4G/5G Network Experience Evaluation Guideline

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Executive Summary

This document describes main mobile broadband (MBB) services in the 5G era, such as video, virtual reality (VR), and mobile gaming, and analyses their experience requirements. Through simulation and testing, this document provides mapping from service experience to air-interface indicators such as the LTE reference signal received power (RSRP), signal to interference plus noise ratio (SINR), and traffic load. The purpose of this document is to guide mobile operators through network deployment and optimisation based on service experience.

Additionally, this document elaborates on methods for evaluating service experiences, such as traffic counters and measurement report (MR) grids. These methods can intuitively indicate experience indicators for operators to proactively identify exceptions and perform network optimisation accordingly.

1. Introduction

The 4G era witnessed the quick popularisation of MBB Internet and smart terminals, as well as the rapid development of MBB applications such as mobile video streaming, mobile gaming, and mobile payments. In the 5G era, immersive mobile experiences provided by VR and augmented reality (AR) technologies have become a new direction for development, while traditional applications (typically mobile gaming and video) will require higher definition and lower latency. Given the fact that LTE will continue to be the main bearer network for MBB services in the 5G era, the user experience will change across the services that impose higher requirements on LTE networks.

User experience directly affects user loyalty and operator revenues. It is generally acknowledged that experience is crucial in guiding network planning and optimisation. However, service experience of video, gaming, and VR cannot be directly perceived on the network side, and is often reflected only in complaints from users or over-the-top (OTT) content providers. It is difficult for operators to perform network planning and optimisation based on Service Level Agreement (SLA) requirements.

This document describes typical services in the 5G era and the main factors that affect service experience, by associating service experience first with network requirements (such as throughput and latency) and finally with air interface indicators which are useful for network planning and optimisation (such as LTE RSRP, SINR, and load).

This document also presents quantitative and visualised methods for evaluating service experience. These methods provide information about network experience satisfaction and geographical location exceptions, thereby effectively guiding network optimisation.

1.1. Abbreviations

Term	Description
ACT	Action
AMR-NB	Adaptive Multi-Rate Narrowband
AMR-WB	Adaptive Multi-Rate Wideband
AR	Augmented Reality
AVG	Adventure Game
CDN	Content Distribution Network
CQI	Channel Quality Indicator
CSFB	Circuit-Switched Fall-back
DASH	Dynamic Adaptive Streaming over HTTP
dB	Decibel
DNS	Domain Name Server
EIRP	Equivalent Isotropically Radiated Power
eNB	Evolved Node B
E-RAB	E-UTRAN Radio Access Bearer
EVS	Enhanced Voice Services
FOV	Field of View
FPS	First Person Shooter
FTG	Fighting Game
GPS	Global Positioning System
HMD	Head Mounted Display
IMS	IP Multimedia Subsystem
ITU-T	International Telecommunication Union-Telecommunication Standardization Sector
KPI	Key Performance Indicator
KQI	Key Quality Indicator
LTE	Long-Term Evolution
MBB	Mobile Broadband
MOS	Mean Opinion Score
MR	Measurement Report
MTP	Motion to Photons
MUG	Music Game
OTT	Over the Top
PDCP	Packet Data Convergence Protocol
PESQ	Perceptual Evaluation of Speech Quality
POLQA	Perceptual Objective Listening Quality Analysis
PPD	Pixel Per Degree
PRB	Physical Resource Block
PUZ	Puzzle Game

Term	Description
QCI	QoS Class Identifier
QoS	Quality of Service
RAC	Racing Game
RAN	Radio Access Network
RF	Radio Frequency
RLC	Radio Link Control
RoHC	Robust Header Compression
RPG	Role-Playing Game
RRC	Radio Resource Control
RS	Reference Signal
RSRP	Reference Signal Received Power
RTS	Real-Time Strategy Game
RTT	Round-Trip Time
SD	Standard Definition
SINR	Signal to Interference plus Noise Ratio
SLA	Service Level Agreement
SLG	Simulated Life Game
SPG	Sports Game
STG	Shooting Game
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TTI	Transmission Time Interval
UDP	User Datagram Protocol
VoLTE	Voice over Long Term Evolution
VR	Virtual Reality
YOY	Year on Year

2. Typical Services and Experience Requirements

2.1. Video Playback

Video has become a primary source of mobile network traffic, and video experience is a key indicator for network quality of service (QoS). Video services place higher requirements on networks as video resolution increases (from 360p/720p to 1080p/2K), therefore experience assurance is particularly important for operators.

2.1.1. Key Factors to Video Experience

A complete video service includes buffering, download, and playback. User experience can also be affected by video quality (definition), initial buffering delay, and frame freezing (smoothness).

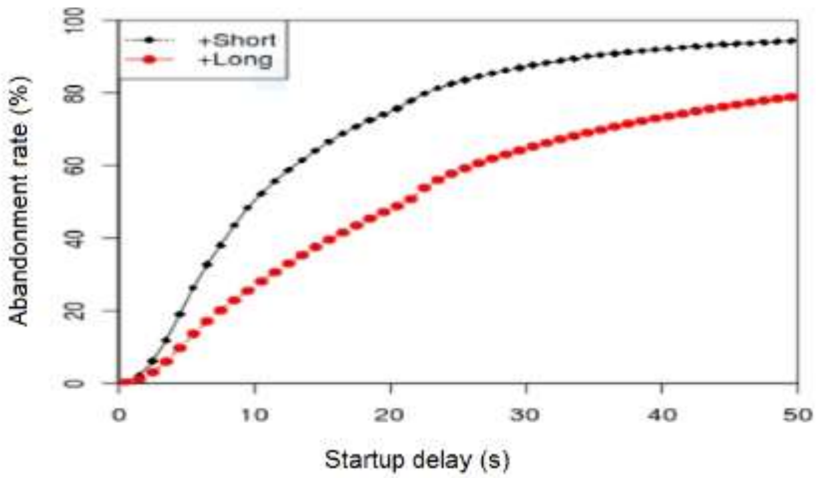
Video quality: the quality of video is determined by resolution and bit rate. The higher the resolution and bit rate, the clearer the image. Generally, video quality is more affected by an OTT content provider's video sources and user behaviour than by the mobile network. A higher video resolution or bit-rate requires a higher download rate (bandwidth).

