ETSI TR 126 925 V18.1.0 (2024-05)

5G;
Typical traffic characteristics of media services on 3GPP networks
(3GPP TR 26.925 version 18.1.0 Release 18)

Reference
RTR/TSGS-0426925vi10

Keywords
5G

ETSI

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Contents

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Introduction

The document presents typical media traffic characteristics (including bandwidth and latency requirements) that are of importance for 3GPP standardization work. This includes demands based on current services, but also expectations for new services or emerging services, considering developments in the industry in terms of efficiency improvements.

1 Scope

The present document includes following information:

- Media centric Third-Party and Operator services currently deployed or expected to be deployed until 2025 on 3GPP defined 4G and 5G networks.
- Typical deployment characteristics today, such as bandwidth requirements, client buffered and rate adaptation reception characteristics, codecs, protocols in use and latency requirements.
- An overview on technological developments for existing and emerging services and their impact on typical traffic characteristics of media services, e.g. evolution of compression technologies, new demands for high quality, new experiences, etc.
- A summary on typical characteristics and requirements for different media services on 3GPP networks.
- An identification of the applicability of existing 5QIs for such services and potentially identify requirements for new 5QIs or QoS related parameters.

2 References

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

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[2] 3GPP TS 26.234: "Transparent end-to-end Packet-switched Streaming Service (PSS); Protocols and codecs".

[3] 3GPP TS 26.247: "Transparent end-to-end Packet-switched Streaming Service (PSS); Progressive Download and Dynamic Adaptive Streaming over HTTP (3GP-DASH)".

[4] 3GPP TS 26.346: "Multimedia Broadcast/Multicast Service (MBMS); Protocols and codecs ".

[5] 3GPP TS 26.116: "Television (TV) over 3GPP services; Video profiles".

[6] 3GPP TS 26.118: "3GPP Virtual reality profiles for streaming applications".

[7] 3GPP TS 26.114: "IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction".

[8] 3GPP TS 26.223: "Telepresence using the IP Multimedia Subsystem (IMS); Media handling and interaction ".

[9] Recommendation ITU-R BT.1872-3 (10-2019): " User requirements for broadcast auxiliary services including digital television outside broadcast, electronic/satellite news gathering and electronic field production".

[10] 3GPP TS 23.501: "System Architecture for the 5G System ".

[11] 3GPP TS 26.238: "Uplink streaming".

[12] ST 2110-10:2017 - SMPTE Standard - Professional Media Over Managed IP Networks: System Timing and Definitions.

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[33] ST 2082-1:2015 - SMPTE Standard - 12 Gb/s Signal/Data Serial Interface — Electrical ST 2082-1:2015.

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[46] 3GPP TR 26.955, " Video codec characteristics for 5G-based services and applications".

[47] 3GPP TR 26.926, "Traffic Models and Quality Evaluation Methods for Media and XR Services in 5G Systems".

3 Definitions of terms, symbols and abbreviations

3.1 Terms

For the purposes of the present document, the terms and definitions given in 3GPP TR 21.905 [1] and the following apply. A term defined in the present document takes precedence over the definition of the same term, if any, in 3GPP TR 21.905 [1].

360 VR: Virtual Reality 360° video and 3D audio content.

720p: a video with an image resolution of 1 280 × 720 pixels.

4K UHD: a video with an image resolution of 3 840 x 2 160 pixels a.k.a. 2160p.

8K UHD: a video with an image resolution of 7680 x 4 320 pixels a.k.a. 4320p.

3.2 Symbols

Void.

3.3 Abbreviations

For the purposes of the present document, the abbreviations given in 3GPP TR 21.905 [1] and the following apply. An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in 3GPP TR 21.905 [1].

A/V	Audio/Visual
FEC	Forward Error Correction
FFS	For Further Study
HD	High Definition
PSS	Packet Switched Streaming
STB	Set-Top-Box
TV	Television
UHD	Ultra High Definition
VR	Virtual Reality

4 Media centric Third-Party and Operator services

4.1 A/V Streaming - Enhanced TV

Live and on-demand audio-visual streaming and enhanced TV services are considered according to 3GPP PSS [2], 3GP-DASH [3], MBMS [4], 5G Media Streaming for downlink [45] and the media profiles in TS 26.116 [5].

4.2 VR 360° Streaming

Live and on-demand VR 360° streaming are considered according to 3GPP PSS [2], 3GP-DASH [3], MBMS [4], 5G Media Streaming [45] and the media profiles in TS 26.118 [6].

4.3 Conversational Multimedia Telephony and Telepresence

Conversational Multimedia Telephony and Telepresence are considered according to MTSI [7] and IMS Telepresence [8].

4.4 Live uplink A/V streaming

Professional and consumer live uplink streaming contribution of A/V content are considered according to FLUS [11] and 5G Media Streaming for uplink [45].

4.5 Cloud gaming

Cloud gaming (a.k.a. game streaming) implies that, while the game is being played by a user on a UE or on a device attached to a UE, game processing and rendering is totally or partly performed in a network entity, potentially at the edge of the network. The traffic typically consists of uplink and downlink game status/control information traffic between a client and a server and of downlink streaming of rendered and encoded 2D or VR360 video.

4.6 XR Media Services

For details on XR Services including AR and VR, refer to 3GPP TR 26.928 [30] and 3GPP TR 26.926 [47]. Identified scenarios are:

- Viewport-independent 6DOF streaming, see TR 26.928 [30], clause 6.2.2.
- Viewport-dependent 6DOF streaming, see TR 26.928 [30], clause 6.2.3.

- Raster-based split rendering, see TR 26.928 [30], clause 6.2.5.
- Generalized split rendering, see TR 26.928 [30], clause 6.2.6.
- XR conversational, see TR 26.928 [30], clause 6.2.7.

5 Typical current deployment characteristics

5.1 Typical streaming/broadcast video and audio bitrates

These figures are valid for both HDR/non-HDR video:

- 720p HD: 2 - 5 Mbps

NOTE 1: Today typically 3 Mbps for HEVC [15] and 5Mbps for AVC [14], but bitrate reductions expected with better encoding and coding tools. See Clause 6.1.2.

- Full HD: 3 - 12 Mbps

NOTE 2: Today typically 5-7 Mbps for HEVC [15] and 10-12 Mbps for AVC [14], but bitrate reductions expected with better encoding and coding tools. See Clause 6.1.2 as well as TR 26.955 [46].

- 4k UHD: 5- 25Mbps

NOTE 3: Today typically 8-16 Mbps for HEVC [15] and 15-25 Mbps for AVC [14], but bitrate reductions expected with better encoding and coding tools. See Clause 6.1.2 as well as TR 26.955 [46].

- 8k UHD: 25 - 80 Mbps

NOTE 4: Initially up to 80 Mbps for HEVC [15], but bitrate reductions expected with better encoding and coding tools. More advances with new codecs are expected, See Clause 6.1.2.

These figures apply for audio:

- Normal quality audio: mono/stereo: 24-48 kbps
- High quality audio: mono/stereo/immersive 24-512 kbps
- Extreme quality audio: mono/stereo/immersive 512 kbps

5.2 Typical streaming/broadcast 360 VR bitrates

The following bitrates apply for streaming/broadcast 360 VR.

- Basic 360 VR: 2.5 - 25 Mbps

NOTE 1: For viewport agnostic, the 4k UHD numbers as defined in clause 5.1 apply. For viewport dependent, the bitrates likely can be reduced to half.- According to [44], with HEVC at 1080p@30fps between 2.5 - 5 Mbps on average with caps at 5 – 8 Mbps.

- HD 360 VR: 10 - 80 Mbps

NOTE 2: For viewport agnostic, the 8k UHD numbers as defined in clause 5.1 apply. For viewport dependent, the bitrates likely can be reduced to half.

- According to [44], with HEVC at 4K@30fps between 10 - 18 Mbps on average with caps between 15 – 25 Mbps.
    - Retinal VR: 15 - 150 MbpsNOTE 3: For viewport agnostic, the 8k UHD numbers multiplied by 4 as defined in clause 5.1 apply. For viewport dependent, the bitrates likely can be reduced to half or even on third.

