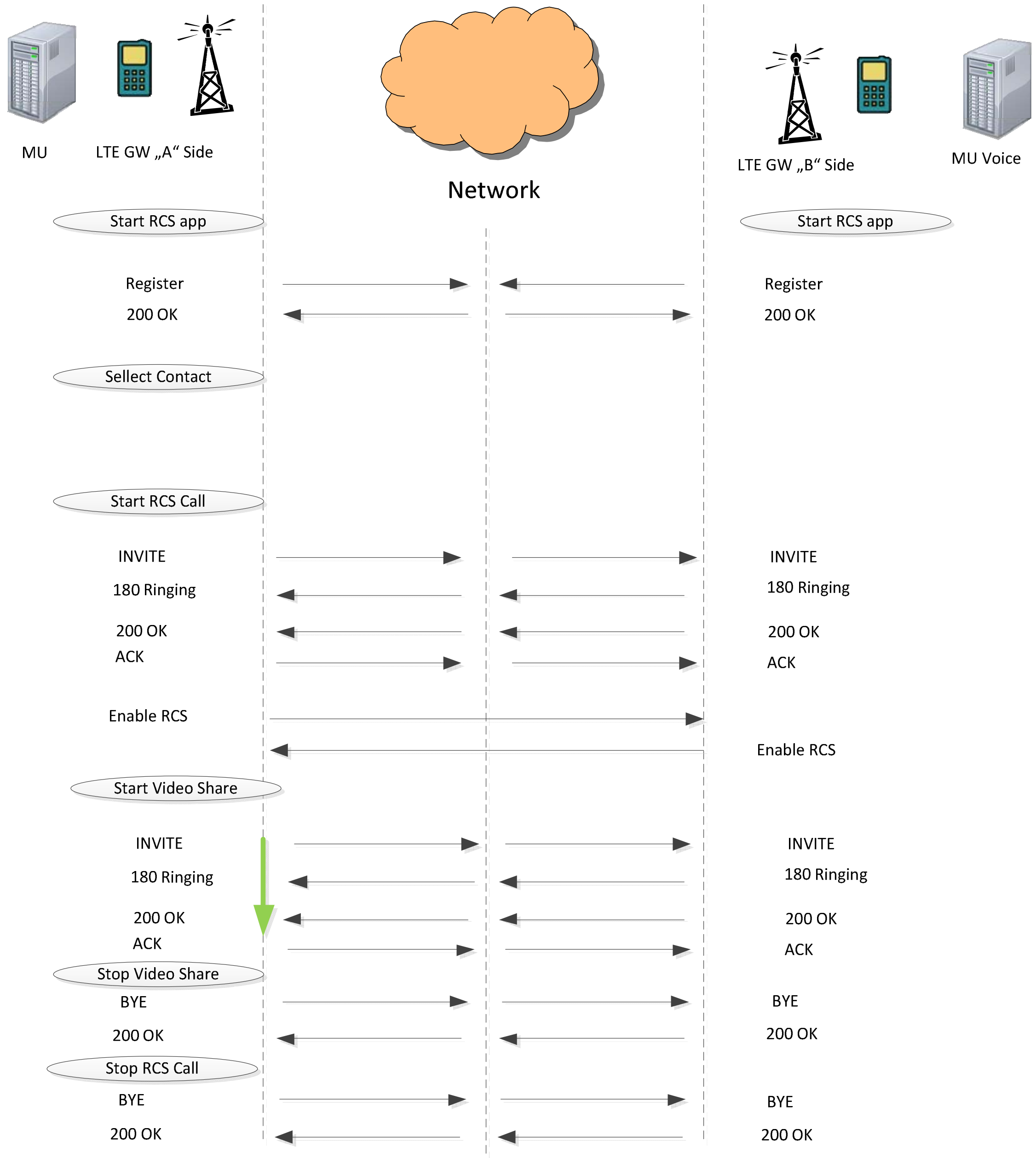

Figure 4: Measurement of the Video Share Setup Time

5.5.3 RCS-e Video Share Setup Failure Ratio

If the trigger points defined in Figure 4 are not detected during the test, the RCS-e Video Share Setup is deemed to have failed. The ratio of failed Video Share Setups to the total Video Share Setups initiated is represented by the Video Share Setup Failure Ratio as a percentage.

5.5.4 RCS-e Bitrate Video/Total

The RCS-e Bitrate in Mbit/s is divided into Bitrate for Video Share and Total. The Bitrate Video and Bitrate Total are detected by the RCS-e measuring unit. The Bitrate Total is the entire bitrate of the stream.

5.5.5 RCS-e Video Share I-/P-/B-Slices

The number of RCS-e I-/P-/B Slices are captured from the H.264 video stream and represented as a percentage frequency.

5.5.6 RCS-e Video Share Packet Loss

The RCS-e packet loss defines the number of lost IP packets of a RCS-e stream in percent. Here, correction mechanisms are taken into account and only the packets are shown that are actually lost.