The following table lists the typical average bit rates of YouTube mobile videos.

Video Source	Image Quality	Resolution	Typical Bit Rate (Mbit/s)		Video Encoding (Profile and Level)
			H264	VP9	
YouTube	360p	640 x 360	0.35	0.3	H.264 Main
	480p	854 x 480	0.66	0.55	H.264 Main
	720p	1280 x 720	1.3	1.1	H.264 Main
	1080p	1920 x 1080	2.5	2	H.264 High
	2K	2560 x 1440	6	6	H.264 High
	4K	3840 x 2160	15.5	15.5	H.264 High

Initial buffering delay: The time a user has to wait before a video is played, namely the initial play delay. It includes the time it takes to access the video service (through DNS enquiry, TCP link setup, and directory download) and for the initial buffering and download.

According to a research report by UMass, some users abandon a video if the initial buffering delay exceeds two seconds, and this user percentage increases by 5.8% for every one-second increase. Users are especially intolerant of delays for short videos.



NOTE

Short videos: < 30 minutes (e.g. a news clip); median duration: 1.8 minutes

Long videos: ≥ 30 minutes (e.g. a movie); median duration: 43.2 minutes

Frame freezing: Frame freezing occurs during video playback and severely affects user experience. The freezing-free ratio helps measure video experience and is calculated using the following formula: Samples of video call data records (CDRs) without frame freezing/total samples of video CDRs.

The standard video experience consists of an initial buffering delay of fewer than three seconds and zero frame freezing. If the initial buffering delay is one second or less, and no frame freezing occurs, video experience is considered excellent.

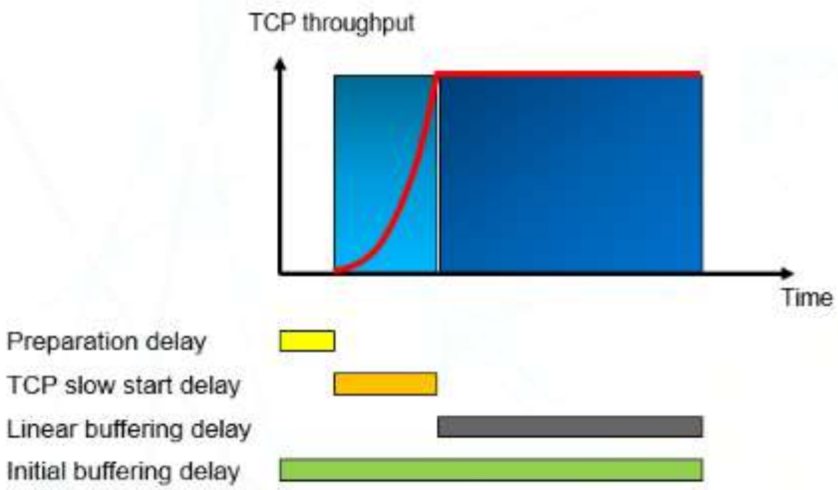
2.1.2. Video Experience Requirements on Networks

Video service falls into two stages, namely initial buffering and playback, which vary in terms of rate requirements.

Initial buffering phase

Initial buffering delay = Initial buffering preparation delay + Initial download delay

During the initial buffering preparation (including DNS enquiry and TCP link setup), the OTT interaction mechanism and transmission path round-trip time (RTT) are the main factors that affect the experience. In the initial buffering and download phase, influencing factors include initial buffering volume thresholds, transmission path RTT, and TCP download rates (bandwidth). The initial buffering and download phase is short (usually a few seconds or even hundreds of milliseconds), and therefore has high requirements on RTT and short-term burst bandwidth.



The following table maps initial buffering rate, video resolution, RTT, and initial buffering delay of YouTube.

Resolution	RTT (ms)	Initial Buffering Duration (s)	Required Rate (Mbit/s)
720p	40	3	2.13
720p	40	2	3.498
720p	40	1	10.419
720p	50	3	2.221
720p	50	2	3.782
720p	50	1	14.645
720p	60	3	2.327
720p	60	2	4.117
720p	60	1	30.246
1080p	40	3	4.113
1080p	40	2	6.818
1080p	40	1	21.48
1080p	50	3	4.308
1080p	50	2	7.442
1080p	50	1	32.763
1080p	60	3	4.535
1080p	60	2	8.192
1080p	60	1	62.757
2K	40	3	9.998
2K	40	2	16.83
2K	40	1	57.978
2K	50	3	10.542

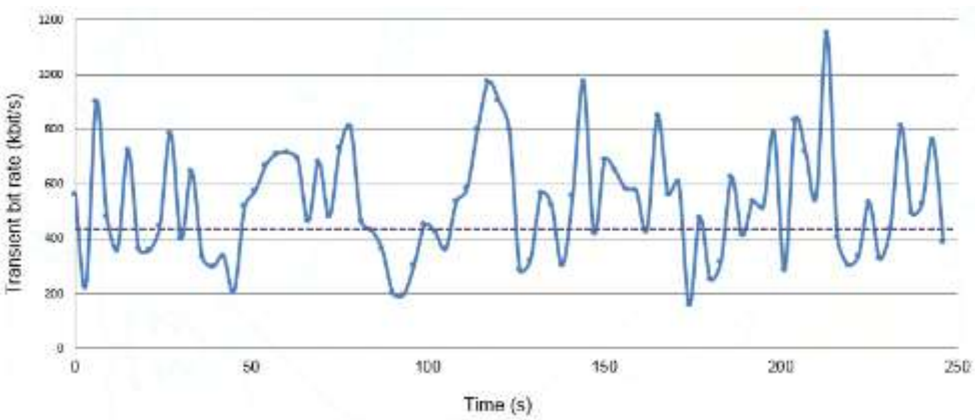
Resolution	RTT (ms)	Initial Buffering Duration (s)	Required Rate (Mbit/s)
2K	50	2	18.558
2K	60	3	11.194
2K	60	2	20.91

NOTE

The above table uses YouTube as the video source, H.264 video encoding, the DASH player, and a video buffer size of 4s.