- According to [44], with HEVC at 8K@30fps between 30 - 35 Mbps on average with caps at 42 Mbps.

NOTE 4 on framerates: cinema content is usually captured and distributed at 24 fps. TV content is usually captured and distributed at 50 or 60 fps depending on the region. The above video bitrates include frame rates of up to 60fps.

NOTE 5 on bitrate ranges: the bitrates are dependent on codec (e.g. AVC/H.264 [14] or HEVC/H.265 [15]), on content type and whether it is live or on-demand. Bitrates are expected to be reduced with expected encoder implementation enhancements and new codecs. In the past, that reduction has been observed to be in the order of 50 % every 10 years.

5.3 Typical conversational speech/audio and video bitrates

The following bitrates are typical in commercial services according to clause 4.3:

- Narrowband voice (mono): 7.2-13.2 kbps
- Wideband voice (mono): 7.2-24.4 kbps
- Super-wideband voice (mono): 9.6-24.4 kbps
- Fullband voice (mono): 16.4-[TBD] kbps

NOTE: Unlike NB, WB and SWB which all are audio bandwidths used in commercial networks, no FB usage has yet been seen for commercial conversational services.

- VGA video: 300 - 900 kbps
- 720p HD video: 800 - 1500 kbps
- Telepresence video 1080p: 1500 - 3000 kbps

5.4 Typical uplink A/V streaming of A/V, including 360 VR content

5.4.1 Professional production content bitrates

Table 5.4.1-1 lists common content formats from professional cameras that are output in uncompressed form over the SDI interface, and their corresponding bit rates:

NOTE 1: The formal standard references are listed in table 5.4.1-1.

NOTE 2: See also 3GPP TR 22.827 [35].

NOTE 3: The SDI rates given below are for wired connections within professional production environments.

Table 5.4.1-1: SDI interface, and their corresponding bit rates

Standard	Name	Bitrates	Example video formats
SMPTE 259M [31]	SD-SDI	270 Mbit/s, 360 Mbit/s, 143 Mbit/s, and 177 Mbit/s	480i, 576i
SMPTE 344M [31]	ED-SDI	540 Mbit/s	480p, 576p
SMPTE 292M [31]	HD-SDI	1.485 Gbit/s, and 1.485/1.001 Gbit/s	720p, 1080i
SMPTE 372M [31]	Dual Link HD-SDI	2.970 Gbit/s, and 2.970/1.001 Gbit/s	1080p60
SMPTE 424M [31]	3G-SDI	2.970 Gbit/s, and 2.970/1.001 Gbit/s	1080p60
SMPTE ST-2081 [32]	6G-SDI	6 Gbit/s	2160p30
SMPTE ST-2082 [33]	12G-SDI	12 Gbit/s	2160p60
SMPTE ST-2083 [34]	24G-SDI	24 Gbit/s	2160p/4k@120,8k@60

SMPTE ST 2110 [12] can support transport of compressed or uncompressed streams from cameras with the resolutions above. According to [13], in case of uncompressed streams, the video rates for YCbCr 4:2:2 10b format can be derived as follows.

Table 5.4.1-2: ST 2110 Video uncompressed streams bitrates

Format	Corresponding ST 2110 Video uncompressed bitrate (YCbCr 4:2:2 10b) [12] in Gbps
480p30, 576p25	0.221
720p50/60	0.982 / 1.178
1080p50/60	2.210 / 2.650
2160p50/60	8.842 / 10.600
2160p100 (4K)	17.684
4320p50/60/100 (8K)	36.103 / 43.280 / 72.206

Audio uncompressed channels come in addition, but their bitrates are some magnitude smaller than uncompressed video.

Light compression can also be used for such production environments using for example VC-2 (ST 2042-1:2017 - SMPTE Standard - VC-2 Video Compression) [36] which is typically used with a coding ratio of 4:1 and MJ2 - Motion JPEG 2000 (ISO/IEC IS 15444-3 | ITU-T T.802)[37] which is typically used with a coding ratio of 6:1.

Compression is used for portable cameras equipped with wireless (Wireless LAN and 4G LTE) modules. The bitrates are 9, 6, 3, 2Mbps for up to 720p@60fps and 9, 6, 3 Mbps for 1080@30fps with AVC/H.264 [14].

5.4.2 Live uplink contribution professional content bitrates

Contribution feeds as seen in real deployed systems:

- 80 Mbps 422 10 bits for HD @60fps
- 165 Mbps 422 10 bits for UHD (4K, AVC/H.264 [14]) @60fps
- 120 Mbps 422 10 bits for UHD (4K, HEVC/H.265 [15]) @60fps

- Normal quality 360 VR: [TBD 96 Mbps]
- High quality 360 VR: [TBD 140 Mbps]

NOTE 1: The above video profiles include frame rates of up to 60fps.

- Broadcast auxiliary services (e.g. electronic news gathering) according to Recommendation ITU-R BT.1872-3 [9]:21-35 Mbps for HD TV (AVC/H.264 [14])
- 18-30 Mbps for HD TV (HEVC/H.265 [15])
- 96-145 Mbps for UHD TV (4K, HEVC/H.265 [15])
- 140-285 Mbps for UHD TV (8K, HEVC/H.265 [15])
- Compressed audio: 96-180 kbps per channel
- Uncompressed audio: 768-1152 kbps per channel

NOTE 2: The lower video rates from BT.1872-3 concern a single codec while the higher rates have headroom for 3 coding steps in tandem.

5.4.3 User generated content

The following bitrates are typical for user generated content:

- 720p HD: 3 - 10 Mbps

NOTE 1: Today typically 5-10 Mbps for AVC [14] depending on frame rate and dynamic range, but bitrate reductions expected with better encoding and coding tools. See Clause 6.1.2.

- Full HD: 5 - 15 Mbps

NOTE 2: Today typically 8-15 Mbps for AVC [14], but bitrate reductions expected with better encoding and coding tools. See Clause 6.1.2.

- 4k UHD: 10 - 85 Mbps

NOTE 3: Today typically 8-16 Mbps for HEVC [15] and 15-25 Mbps for AVC [14], but bitrate reductions expected with better encoding and coding tools. See Clause 6.1.2.

- 8k UHD: 20 - 150 Mbps

NOTE 4: No information is available today, but the numbers are extrapolated from 4K by multiplying with a factor 4.

- Basic 360 VR: the numbers for 4k UHD apply
- HD 360 VR: the numbers for 8k UHD apply
- Retinal VR: this is roughly 16k as expected, so a factor 4 to 8k UHD is applicable, i.e. 40-300 Mbps
- Normal quality audio: 24.4kbps/channel

NOTE 5: Super-wideband 3GPP audio codecs may be used.

- High quality audio: 128kbps/channel

NOTE 6 on user generated content: the video bitrate ranges are wide as user generated content may have different quality expectations including semi-professional content.

5.5 Typical Traffic Characteristics for Cloud gaming

For cloud gaming, the downlink streaming of 720p/1080p/4k @60fps encoded A/V typically consists of a 5-35Mbps bitstream. One instance of a cloud gaming service requires a minimum uplink bitrate of 1.5 Mbps [21].

In the future the cloud gaming is presumed to reach up to 8k resolutions and up to 120fps downlink bitstreams. No information on such currently deployed services are available to formulate typical bitrates. However, Clause 6 provides indication that allows estimation of bitrates. Annex A provides background information on deployed cloud gaming services.

Different game types result in different round-trip user interaction delay requirements (sometimes referred also as acceptable game latency). As discussed in TR 26.928 [30], clause 4.2, with regards to such requirements, games may be divided into the following 4 types: games requiring (i) at most 50 ms, (ii) at most 100 ms, (iii) at most 200ms, and (iv) games with no latency requirements. The game latency impacts the traffic model as well as the requirements on the delivery system. The shorter the latency requirements, the higher the expected bitrate.

Cloud gaming traffic characteristics are also discussed in 3GPP TR 26.928 [30], clause 6 as well was TR 26.926 [47].