5.5.7 RCS-e Video Quality Score

The RCS-e Video Quality Score, is measured according to Recommendation ITU-T P.1201 [4].

5.5.8 RCS-e Video Share Drop Ratio

The ratio of dropped Video Shares to the total number of connections initiated is represented by the RCS-e Video Share Drop Ratio as a percentage.

5.5.9 RCS-e Video Share Failure Ratio

If the trigger points defined in Figure 5 are not detected during the test, or the video is not received within 10 seconds, the RCS-e Video Share is deemed to have failed.

The ratio of failed Video Shares to the total number of Video Shares initiated is represented by the RCS-e Service Failure Ratio as a percentage.

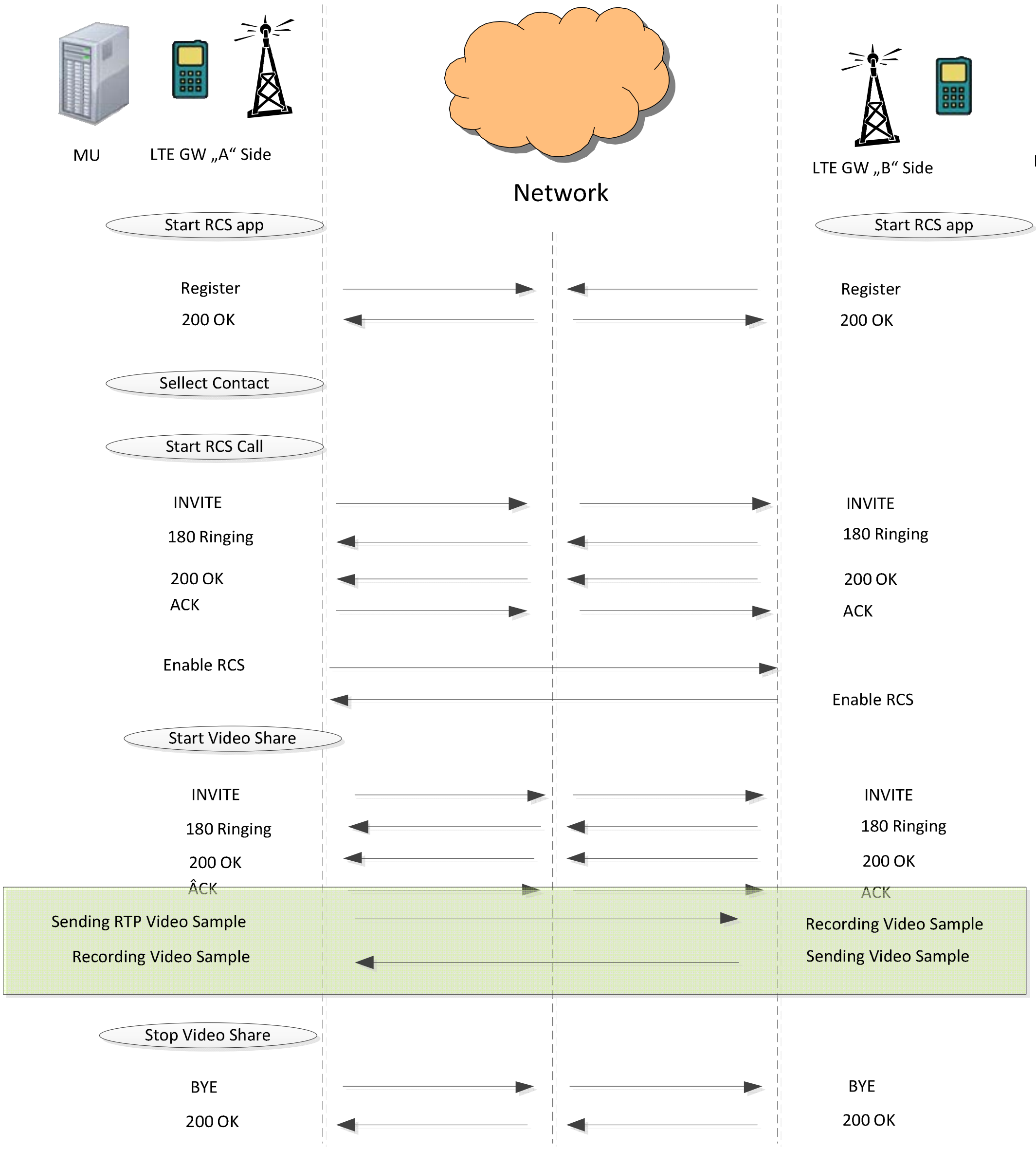

Figure 5: measurement of the RCS-e Video Quality Score

Annex A (informative): Bibliography

ETSI TR 101 290 (V1.3.1): "Digital Video Broadcasting (DVB); Measurement guidelines for DVB systems".

History

Document history		
V1.1.1	December 2015	Publication