Video playback phase

In this phase, the video bitrate is not a constant value and fluctuates around an average, as shown in the following figure. When the video image is rich in detail and frequently switched, the transient bit rate is high. If the download rate is lower than the video bit rate, frame freezing may occur.



According to a test (using YouTube DASH player with 1080p and 720p videos) on a commercial network, the freezing-free ratio reached 90%, 95%, and 98% when download rates were 1.5, 1.7, and 2.0 times the average video bit rate respectively.

Whole-process Perceived Rate/Average Bit Rate	Samples Without Frame Freezing	Samples with Frame Freezing	Samples in Test Period	Samples Without Frame Freezing in Test Period (%)
< 0.9	125	1,708	1,833	6.82
0.9–1.0	235	328	563	41.74
1.0–1.1	334	256	590	56.61
1.1–1.2	378	175	553	68.35
1.2–1.3	731	111	842	86.82
1.3–1.4	455	69	524	86.83
1.4–1.5	488	57	545	89.54
1.5–1.6	532	48	580	91.72
1.6–1.7	668	54	722	92.52
1.7–1.8	612	26	638	95.92
1.8–1.9	678	26	704	96.31

Whole-process Perceived Rate/Average Bit Rate	Samples Without Frame Freezing	Samples with Frame Freezing	Samples in Test Period	Samples Without Frame Freezing in Test Period (%)
1.9–2.0	668	33	701	95.29
2.0–2.1	1,962	30	1,992	98.49
> 2.1	50,401	503	50,904	99.01

The following table displays the speed required to ensure a freezing-free ratio is greater than 98% (twice the bit rate) on YouTube.

Resolution	Speed for a 98% Freezing-free Ratio (Mbit/s)
720p	2.6
1080p	5
2K	12

The preceding analysis reveals that the initial buffering phase is short, yet requires a high network speed and good network coverage. The video playback phase, however, has relatively relaxed network rate requirements but calls for a large capacity due to long duration and high probability of simultaneous multi-user access.

2.1.3. Video Experience Requirements on RF Performance

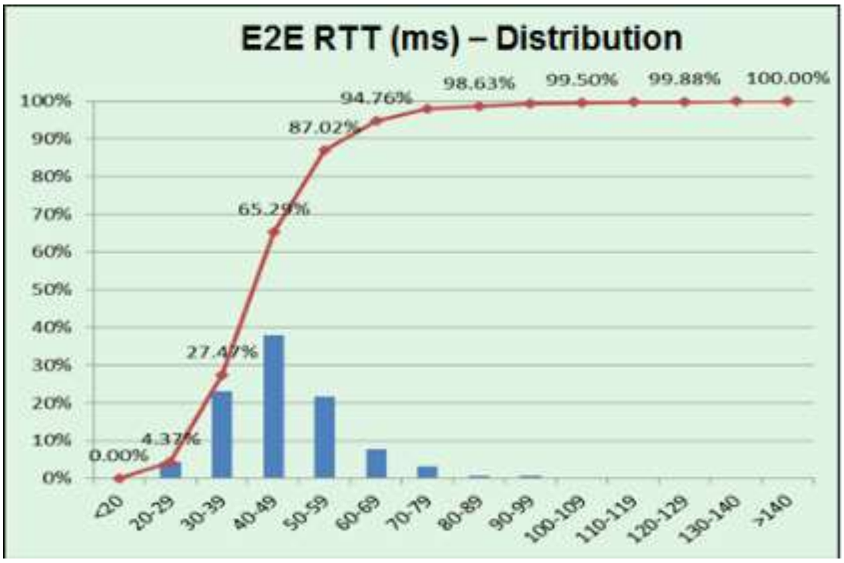
This section describes how to associate network capabilities with air interface capabilities in network planning.

Data throughput and target RTT selection

End-to-end (E2E) RTT and TCP data throughput are complementary, and therefore their target values should be considered equally during network planning.

Generally, network infrastructure optimisation is required to shorten E2E RTT, which is both time-consuming and costly. Additionally, some paths and nodes are beyond the operator’s control and cannot be optimised. For these reasons, a short and feasible target E2E RTT is usually determined first, in order to serve as the basis to determine a target TCP data throughput.

If the optimisation of network infrastructure or content delivery network (CDN) is not used, the target E2E RTT is determined based on real values on live networks. Considering RTT fluctuation, the planned E2E RTT should correspond to a 90% fulfilment ratio.



If RTT optimisation is required during network planning, the after-optimisation target RTT is used as the planned target RTT.

Once the RTT is determined, video experience and TCP data throughput mapping can be achieved. Considering a TCP and IP header overhead of about 2.7% (which is negligible, and calculated based on a 40-byte TCP/IP header and a payload of 1,460), the following equation is true: PDCP throughput over the air interface = TCP data throughput x (100% + 2.7%).