5.6 XR Traffic Characteristics

Initial typical bitrates and traffic characteristics for XR services are collected in 3GPP TR 26.928 [30], clause 6 as well was TR 26.926 [47]. A summary of expected bitrates and traffic characteristics are provided:

- Viewport-independent 6DOF streaming
  - Downlink only
  - HTTP Streaming
  - Up to 100 Mbps to address high-quality 6DOF VR services to allow 2k per eye streaming at 90 fps (see TR 26.928, clause 4.2 and 6.2.2)
- Viewport-dependent 6DOF streaming
  - Downlink only
  - HTTP streaming, HTTP/TCP level information and responses are exchanged every 100-200 ms in viewport-dependent streaming.
  - Up to 25 - 50 Mbps to address high-quality 6DOF VR services to allow 2k per eye streaming at 90 fps (see TR 26.928, clause 4.2 and 6.2.3)
- Raster-based split rendering
  - Primarily downlink
    - for H.264/AVC the bitrates are in the order of up to 50 Mbps per eye buffer, i.e. up to 100 Mbit/s.
    - for H.265/HEVC the bitrates are in the order of 20 - 30 Mbps per eye buffer, i.e. 40 – 60 Mbit/s
    - packet latency requirements are in the range of 15ms
    - Application Layer FEC may be used with overhead from 10-50%, typically something like 30%
  - uplink pose information (see TR 26.926 [47], clause 5.8)
    - typically every 10 -15ms constant packet size of up to 100 bytes
    - packet latency requirements are 10 – 15 ms
- Generalized split rendering
  - details are FFS
- XR conversational
  - details are FFS

6 Overview on technological developments for existing and emerging services

6.1 Technology Developments

6.1.1 Overview

This clause collects developments in the industry on compression advances, content formats, protocol improvements and other advances that may impact the traffic characteristics documented in clause 4.

6.1.2 Compression Improvements

Due to the increasing consumption of video content with higher resolutions, the need for more efficient video compression techniques is growing. The first version of the High Efficiency Video Coding (HEVC) standard [15], jointly developed by the ITU-T VCEG and the ISO MPEG, was finalized in 2013. A wide range of products and services support HEVC [15] for video encoding/decoding, especially for Ultra High Definition (UHD) content, where HEVC [15] can provide around 50% bitrate savings for the same subjective quality as its predecessor H.264/AVC [14].

Both codecs are defined as part of the TV Video Profiles in TS 26.116 [5] and are also the foundation of the VR Video Profiles in TS 26.118 [6].

Work on video compression technologies beyond the capabilities of HEVC [15] are continued by the MPEG/ITU, with the creation of the Joint Video Exploration Team (JVET) on future video coding in October 2015. Many new coding tools have been proposed in the context of JVET, which eventually led to a Call for Proposals on video coding technologies with video compression capabilities beyond HEVC [15]. The reference software used in the exploration phase of JVET, called Joint Exploration Model (JEM), was leveraged as the base for the majority of responses to the call. Results included responses demonstrating compression efficiency gains of around 40 % or more with respect to HEVC [15]. This initiated the work by the Joint Video Experts Team (JVET) on the development of a new video coding standard, to be known as Versatile Video Coding (VVC).

MPEG has started working on a new video coding standard to be known as MPEG-5 Essential Video Coding (EVC) in January 2019. MPEG-5 EVC aims to provide a standardized video coding solution to address business needs in some use cases, such as video streaming, where existing ISO video coding standards have not been as widely adopted as might be expected from their purely technical characteristics. In addition, a main profile adds a small number of additional tools, each of which is individually capable of being either properly deactivated or switched to the corresponding basic tool. The target coding efficiency for the call for proposals was to be at least as efficient as HEVC. This target was exceeded by approximately 24 % in the responses to the call for proposals, which were evaluated at this meeting. The development of the MPEG-5 EVC standard is expected to be completed in 2020.

Figure 6.1.2-1 shows the typical improvements of video compression rates over time as well as the target for the VVC standard. It is also observed that compression technologies have enabled the reduction of bitrates by 50 % in a time frame of 7-10 years. Most of the gains come by the increase of encoding and decoding complexity, spurred according to Moore's law.

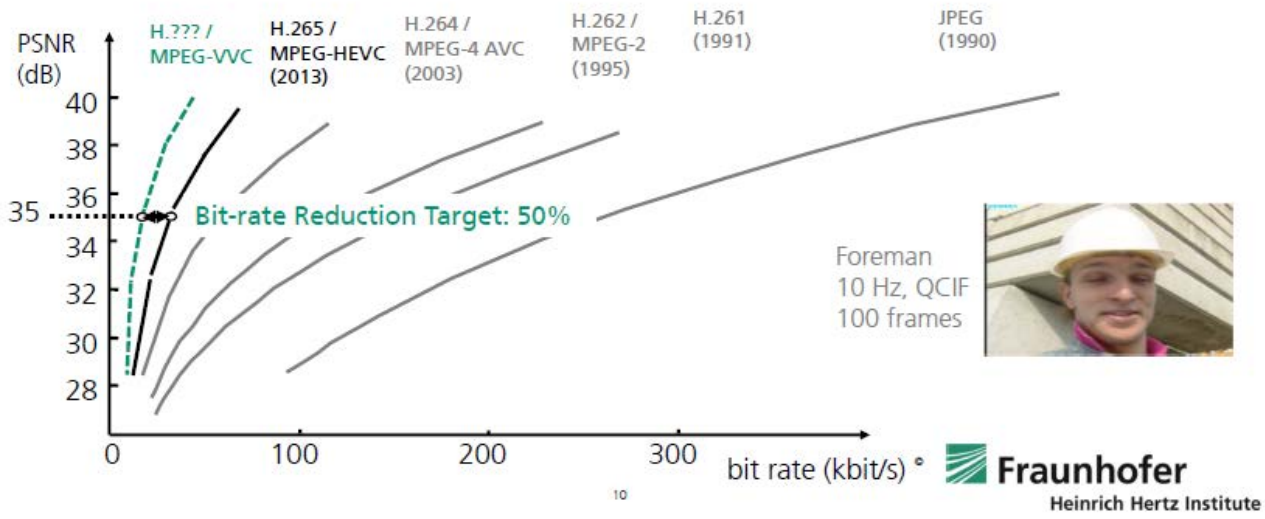

Figure 6.1.2-1: Video bitrate efficiency improvements and target for the final VVC standard [reproduced with appropriate permission from Fraunhofer]

Table 6.1.2-1 provides a summary of the expected compression efficiency of different codecs and expectations on target bitrates for different video technologies.

Table 6.1.2-1: Expected Video coding standards performance and bitrate target

Codec	Coding performance (Random-Access)		Targeted bitrate (Random Access)
	Objective	Subjective	
HEVC	-40 % vs AVC [24][25][26]	-60 % vs AVC [24][25][26]	4k: ▪ Statmux: 10-13 Mbps ▪ CBR: 18-25 Mbps 8k: ▪ CBR: 40-56 Mbps ▪ High quality: 80-90 Mbps [24][25][26]
EVC-Baseline	-30 % vs AVC [27][29]	n/a	Not expected for 4k or 8k
EVC-Main	-24 % vs HEVC [27][29]	n/a	Expected [27][29] 4k CBR: 15-19Mbps 8k CBR: 30-60 Mbps
VVC	-30 % vs HEVC [28][29] Best-CfP: -42 % vs HEVC Target: -50 % vs HEVC	n/a	Expected [28][29] 4k CBR: 10-15 Mbps 8k CBR: 25-35 Mbps
NOTE: (Average) with peaks up to 25Mbps thanks to STATMUX.			

Also noteworthy is the improvement of encoders over time even for existing standards which also leads to bitrate reductions at the same quality.

Based on this information it can be expected that within the time frame until 2025, video compression technology permit bitrate reductions by a factor of 50 % compared to what is today possible with HEVC [15].

However, not to forget according to Jevons Paradox, stating that the efficiency with which a resource is used tends to increase (rather than decrease) the rate of consumption of that resource. This may well mean that the compression efficiency gains spur even more traffic.