PDCP throughput and SINR/RSRP/load mapping

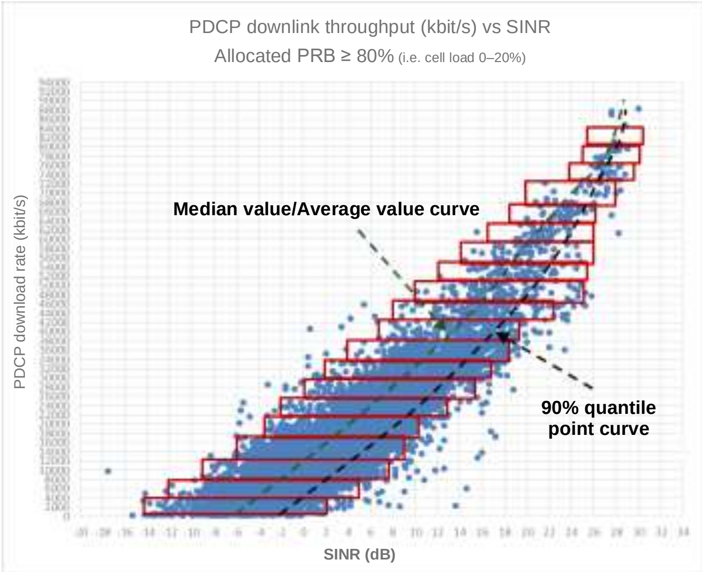
Based on the target PDCP throughput achieved in the previous step, we can map it to network RF planning parameters. Due to variations in the radio environment, equipment capability, and software implementation, theoretically there is no accurate mapping between PDCP throughput and RF planning target SINR, RSRP, and physical resource block (PRB) usage. In actual engineering applications, a large number of live network test samples are analysed to approximate a mapping model between PDCP throughput and SINR/RSRP in different networks, ground objects, and load scenarios. Model accuracy is positively correlated to the test scenario, sample quantity, and sample distribution. The more test scenarios, the larger the sample quantity, and the more comprehensive the sample distribution, the more accurate the model.

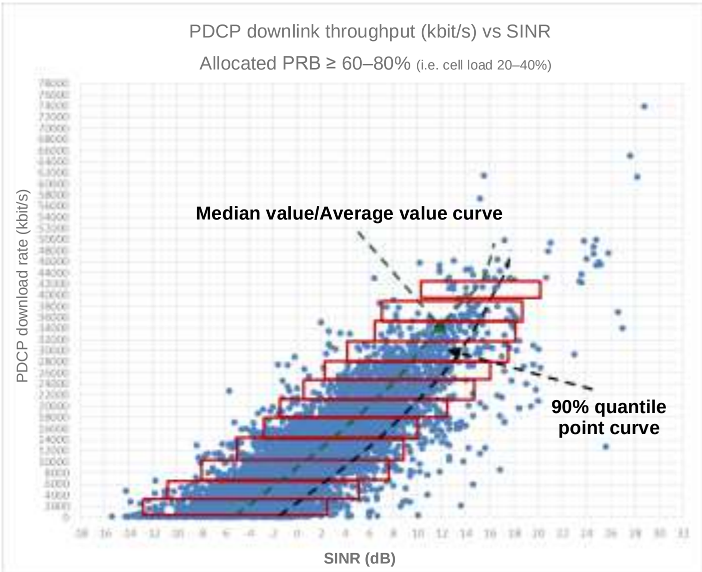
Piecewise fitting is a common method used to map PDCP downlink throughput to SINR and RSRP:

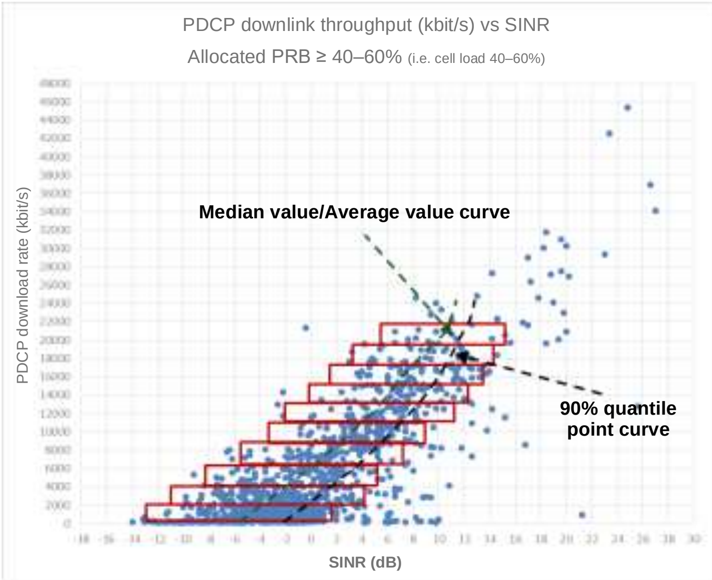
1) Differentiate scenarios (by network mode, frequency band, bandwidth, indoor/outdoor, and PRB load). Specifically, identify cell load based on the number of PRBs allocated to each sampling point in seconds by the driving test (DT) tool. For example, during a DT, actual cell load is about 10% if 90 of 100 PRBs are allocated to the tested terminal, and is about 20% if 80 of 100 PRBs are allocated to the tested terminal. Then draw a scatter chart of PDCP downlink throughput and SINR.

2) Divide the SINR segment at a fixed step (for example, 2 dB) and, at all points in the segment, measure the median distribution value or 90% distribution value of PDCP

downlink throughput and SINR. The following figures are examples of the curves of median distribution values and quantile points using a 20 MHz bandwidth cell in the outdoor, LTE TDD (1.9 GHz) scenario.

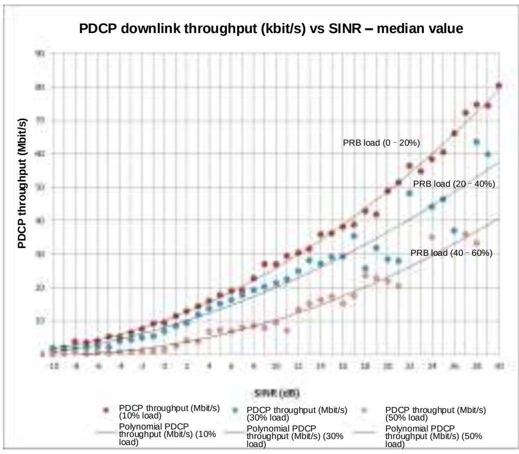


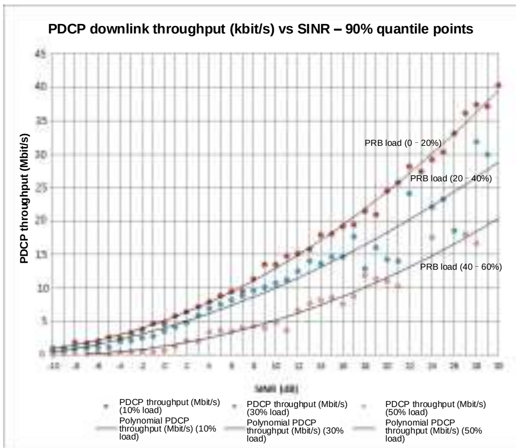




3) Summarise the median value or 90% quantile point results in different scenarios and draw mapping curves between PDCP downlink throughput and SINR.

   The following figures are examples of the mapping curves between PDCP downlink throughput and SINR using a 20 MHz bandwidth cell in the outdoor, LTE TDD (1.9 GHz) scenario.



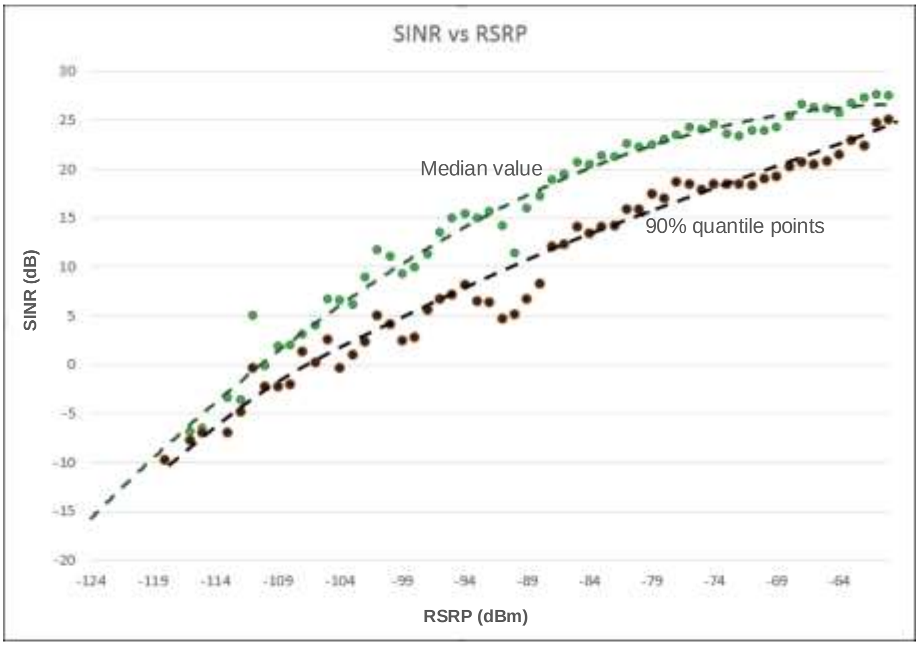


   In general, it is recommended that median value fitting be used as the mapping curve between PDCP downlink throughput and SINR.

4) Draw a scatter chart of SINR and RSRP based on scenarios differentiated by network mode, frequency band, and indoor/outdoor.

5) Divide the RSRP segment at a fixed step (for example, 0.2 dB) and, at all points in the segment, measure the median distribution value or 90% distribution value of SINR and RSRP.

6) Summarise the median value or 90% quantile point results in different scenarios and draw the mapping curve between SINR and RSRP.



In general, it is recommended that median value fitting be used as the mapping curve between SINR and RSRP.

The following table displays the mapped relationship between user throughput and SINR/RSRP based on the test data on China Unicom's live network.

Frequency Band (MHz)	Bandwidth (MHz)	Downlink Rate (Mbit/s)	Light-Load Network		Medium-Load Network		Heavy-Load Network	
			PRB Load ≤ 20%		20% < PRB Load ≤ 40%		40% < PRB Load < 60%	
			SINR (dB)	RSRP (dBm)	SINR (dB)	RSRP (dBm)	SINR (dB)	RSRP (dBm)
1800	20	4.9	–4.1	–115.5	0.4	–109.4	2.6	–105.6
		5	–4.1	–115.5	0.4	–109.4	2.6	–105.6
		5.5	–3.5	–115	1	–108.5	3	–104.5
		6.1	–3.1	–114.6	1.4	–107.9	3.6	–103.8
		6.2	–3.1	–114.6	1.4	–107.9	3.6	–103.8
		6.6	–2.5	–113	3	–107	4	–103
		6.9	–2.2	–112.4	2.4	–106.4	4.6	–102.1
		7.2	–2.2	–112.4	2.4	–106.4	4.6	–102.1
		7.4	–1.5	–110.5	3	–105.5	5	–101
		8	–1.1	–110	3.4	–104.9	5.6	–100.4
		8.3	–1.1	–110	3.4	–104.9	5.6	–100.4

Frequency Band (MHz)	Bandwidth (MHz)	Downlink Rate (Mbit/s)	Light-Load Network		Medium-Load Network		Heavy-Load Network	
			PRB Load ≤ 20%		20% < PRB Load ≤ 40%		40% < PRB Load < 60%	
			SINR (dB)	RSRP (dBm)	SINR (dB)	RSRP (dBm)	SINR (dB)	RSRP (dBm)
		8.6	–0.5	–109.5	4	–104	6	–99.5
		9.7	1.1	–107.9	5.3	–101.7	7.5	–97
		10.1	1.1	–107.9	5.3	–101.7	7.5	–97
		11.8	3.1	–105	7.1	–98.5	9.4	–93.6
		21.3	10.6	–92.1	14.5	–83.8	17.6	–78.6
		28.8	15.5	–81.9	20	–71.4	24.2	–65.7

Based on the above analysis, 1080p requires a rate of 32.8 Mbit/s to achieve a 1s initial buffering delay and a rate of 5 Mbit/s with a 98% freezing-free ratio at the video playback phase, assuming a network RTT of 50ms. If the operator has three 20 MHz carriers, each carrier's required rate is 10.9 Mbit/s (32.8/3). In a lightly loaded network (PRB usage ≤ 20%), the corresponding RSRP is –105 dBm and SINR is 3.1 dB. Network edge rate must be planned based on a –105 dBm RSRP and a 3.1 dB SINR in order to enable instant playback of videos anytime anywhere.