Additional updates to video chararacteristics and bitrates are available in 3GPP TR 26.955 [46].

6.1.3 New Media Formats

New media formats in media generation and distribution are developed on a continuous basis, taking into account improvements in capturing and display systems. TR 26.949 [16] collects TV distribution formats as they were considered in the time frame of 2015 to 2017. The parameters are summarized in clause 5.4 of the present document in terms of:

- Spatial Resolution
- Frame Rates
- Colorimetry
- Other parameters

The latest versions, referred to as UHD-1 Phase 2 is summarized according to TR 26.949, clause 5.4.8:

- UHD-1 phase 2 exclusively operates with HEVC [15] and bit depth of 10 bit.
- only square pixel resolutions and only progressive scan are supported.
- Only BT.2100 [33] non-constant luminance YCbCr is supported.
- High Dynamic Range is added either through the PQ10 or the HLG10 system, relying on HEVC Main-10 Level 5.1 [15].
- Optional SEI metadata may be provided in the bitstream using HEVC specified SEI messages [15].
- Addition of High Frame Rate (HFR), i.e. addition of 100, 120 000/1 001 and 120 Hz.
- HFR can be supported by two means:
  - Single PID HFR bitstream.
  - Dual PID and temporal scalability when the bitstream is intended to be decodable by UHD phase 1 receivers at half frame rate.
- For HFR on HEVC Main-10 Level 5.2 [15] is required.
- Support for Next Generation Audio (NGA) to enable immersive and personalized audio content.

The TV Video Profiles in TS 26.116 [5] address coded representations of the UHD-1 phase 2 signals to the most extent. Table 6.1.3-1 provides an overview of the TV relevant formats considered in the context of 3GPP TV Video Profiles.

Table 6.1.3-1: TV over 3GPP services Video Profile Operation Points (TS 26.116 [5])

Operation Point name	Resolution format	Picture aspect ratio	Scan	Max. frame rate	Chroma format	Chroma sub-sampling	Bit depth	Colour space format	Transfer Characteristics
H.264/AVC 720p HD	1280 × 720	16:9	Progressive	30	Y'CbCr	4:2:0	8	BT.709	BT.709
H.265/HEVC 720p HD	1280 × 720	16:9	Progressive	30	Y'CbCr	4:2:0	8	BT.709	BT.709
H.264/AVC Full HD	1920 × 1080	16:9	Progressive	60	Y'CbCr	4:2:0	8	BT.709	BT.709
H.265/HEVC Full HD	1920 × 1080	16:9	Progressive	60	Y'CbCr	4:2:0	8; 10	BT.709; BT.2020	BT.709; BT.2020
H.265/HEVC UHD	3840 × 2160	16:9	Progressive	60	Y'CbCr	4:2:0	10	BT.2020	BT.2020
H.265/HEVC Full HD HDR	1920 x 1080	16:9	Progressive	60	Y'CbCr	4:2:0	10	BT.2020	BT.2100 PQ
H.265/HEVC UHD HDR	3840 x 2160	16:9	Progressive	60	Y'CbCr	4:2:0	10	BT.2020	BT.2100 PQ

Operation Point name	Resolution format	Picture aspect ratio	Scan	Max. frame rate	Chroma format	Chroma sub-sampling	Bit depth	Colour space format	Transfer Characteristics
H.265/HEVC Full HD HDR HLG	1920 x 1080	16:9	Progressive	60	Y'CbCr	4:2:0	10	BT.2020	BT.2100 HLG
H.265/HEVC UHD HDR HLG	3840 x 2160	16:9	Progressive	60	Y'CbCr	4:2:0	10	BT.2020	BT.2100 HLG

Looking further into the future, there is currently only one 8K broadcast service being supported by the Japanese public service broadcaster NHK. This runs 12 hours a day and is to be utilised as part of the promotion for the upcoming Olympics in Japan. This service is supported by a government initiative. A summary of the service launched by NHK was published in [42] and a more detailed description can be found in the IBC paper "Ready for 8K UHDTV broadcasting in Japan" [41].

France Television performed a trial of 8K delivery of the Roland Garros Tennis Championship by 5G in June 2019, details can be found in [43].

For VR production and distribution, the ITU recommends [TBA] that for eye not to perceive pixels for 360VR, 30K by 15K images should be provided, though other criteria may be used in system design.

6.1.4 Protocol Improvements

Most of the video traffic used HTTP and TCP for the delivery of both wired and wireless Internet. The transmission control protocol, mostly known as TCP or TCP/IP [17], has been invented over 40 years ago. Over the years, it has been evolving steadily and it became the number transport protocol on the Internet and finally also on mobile networks. Today, around 75 % of the traffic is encrypted in mobile data networks. TCP, in combination with TLS, requires three round trips before the actual data can be sent. However, innovation in the context became very hard.

In response to these challenges with slow innovation of TCP, Internet companies started experimenting with proprietary protocols build on top of UDP. UDP is a very basic protocol. It only provides the bare minimum functionality and is suitable for new protocols to be built on top of it. It is well supported by all infrastructure on the Internet. Protocols on top of UDP can be implemented in applications so it allows for rapid deployment of new versions. The experiments with UDP protocols evolved into an effort to bring in the engineering community and openly collaborate on a single protocol framework on top of UDP. An Internet Engineering Task Force working group has been established to specify a protocol called QUIC (Quick UDP Internet Connections). Shortly it is likely to see the first QUIC Internet standard published [18].

The QUIC Protocol is the next generation of transport for the Internet and is on track to become ubiquitous on end-user platforms and within all server-side workflows. Unlike prior standard technologies, which are struggling to innovate due to compatibility issues with existing infrastructure, QUIC operates on top of UDP in the application layer and that brings flexibility to deploy new features in rapid iterations.

Google introduced QUIC back in 2013 as an experimental protocol to reduce TCP connection and transport latencies. QUIC, on the other hand, minimizes the number of set-up round trips by combining UDP transport and its own cryptographic handshake. For connections to the same origin server, QUIC facilitates a zero round trip time.

QUIC also re-implements TCP loss recovery, over UDP and, by using multiplexed connections, it eliminates TCP's head-of-line blocking. This ensures that lost packets do not block any other stream but only those with data in it. Last, but not least, QUIC moves congestion control to the application and the user space, enabling a rapid evolution for the protocol, as opposed to kernel space TCP.

QUIC has been nicely designed to allow being enabled seamlessly for existing video workflows without any changes needed in the video formats or players. Akamai reports as an example that for a measurement is from a soccer event in 2018 a 25 % throughput improvement for median viewer was achieved compared to TCP for desktop. QUIC performances gains for mobile compared to TCP/TLS, while still positive, have found not to be that significant as for desktop, in particular for video content.

There are several features of the protocol that contribute to improved video quality, among others:

- With usual TCP&TLS session which forms HTTPS streaming, there has to be 3 or 4 exchanges of information back and forth between the client and the server, before the video request is made. With QUIC there's none needed, assuming the client communicated with the server previously. The first packet from the client already contains the video request, and the first packet from the server already contains the video content. Or in case the client has not communicated with the server before, it only takes one roundtrip between the client and server. This reduces latency greatly. TCP has similar features available with new version of TLS, TLS-1.3, and TCP Fast Open feature but its deployment is complicated. The 0-RTT (zero round trip time) connection establishment for returning connections applies to about 50 % of connections.
- QUIC also uses more efficient loss recovery - due to more information being provided about the lost packets and timing.
- QUIC allows Multiplexing without head-of-line blocking, i.e. if one packet gets lost then all requests affected.
- QUIC also allows connection migration across different access networks.
- It also allows to measure the performance of each feature and use and apply it selectively for specific deployments.

On mobile QUIC benefits seem to also be tangible. On mobile Android devices, Google claims[i] that QUIC has helped to reduce latency of Google Search responses by 3.6 % and YouTube video buffering by 15.3 %. It was found that by November 2017, QUIC represented 20 % of the total mobile traffic and expected to grow to 35 % by end of 2018. According to MVI data, video accounts for approximately 58 % of the total mobile internet traffic and video represents 64 % of the total QUIC traffic. By November 2018, approximately 90 % of internet traffic will be encrypted and QUIC will be 32 % of global Internet traffic.