2.2. VR Video

VR blocks the connection between human eyes and the real world and presents a virtual 3D environment through computer rendering. This environment can be a replica of the real world or an imaginary setting. People interact in real time with this 3D virtual environment.

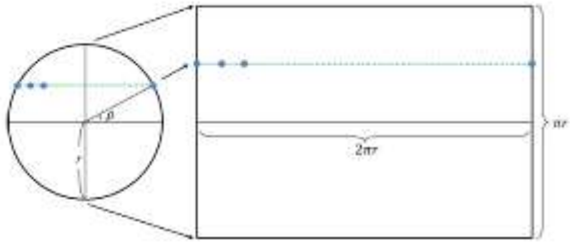
VR has multiple applications: Video on Demand (VOD) and event on-live using 360° panoramic video, standalone VR games, online VR games, and simulated VR environments using computer graphics technology.

VR 360° video provides the observer an all-around physical space view field surrounded 360° horizontally (longitude) and 180° vertically (latitude). A user can enjoy an immersive experience by switching view angle through head position adjustment or through an input device such as a mouse or a remote control.

Since 2016 the VR industry has grown quickly with substantial capital investment. VR is recognized as a major service in the 5G era.

2.2.1. Key Factors to VR Video Experience

A user's view in a virtual environment can be considered as a spatial sphere with a 360° horizontal view if expanded from left to right and a 180° vertical view if expanded from top to bottom. When a user uses a VR terminal, the visual data actually seen by a single eye is only a part of all spherical data, and this part's area is determined by a Field of View (FOV) provided by the terminal.



The visual data of a single eye is only 1/8 and 2/9 of the spherical data with a 90° and 120° FOV, respectively.

Factors that affect VR experience include image quality and interaction experience:

- Image quality experience refers to the resolution of VR videos. Full view and FOV in VR are different. The traditionally defined OTT video resolution corresponds to full-view resolution in VR. However, where users really perceive VR image quality is single-eye resolution (also known as FOV resolution). Take online 4K VR 360° video on YouTube for example. The actual single-eye resolution is only 960 x 960 and only 10 pixels per degree (PPDs) in 90° FOV, which is far lower than the 60 PPDs required by the retina in normal eyesight conditions. The actual video experience is worse than that of SD videos watched on traditional TVs, PCs, and tablets.
- Interaction experience of VR 360° videos is mainly reflected by Motion-to-Photon (MTP) latency. Dominant industry consensus is that immersive terminal MTP latency cannot exceed 20 ms to avoid dizziness. That is, when a user changes a view angle for example, by turning their head, the overall latency of processing by the terminal, network, and cloud should ensure consistency between head movement and change of FOV image, and FOV image update delay should not exceed 20ms.

2.2.2. VR Video Experience Requirements on Networks

VR online videos can be transmitted in two modes: Full-view and FOV transmission.

- The full-view transmission solution transmits all 360° images to terminals and, when the images need to be switched upon user head movement, all the processing is completed locally. This solution requires larger bandwidths but can tolerate a higher latency.
- In the FOV transmission solution, only high-quality images from the current viewing angle are transmitted to terminals in order to save bandwidths. For example, a VR video is divided into 30 viewing angles, and terminals request the corresponding FOV file from the server based on the user's current viewing angle. In this solution, the bit rate can be reduced to about 20% of that required by full-view mode. However, this solution requires a lower latency, as new FOV files must be sent and switched immediately when users turn their heads.

VR evolution goes through the following four stages which vary in network requirements:

1. **Pre-VR:** The representative head mounted display (HMD) terminal is Samsung Gear, and the representative content is 4K VR 360° video on YouTube. In this stage, full-view transmission is the mainstream solution.

2. **Entry-Level VR:** Terminal full-view resolution is increased to 8K so that the image quality is close to the PPD of 480p videos viewed on a PC. The industry prefers the full-view transmission solution to ensure good interactivity.

3. **Advanced VR:** Screen resolution, chip performance, human body engineering, and content quality of HMD terminals are greatly improved. Due to improved image quality, the full-view transmission solution requires large network bandwidth.

4. **Ultimate VR:** User experience is the best in this stage due to further development of HMD terminals and content. Single-eye image quality reaches retina-level. Due to extremely high network bandwidth requirements in the full-view transmission solution, FOV transmission must be used but it requires a lower network latency.

Standard	Pre-VR	Entry-Level VR	Advanced VR	Ultimate VR
Video resolution	Full-view 4K 2D video (YouTube) (full-view resolution 3,840 x 1,920)	Full-view 8K 2D video (full-view resolution 7,680 x 3,840)	Full-view 12K 2D video (full-view resolution 11,520 x 5,760)	Full-view 24K 3D video (full-view resolution 23,040 x 11,520)
Single-eye resolution	960 x 960 (wearing VR glasses, 90° FOV)	1,920 x 1,920 (wearing VR glasses, 90° FOV)	3,840 x 3,840 (wearing dedicated HMD terminals, 120° FOV)	7,680 x 7,680 (wearing dedicated HMD terminals, 120° FOV)
PPD	11	21	32	64
Equivalent traditional TV screen resolution	240p	480p	2K	4K
Frame rate (frame/second)	30	30	60	120
Typical video bit rate	16 Mbit/s	64 Mbit/s	279 Mbit/s	3.29 Gbit/s
Typical network bandwidth requirement	25 Mbit/s (full-view transmission which is the mainstream solution in this stage)	100 Mbit/s (full-view transmission which is the mainstream solution in this stage)	418 Mbit/s (full-view transmission which is the mainstream solution in this stage)	1 Gbit/s (FOV solution)
Typical network RTT (ms)	40	30	20	10

2.2.3. VR Video Experience Requirements on RF Performance

VR video experience has similar requirements on air interface capabilities as video experience. For details, see section "Video Experience Requirements on RF Performance." Take Pre-VR for example. Assuming a network RTT of 40 ms, a smooth play rate of 25 Mbit/s, and three 20 MHz carriers, the rate of each carrier is about 8.3 Mbit/s (25/3). In a lightly loaded network (PRB usage ≤ 20%), the corresponding RSRP is –110 dBm and SINR is –1.1 dB, which must be used as the basis when planning the network edge rate.