QUIC being an encryption-based protocol such as HTTP/TLS, traffic is typically encrypted over the mobile network delivery, meaning that the use of traditional traffic management tools is limited. On top of what is also observed for HTTP/TLS, QUIC allows multiplexing multiple streams over a single connection, but this comes with the added downside that it is impossible to differentiate between the different streams as the signalling header with the stream identifier is also encrypted.

Another technology and protocol in media distribution is developed under the umbrella of ABR Multicast. With the recent proliferation of live streaming, especially of premium sports, over the Internet, issues in scale and quality have been exposed. The challenge partially lies in scaling the services to millions of simultaneous users, but also in the associated peering and delivery costs as well as the end-to-end latency. Multicast ABR addresses a problem that had already been solved in the context of managed networks for IPTV with IGMP. However, it is quite difficult to replicate IGMP exactly over an unmanaged network. However, in practice it is possible to control delivery of streams quite precisely over a large part of their journey and multicast ABR effectively provides a tunnel through which multiple unicast streams are combined into single ones, just as with traditional multicast. Multiple unicast streams are converted at the entrance of the tunnel into a single stream and then back to multiple unicast as they exit in a process known as transcasting. These processes have since been documented either as a standard or guidelines towards a standard, by both the DVB [19] and by Cable Labs [20]. Products are being built around these guidelines, with many operators currently in proof of concept or field trials of the technology.]

6.1.5 Impact on Media Services

The impact of these new developments on Media Service is FFS.

7 Characteristics and requirements for different media services on 3GPP networks

7.1 Introduction

This clause collects characteristics and requirements for different media services on 3GPP networks.

7.2 Collection process for requirements for different media services on 3GPP networks

Use cases are collected in the context of the present document. The use cases come with an analysis of certain characteristics. For this purpose a template is provided below.

Media Service Description
Name: [e.g. StreamOn] **Date of submission:** [e.g. 18th October 2018] **Source:** [e.g. DT] **Description:** [Free text giving high level description of the service]
Categorization
Type: [e.g. Third Party, Operator, Combination of both, Other] **Status:** [e.g. planned, PoC, trialed, commercially deployed] **Category:** [e.g. Download, Live streaming, On-demand streaming, Interactive, Conversational]
Used Technologies
Codecs: [e.g. AAC, HE-AAC, HEVC/H.265, AVC/H.264 with Profiles and levels used] **Media types and formats:** [e.g. 4K video, HDR video, stereo or 5.1 audio] **Media protocols and containers:** [e.g. DASH, HLS, CMAF, ISOBMFF, MP4, MPEG2 TS, RTP] **Transport protocols:** [e.g. FLUTE, UDP, TCP, QUIC, HTTP 1.1, HTTP 2.0] **Clients:** [e.g. iOS or android application, browser, set top boxes, TV set] **Others:** [e.g. DRM, interactivity framework, QoE and analytics, QoS handling]
Deployment Statistics
[e.g. number of clients, traffic volumes]
Traffic Characteristics
Bitrate Characteristics: [typical target bitrates, minimum required bitrate, maximum necessary bitrate,] **Other KPIs** [end-to-end latency, tune-in times, etc.] **Potential mapping to 5QIs (5G QoS Identifier - See TS 23.501 Clause 5.7.4 and Table 5.7.4-1 copied below):** [which 5QIs among 1, 2, 3, 4, 6, 7, 8, 9, 80, e.g. 6, "video buffered streaming and TCP based"]
References
[list of public documentation in support of the present description e.g. a URL, a scientific publication, an industry or standards organization specification]
Any additional information

Table 7.2-1: Standardized 5QI to QoS characteristics mapping (from TS 23.501)

5QI Value	Resource Type	Default Priority Level	Packet Delay Budget	Packet Error Rate	Default Maximum Data Burst Volume (NOTE 2)	Default Averaging Window	Example Services
1	GBR (NOTE 1)	20	100 ms	10^{-2}	N/A	2000 ms	Conversational Voice
2		40	150 ms	10^{-3}	N/A	2000 ms	Conversational Video (Live Streaming)
3		30	50 ms	10^{-3}	N/A	2000 ms	Real Time Gaming, V2X messages Electricity distribution - medium voltage, Process automation - monitoring
4		50	300 ms	10^{-6}	N/A	2000 ms	Non-Conversational Video (Buffered Streaming)
65		7	75 ms	10^{-2}	N/A	2000 ms	Mission Critical user plane Push To Talk voice (e.g., MCPTT)
66		20	100 ms	10^{-2}	N/A	2000 ms	Non-Mission-Critical user plane Push To Talk voice
67		15	100 ms	10^{-3}	N/A	2000 ms	Mission Critical Video user plane
75		25	50 ms	10^{-2}	N/A	2000 ms	V2X messages
5	Non-GBR (NOTE 1)	10	100 ms	10^{-6}	N/A	N/A	IMS Signalling
6		60	300 ms	10^{-6}	N/A	N/A	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)
7		70	100 ms	10^{-3}	N/A	N/A	Voice, Video (Live Streaming) Interactive Gaming
8		80	300 ms	10^{-6}	N/A	N/A	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)
9		90					
69		5	60 ms	10^{-6}	N/A	N/A	Mission Critical delay sensitive signalling (e.g., MC-PTT signalling)
70		55	200 ms	10^{-6}	N/A	N/A	Mission Critical Data (e.g. example services are the same as QCI 6/8/9)
79		65	50 ms	10^{-2}	N/A	N/A	V2X messages
80		68	10 ms	10^{-6}	N/A	N/A	Low Latency eMBB applications Augmented Reality
82	Delay Critical GBR	19	10 ms (NOTE 4)	10^{-4}	255 bytes	2000 ms	Discrete Automation (see TS 22.261 [2])
83		22	10 ms (NOTE 4)	10^{-4}	1358 bytes (NOTE 3)	2000 ms	Discrete Automation (see TS 22.261 [2])

84		24	30 ms (NOTE 6)	10^{-5}	1354 bytes	2000 ms	Intelligent transport systems (see TS 22.261 [2])
85		21	5 ms (NOTE 5)	10^{-5}	255 bytes	2000 ms	Electricity Distribution-high voltage (see TS 22.261 [2])

NOTE 1: A packet which is delayed more than PDB is not counted as lost, thus not included in the PER.
NOTE 2: It is required that default MDBV is supported by a PLMN supporting the related 5QIs.
NOTE 3: This MDBV value is set to 1354 bytes to avoid IP fragmentation for the IPv6 based, IPSec protected GTP tunnel to the 5G-AN node (the value is calculated as in Annex C of TS 23.060 [56] and further reduced by 4 bytes to allow for the usage of a GTP-U extension header).
NOTE 4: A delay of 1 ms for the delay between a UPF terminating N6 and a 5G-AN should be subtracted from a given PDB to derive the packet delay budget that applies to the radio interface.
NOTE 5: A delay of 2 ms for the delay between a UPF terminating N6 and a 5G-AN should be subtracted from a given PDB to derive the packet delay budget that applies to the radio interface.
NOTE 6: A delay of 5 ms for the delay between a UPF terminating N6 and a 5G-AN should be subtracted from a given PDB to derive the packet delay budget that applies to the radio interface.

NOTE 1: For Standardized 5QI to QoS characteristics mapping, the table will be extended/updated to support service requirements for 5G, e.g. ultralow latency service.

NOTE 2: It is preferred that a value less than 64 is allocated for any new standardised 5QI of non-GBR Resource Type. This is to allow for option 1 to be used as described in clause 5.7.1.3 of TS 23.501 (as the QFI is limited to less than 64).

7.3 Summary of Responses for Streaming Services

7.3.1 Introduction

This clause provides a summary of the information that was collected for streaming services. The current information is based on responses from:

- Comcast VIPER
- AWS Media Services
- Hulu
- Bitmovin
- ARD Mediathek

The services address live and On-demand streaming services.

7.3.2 Used Technologies

7.3.2.1 Codecs

The pre-dominant video codec for the services is still H.264/AVC [14]. H.265/HEVC [15] is also in deployments. Non-MPEG video codecs such as VP9 or AV1 are in use or expected to be in use, but much less prominent than MPEG codecs according to the responses.

The pre-dominant audio codec is AAC, with some variants. Also, AC-3 is used quite often to support beyond stereo experiences. No other audio codec was mentioned.

7.3.2.2 Media types and formats

Most of the services run HD video. Experiments with UHD and HDR are ongoing. Different HDR formats (i.e. HDR10, Dolby Vision, HLG, and SL-HDR) are in use.

For audio, stereo is deployed and 5.1 is also broadly used. Some initial experiments are done with multichannel audio, NGA (Next Generation Audio).

7.3.2.3 Media protocols and containers

All services deploy DASH and HLS. On HLS, the MPEG-2 TS still is maintained. For DASH, exclusively ISO BMFF based distribution is used but a convergence towards a common segment format based on CMAF for DASH and HLS is expected.

Other formats such as RTMP or RTP are maintained to some extent for legacy reasons.

7.3.2.4 Transport protocols

HTTP/1.1 with TCP/IP is used almost exclusively. Some initial tests are ongoing on HTTP/2.0 and HTTP/3 (also known as HTTP over QUIC). No received response indicates the use of multicast protocols.

7.3.2.5 Clients

All services provide client application running on iOS and Android for Browser-based playback as well as applications or native integration into set-top boxes, HDMI sticks (i.e. Apple TV™, Fire TV™, Chromecast™, Tizen™) and TV Sets. Game consoles are also targeted.

7.3.2.6 Other Technologies

Other mentioned deployed technologies are:

- DRM (Widevine, PlayReady, Fairplay) based on common encryption
- Analytics collection and QoE Measurement
- Dynamic Ad Insertion
- Watermarking

7.3.3 Traffic Characteristics

7.3.3.1 Bitrate Characteristics

Average streaming bitrate

- STBs: 4-6Mbps
- Mobile applications and browsers: less than 4-6Mbps
- 4K content: 10-15Mbps

Average segment durations

- 6 seconds

7.3.3.2 Other KPIs

Other KPIs such as video start time, video playback failure, rebuffering ratio, connection induced rebuffering, minimum/average bitrates, ABR adaptation frequency were not reported.

7.3.3.3 Potential mapping to 5QIs

Commonly targeted 5QI values are 6 and 8. Others are considered interesting for specific services.

8 Applicability of existing 5Qis

8.1 QoS Model

8.1.1 Overview

Clause 5.7 of TS 23.501 [10] explains the QoS Model for 5G. The 5G QoS model is based on QoS Flows. The 5G QoS model supports both QoS Flows that require guaranteed flow bit rate (GBR QoS Flows) and QoS Flows that do not require guaranteed flow bit rate (Non-GBR QoS Flows). The 5G QoS model also supports Reflective QoS (see clause 5.7.5 of TS 23.501 [10]).

A QoS Flow ID (QFI) is used to identify a QoS Flow in the 5G System. User Plane traffic assigned to the same QoS Flow within a PDU Session receives the same traffic forwarding treatment (e.g. scheduling, admission threshold).

The QFI may be dynamically assigned or may be equal to the 5QI, for more details on existing 5QI see clause 8.1.2.

A QoS Flow may either be 'GBR', 'Non-GBR' or "Delay Tolerant GBR" depending on its QoS profile and it contains QoS parameters as follows:

- For each QoS Flow, the QoS profile includes the QoS parameters:
  - 5G QoS Identifier (5QI); and
  - Allocation and Retention Priority (ARP).
- For each Non-GBR QoS Flow only, the QoS profile can also include the QoS parameter:
  - Reflective QoS Attribute (RQA).
- For each GBR QoS Flow only, the QoS profile also include the QoS parameters:
  - Guaranteed Flow Bit Rate (GFBR) - UL and DL; and
  - Maximum Flow Bit Rate (MFBR) - UL and DL; and
- In the case of a GBR QoS Flow only, the QoS profile can also include one or more of the QoS parameters:
  - Notification control;
  - Maximum Packet Loss Rate - UL and DL

The usage of a dynamically assigned 5QI for a QoS Flow requires in addition the signalling of the complete 5G QoS characteristics (described in clause 5.7.3 of TS 23.501 [10]) as part of the QoS profile.

The principle for classification and marking of User Plane traffic and mapping of QoS Flows to Access Network (AN) resources is illustrated in Figure 8-1-1.

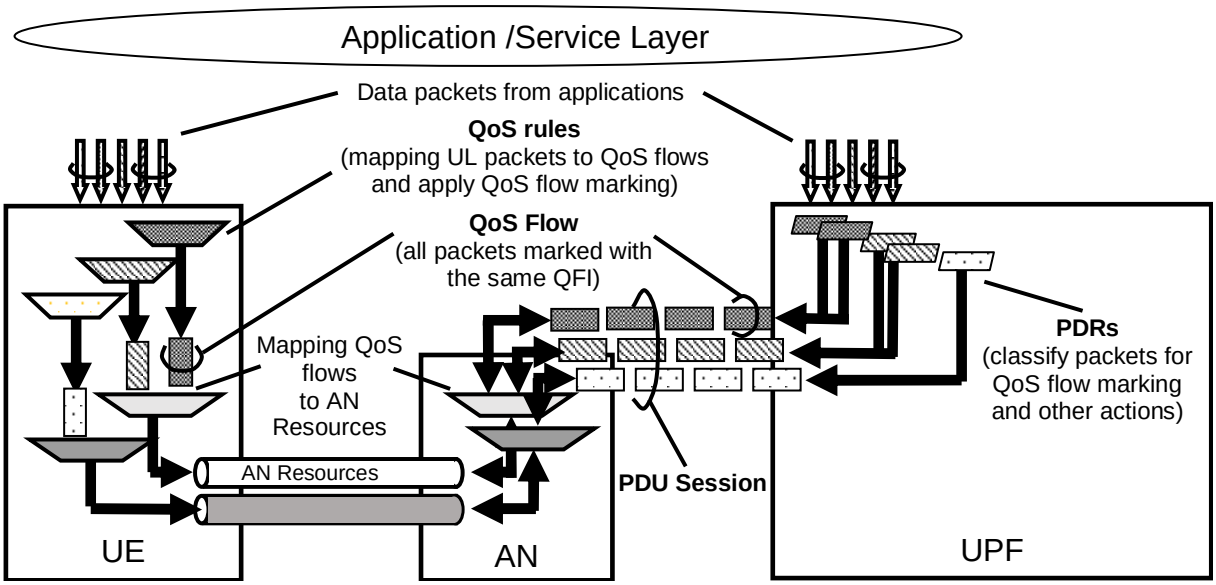

Figure 8.1-1: The principle for classification and User Plane marking for QoS Flows and mapping to AN Resources

8.1.2 5G QoS Parameters

A 5QI is a scalar that is used as a reference to 5G QoS characteristics defined in clause 5.7.4 of TS 23.501 [10], i.e. access node-specific parameters that control QoS forwarding treatment for the QoS Flow (e.g. scheduling weights, admission thresholds, queue management thresholds, link layer protocol configuration, etc.).

Standardized 5QI values have one-to-one mapping to a standardized combination of 5G QoS characteristics as specified in Table 5.7.4-1 of TS 23.501 [10] and shown below in Table 8.1.4-1.

A summary of the most relevant QoS Parameters is provided as follows. For details, please refer to TS 23.501 [10], clause 5.7.2:

- Guaranteed Flow Bit Rate (GFBR) - UL and DL, for GBR QoS Flows only: denotes the bit rate that is guaranteed to be provided by the network to the QoS Flow over the Averaging Time Window.
- Maximum Flow Bit Rate (MFBR) -- UL and DL, for GBR QoS Flows only limits the bit rate to the highest bit rate that is expected by the QoS Flow (e.g. excess traffic may get discarded or delayed by a rate shaping or policing function).
- The Maximum Packet Loss Rate (UL, DL) indicates the maximum rate for lost packets of the QoS flow that can be tolerated in the uplink and downlink direction, if the flow is compliant to the GFBR and is only used for voice media traffic in rel.16.