2.3. Mobile Gaming

Mobile games, running on mobile terminals such as phones and tablets, have developed rapidly with the recent popularity of smartphones and mobile Internet. According to Newzoo, the global consumption of mobile games is estimated to be US$ 68.5 billion in 2019, with a year-on-year (YOY) growth rate of 10.2%.

Mobile games fall into the following types: Action (ACT), Adventure Game (AVG), Role-Playing Game (RPG), First Person Shooter (FPS), Racing Game (RAC), Sports Game (SPG), Simulated Life Game (SLG), Real-Time Strategy Game (RTS), Music Game (MUG), Fighting Game (FTG), Puzzle (PUZ), Shooting Game (STG), and so on. FPS, RTS, and RAC games have high requirements on network latency and low requirements on network bandwidth, and will be discussed in detail within this document.

2.3.1. Main Factors Affecting Mobile Game Experience

A mobile game usually involves three phases:

Phase 1: Patch start and download. Download time affects user experience.

Phase 2: Loading. Load time affects user experience.

Phase 3: Battle. Response latency affects user experience.

The patch download is mostly completed over Wi-Fi, and as such user experience is not affected by mobile network quality. In the loading phase, latency depends on terminal storage read speed and is independent of the mobile network. The battle phase is the most crucial part where user experience might be affected, because the game latency mainly depends on the network transmission latency.

2.3.2. Requirements of Mobile Gaming Experience on Networks

The Worcester Polytechnic Institute conducted a study titled *On Latency and Player Actions in Online Games*. The following figure illustrates the relationship between experience and latency for FPS and RAC games.

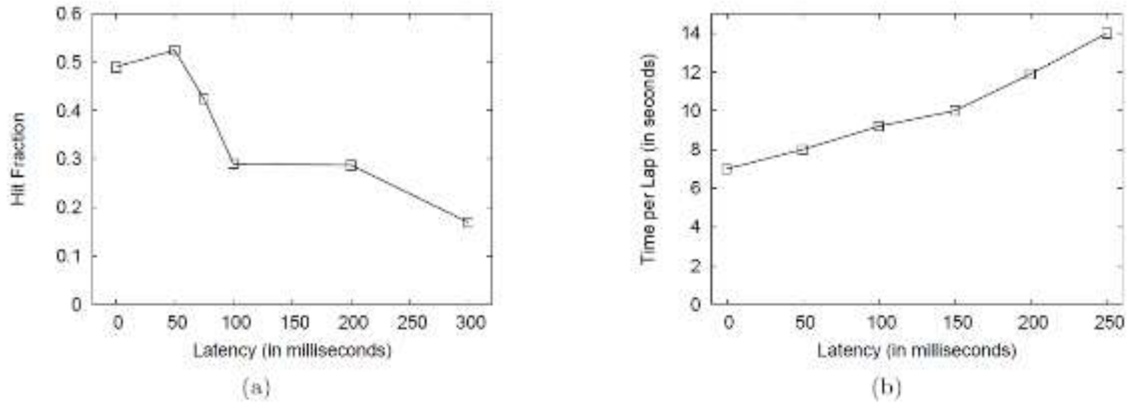


Figure (a) depicts the relationship between latency and the scores of an FPS game. As latency increases, the scores drop sharply at 100 ms latency.

Figure (b) depicts the relationship between latency and the time to complete a lap around a racetrack of a RAC game. As latency increases, the time to complete a lap around a racetrack increases sharply at 150 ms latency. (For details about this study, see https://digitalcommons.wpi.edu/cgi/viewcontent.cgi?article=1053&context=computerscience-pubs.)

OTT game companies pay much attention to user experience and usually measure E2E latency through heartbeat packets. The following are the RTS mobile game experience standards of an OTT content provider:

- Latency: Game latency is collected and calculated by sending user datagram protocol (UDP) heartbeat packets every 5s.
- High frame freezing round: RTT latency of all packets sent in one round of game is sampled and calculated based on the number of sampling points that fall into the following five ranges.

High frame freezing round is identified if the following is true: Packets sent in each range x Weight/Total packets > 15%.

Latency Range (ms)	Weight
< 100	0
100–200	0.2
200–300	0.5
300–460	0.7
> 460	1

- Cell frame freezing ratio = Rounds with frame freezing/Total rounds
- A cell with frame freezing is identified if its frame freezing ratio exceeds 7% and the number of rounds in a week is greater than 70.

According to the preceding analysis, E2E latency must be lower than 100 ms to ensure good user experience of mobile games.