8.1.3 5G QoS Characteristics

In addition to 5G QoS parameters, also 5G QoS characteristics are defined in TS 23.501 [10]. The characteristics describe the packet forwarding treatment that a QoS Flow receives edge-to-edge between the UE and the UPF in terms of the following performance characteristics:

1) Resource Type (GBR, Delay critical GBR or Non-GBR): determines if dedicated network resources related to a QoS Flow-level Guaranteed Flow Bit Rate (GFBR) value are permanently allocated (e.g. by an admission control function in a radio base station).
2) Priority level; indicates a priority in scheduling resources among QoS Flows.
3) Packet Delay Budget; defines an upper bound for the time that a packet may be delayed between the UE and the UPF that terminates the N6 interface. In uncongested scenarios, 98 percent of the packets are expected to be within the packet delay budget.

4) Packet Error Rate: The Packet Error Rate (PER) defines an upper bound for the rate of PDUs (e.g. IP packets) that have been processed by the sender of a link layer protocol (e.g. RLC in RAN of a 3GPP access) but that are not successfully delivered by the corresponding receiver to the upper layer (e.g. PDCP in RAN of a 3GPP access).

5) Averaging window (for GBR and Delay-critical GBR resource type only); The Averaging window represents the duration over which the GFBR and MFBR are calculated.

6) Maximum Data Burst Volume (for Delay-critical GBR resource type only): denotes the largest amount of data that the 5G-AN is required to serve within a period of 5G-AN PDB.

Standardized or pre-configured 5G QoS characteristics, are indicated through the 5QI value. Signalled 5G QoS characteristics are provided as part of the QoS profile and include all of the characteristics listed above.

8.1.4 Standardized 5QI to QoS characteristics mapping

The one-to-one mapping of standardized 5QI values to 5G QoS characteristics is specified in table 5.7.4-1 if TS 23.501 [10] and shown below in Table 8.1.4-1.

Table 8.1.4-1: Standardized 5QI to QoS characteristics mapping (identical to Table 5.7.4.1-1 in TS 23.501 [10])

5QI Value	Resource Type	Default Priority Level	Packet Delay Budget	Packet Error Rate	Default Maximum Data Burst Volume (NOTE 2)	Default Averaging Window	Example Services
1	GBR (NOTE 1)	20	100 ms	10^{-2}	N/A	2000 ms	Conversational Voice
2		40	150 ms	10^{-3}	N/A	2000 ms	Conversational Video (Live Streaming)
3		30	50 ms	10^{-3}	N/A	2000 ms	Real Time Gaming, V2X messages Electricity distribution - medium voltage, Process automation - monitoring
4		50	300 ms	10^{-6}	N/A	2000 ms	Non-Conversational Video (Buffered Streaming)
65		7	75 ms	10^{-2}	N/A	2000 ms	Mission Critical user plane Push To Talk voice (e.g., MCPTT)
66		20	100 ms	10^{-2}	N/A	2000 ms	Non-Mission-Critical user plane Push To Talk voice
67		15	100 ms	10^{-3}	N/A	2000 ms	Mission Critical Video user plane
75		25	50 ms	10^{-2}	N/A	2000 ms	V2X messages
5	Non-GBR (NOTE 1)	10	100 ms	10^{-6}	N/A	N/A	IMS Signalling
6		60	300 ms	10^{-6}	N/A	N/A	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)
7		70	100 ms	10^{-3}	N/A	N/A	Voice, Video (Live Streaming) Interactive Gaming
8		80	300 ms	10^{-6}	N/A	N/A	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)
9		90					
69		5	60 ms	10^{-6}	N/A	N/A	Mission Critical delay sensitive signalling (e.g., MC-PTT signalling)
70		55	200 ms	10^{-6}	N/A	N/A	Mission Critical Data (e.g. example services are the same as QCI 6/8/9)
79		65	50 ms	10^{-2}	N/A	N/A	V2X messages
80		68	10 ms	10^{-6}	N/A	N/A	Low Latency eMBB applications Augmented Reality
82	Delay Critical GBR	19	10 ms (NOTE 4)	10^{-4}	255 bytes	2000 ms	Discrete Automation (see TS 22.261 [2])
83		22	10 ms (NOTE 4)	10^{-4}	1358 bytes (NOTE 3)	2000 ms	Discrete Automation (see TS 22.261 [2])
84		24	30 ms (NOTE 6)	10^{-5}	1354 bytes	2000 ms	Intelligent transport systems (see TS 22.261 [2])

85		21	5 ms (NOTE 5)	10^{-5}	255 bytes	2000 ms	Electricity Distribution-high voltage (see TS 22.261 [2])

NOTE 1: A packet which is delayed more than PDB is not counted as lost, thus not included in the PER.
NOTE 2: It is required that default MDBV is supported by a PLMN supporting the related 5QIs.
NOTE 3: This MDBV value is set to 1354 bytes to avoid IP fragmentation for the IPv6 based, IPSec protected GTP tunnel to the 5G-AN node (the value is calculated as in Annex C of TS 23.060 [56] and further reduced by 4 bytes to allow for the usage of a GTP-U extension header).
NOTE 4: A delay of 1 ms for the delay between a UPF terminating N6 and a 5G-AN should be subtracted from a given PDB to derive the packet delay budget that applies to the radio interface.
NOTE 5: A delay of 2 ms for the delay between a UPF terminating N6 and a 5G-AN should be subtracted from a given PDB to derive the packet delay budget that applies to the radio interface.
NOTE 6: A delay of 5 ms for the delay between a UPF terminating N6 and a 5G-AN should be subtracted from a given PDB to derive the packet delay budget that applies to the radio interface.

8.1.5 Considerations for Media Services

8.1.5.1 General

For media traffic considered in the present document, it is appropriate to understand, how they map to QoS parameters and characteristics, including the mapping to the 5QIs as indicated above.

8.1.5.2 Expected TCP/IP Performance for non-GBR

Maximum achievable throughput for a single TCP connection is determined by different factors. One trivial limitation is the maximum bandwidth of the slowest link in the path. But there are also other, less obvious limits for TCP throughput. Packet loss can create a limitation for the connection as well as can the roundtrip time of acknowledgements.

The TCP Receive Window is the amount of data that a receiver can accept without acknowledging the sender. If the sender has not received acknowledgement for the first packet it sent, it will stop and wait and if this wait exceeds a certain limit, it may even retransmit. This is how TCP achieves reliable data transmission. Even if there is no packet loss in the network, this windowing limits throughput. Because TCP transmits data up to the window size before waiting for the acknowledgements, the full bandwidth of the network may not always get used. At any given time, the window advertised by the receive side of TCP corresponds to the amount of free receive memory it has allocated for this connection.

When packet loss occurs in the network, an additional limit is imposed on the connection. In the case of light to moderate packet loss when the TCP rate is limited by the congestion avoidance algorithm.

The *Mathis equation* is a formula that approximates the actual impact of loss on the maximum throughput rate:

$$\text{Max Rate in bps} < (\text{MSS}/\text{RTT})*(1 / \sqrt{p})$$

where

MSS = maximum segment size in bytes

RTT = round trip time in seconds

p = the probability of packet loss

NOTE: *The formula is known as the Mathis equation given in [40] from a 1997 paper titled* The Macroscopic Behavior of the TCP Congestion Avoidance Algorithm [40].

While this equation obviously has limits in the details, it provides an excellent estimate on the estimated TCP throughput. Figure 8.1.5-1 shows the estimated TCP Throughput over One Way Latency for different packet loss rates and MSS 1500 bytes based on this equation, assuming that the RTT is twice the one way latency.

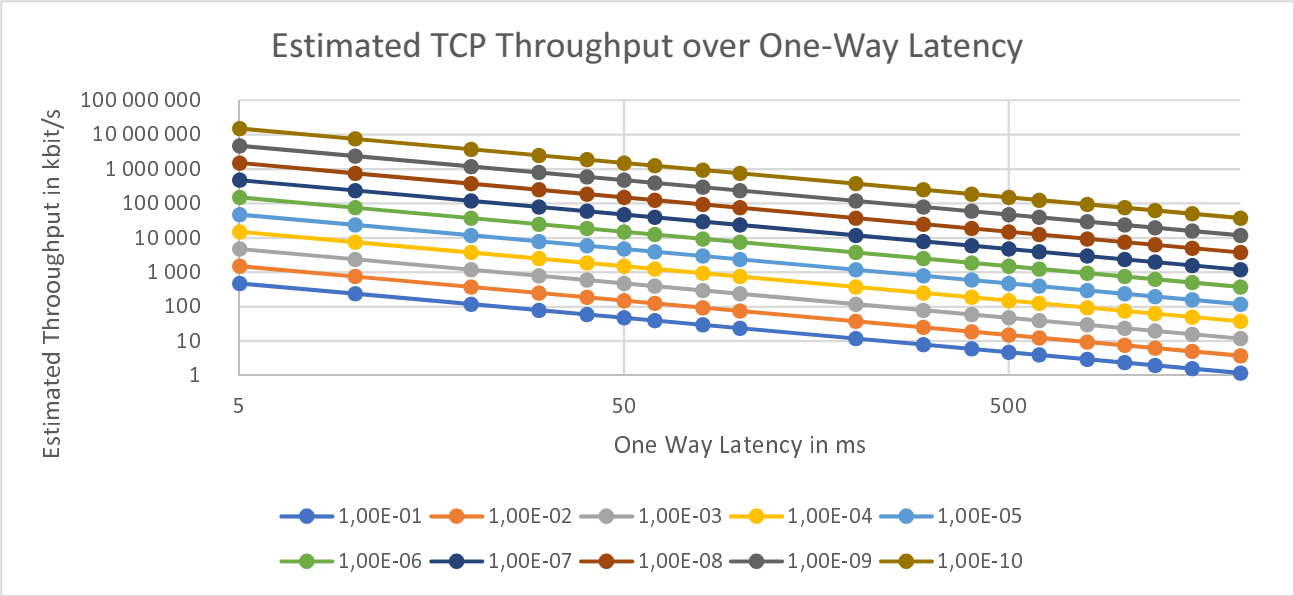

Figure 8.1.5-1: Estimated TCP Throughput over One Way Latency for different packet loss rates and MSS 1500 bytes

Based on this equation, the estimated TCP throughput for some relevant streaming related Non-GBR 5QIs as documented in Table 8.1.4-1 are as follows:

- 5QI = 7 with Latency=100ms and Packet loss rate 10e-3 results in estimated TCP Throughput of 237 kbit/s
- 5QI = 8 and 9 with Latency=300ms and Packet loss rate 10e-6 results in estimated TCP Throughput of 2.5 Mbit/s
- 5QI = 7 with Latency=10ms and Packet loss rate 10e-6 results in estimated TCP Throughput of 75 Mbit/s

The equation shows that typically TCP based streaming traffic is more susceptible to loss rates than to delay. It is also important that packet loss rates and latencies are not guaranteed and may therefore result in lower or higher throughput, for example in case of congestion or cell handoff. Hence, media services using these 5QI assignments preferably apply protocols that enable to adapt to these bitrates and changes.

8.1.5.3 Bitrate considerations for GBR services

If the service would be able to benefit from GBR QoS, then the GFBR and MFBR are relevant. A suitable characterization for a media service is the required minimum bitrate that it needs in order to maintain a good service quality. At the same time, a service is well characterized up to which bitrate it would still provide noticeable quality improvements. Such information may be static or may even change over time, depending on the complexity of the service.

8.1.5.4 Relevant Parameters for Media Services

Based on this discussion, it is proposed that for Media services that use TCP-based distribution systems and permit rate adaptation (such as adaptive streaming services), the following information is worthwhile to be provided:

- The typical bitrate at which the service preferable operates, possibly providing a range of the bitrate. It should also be mentioned if such information is static over the service or would change, and if such information may be configurable.
- If there exists a minimum bitrate that the service should not fall below to maintain a sufficient quality. If this exists, a range would be preferable. It should also be mentioned if such information is static over the service or would change, and if such information may be configurable.
- If there exists a maximum bitrate that the service beyond which the services does not created additional quality. If this exists, a range would be preferable. It should also be mentioned if such information is static over the service or would change, and if such information may be configurable.

Annex A:
Background information on cloud gaming

Interactive online game applications require connectivity between the game application client and a game application server. The game application client typically runs on a UE or on a device attached to a UE. The type of traffic generated depends on the type of game. The Quality of Experience is influenced by the adequacy of the QoS of the connectivity and the traffic requirements from the game application.

For example, Ref. [23] provides delay thresholds per game type in table A-1:

Table A-1: Delay tolerance in traditional gaming (from [23]).

Example game type	Perspective	Delay threshold
First person shooter (FPS)	First person	100 ms
Role playing game (RPG)	Third person	500 ms
Real-time strategy (RTS)	Omnipresent	1000 ms

NOTE: The delay tolerance figures of Table A-1. are applicable to regular 2D games and do not take into account any motion-to-photon requirements of VR gaming applications.

Cloud gaming implies that, while the game is being played by a user on a client, game application processing and rendering is totally or partly performed in a remote network entity, potentially at the edge of the network. The rendered 2D or VR360 video is then encoded and streamed to the client. These extra steps in game workflow influence the overall interactive delay and audio/video quality (e.g. encoding type/profile/level and performances, resolution and frame rate), and hence influence the Quality of Experience.

There is work in ITU-T Study Group 12 regarding assessment of gaming quality. Two recommendations are published:

- ITU-T G.1032 "Influence factors on gaming quality of experience" (https://www.itu.int/rec/T-REC-G.1032-201710-I/en) [38]
- ITU-T P.809 "Subjective evaluation methods for gaming quality" (https://www.itu.int/rec/T-REC-P.809/en). [39]

This last recommendation could be used to produce comparable subjective evaluations of the influence of connectivity characteristics over QoE.

Clause 5.3.1 of [22] discusses cloud gaming requirements. It states that "Taking out the delay for rendering and encoding/decoding processing, the round-trip time (RTT) delay over 5GS should be less than 5 ms".

Cloud gaming services already exist, and new ones are being launched, enabling users to play any type of game (whether it is originally offline or online, single player or multiplayer etc.) on a device like a lite smartphone, tablet or TV which would typically be unable to process such a game. The following is a brief look at several of these services and the public information about their requirements.

Cloud gaming typical traffic characteristics are evolving as services are being deployed. But a tendency can be derived from the data available. Bitrates in the range of 5-35Mbps are expected. Although latency requirements vary from game types and users, cloud gaming services are expected to offer a Quality of Experience which is at least as good as the experience of regular (locally processed) gaming.

Annex B:
Change history

Change history							
Date	**Meeting**	**TDoc**	**CR**	**Rev**	**Cat**	**Subject/Comment**	**New version**
2019-12	SA#86	SP-190993				Provided to TSG SA#86 (for information)	1.0.0
2020-01	MBS SWG Ad hoc telco 31/01/2020	S4-AHI928				Editor's compilation of S4-200037, S4-200319, S4-200011, S4-200067 in S4-AHI927 and online edits during call.	1.1.0
2020-03	SA#87-e	SP-200048				Provided to TSG SA#87-e (for approval)	2.0.0
2020-03	SA#87-e	SP-200048				Approved by TSG SA#87-e	16.0.0
2021-09	SA#93-e	SP-210820	0001	01	B	Typical Traffic Characteristics	17.0.0
2022-03	SA#95-e	SP-220032	0002		B	8K TV Traffic Characteristics	17.1.0
2023-09	SA#101	SP-230922	0003	4	B	XR Traffic Characteristics	18.0.0
2023-12	SA#102	SP-231372	0004		B	[FS_XRTraffic] Application Layer FEC Traffic characteristics	18.1.0

History

Document history		
V18.1.0	May 2024	Publication