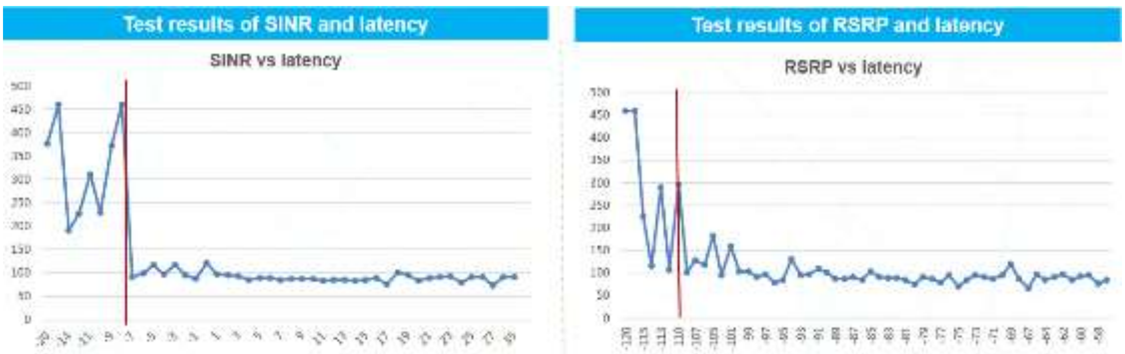
2.3.3. Mobile Game Experience Requirements on RF Performance

Two-way latency between a base station and a game server depends on the backhaul latency, core network processing latency, and server processing latency. Backhaul latency depends on the distance and number of hops between the base station and game server. Processing latency depends on the network element's processing capability. According to live network tests, the latency between an eNodeB and a game server is generally within 30–50 ms. Therefore, the latency on the radio access network (RAN) side must be within 50 ms to ensure smooth user experience.

Latency over air interface can be affected by signal quality and cell load.

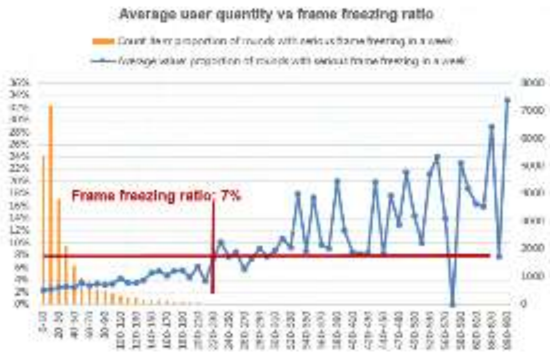
Signal quality

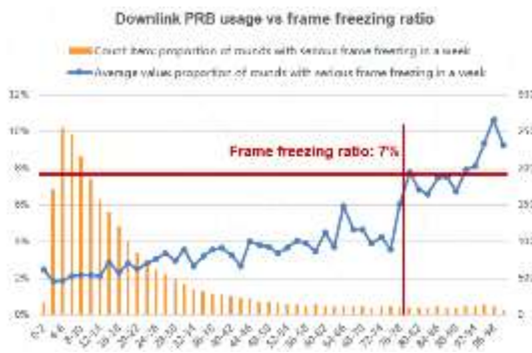
In areas with poor radio signal coverage, the block error rate (BLER) over air interface is high. Therefore, multiple retransmissions over the physical layer are required before data is successfully transmitted, prolonging transmission latency. The following figure shows field test results of a load-free cell. To ensure a loopback latency from the UE to the game server of less than 100 ms, the SINR must be greater than –7 dB and RSRP must be greater than –110 dBm.



Cell load

In heavily loaded cells, game data packages may stay in the scheduling queue for a long period of time due to multi-user scheduling, increasing latency and deteriorating user experience. The following figure shows the correlation between the cell frame freezing ratio reported by mobile game OTT and the number of users and PRB usage in this cell. The number of users in busy hours and downlink PRB usage are in a positive linear relationship with the cell frame freezing ratio. Cell frame freezing ratio exceeds 7% when the number of users is above 220 or when the PRB usage is above 78%.





2.4. VoLTE Call

Compared with GSM or UMTS voice services, VoLTE voice services provide higher spectral efficiency, clearer voice quality, and shorter call setup delay, and users can have high-speed data services while a VoLTE call is proceeding.

2.4.1. Key Factors to VoLTE Experience

For voice over LTE (VoLTE), user experience is reflected by basic key performance indicators (KPIs) and voice quality.

Basic KPIs indicate service availability, including the call success rate, call setup delay, and call drop rate.

- The call success rate refers to the rate that a calling party initiates calls, to the rate the calling party hears the ringback tone. The call success rate is affected by the E2E service process and, excluding the impact of exceptions on the core network, mainly refers to the E-UTRAN radio access bearer (E-RAB) setup success rate with a RAN QoS class identifier (QCI) of 1 (QCI 1).
- The call setup delay is the duration from the time when the calling party initiates a call to the time the calling party hears the ring back tone. Delay is the shortest for VoLTE calls and longer for circuit-switched fall back (CSFB) calls.
- The call drop rate is the proportion of dropped calls after the calling and called parties successfully set up calls. Main factors affecting the call drop rate are on the RAN side, such as handover failure, weak coverage, and strong interference. RAN-side call drop rate is reflected by the proportion of abnormal QCI1 E-RAB releases.

Voice quality indicates service quality. A higher voice bit rate ensures better voice quality. Generally, voice quality is represented by the mean opinion score (MOS).

The following table lists the recommended baseline values of basic VoLTE KPIs. Operators can set the baseline values based on their network development stage.

KPI	Early Stage	Mature Stage	Ultimate Experience
Call success rate	> 99%	> 99.5%	> 99.9%
Call setup delay	< 8s	< 2.5s	< 1s
Call drop rate	< 1%	< 0.4%	< 0.05%

MOS is an important indicator for voice quality of communication systems. Generally, the MOS value is calculated by using a specific algorithm, such as perceptual objective listening quality analysis (POLQA), to compare the source signal with the receiver signal. The MOS value ranges from zero to five. The higher the value, the better the voice quality. Generally, user experience is good when the MOS value is greater than or equal to four, and is acceptable when the MOS value is greater than or equal to three.

MOS	Quality	Service Loss
5	Excellent	None (excellent service experience, smooth and clear)
4	Good	A little (perceived service loss but user experience not affected)
3	Medium	Slight
2	Poor	Significant
1	Bad	Extreme (services interrupted)

The average (50%) or bottom (10%) MOS value can be used to evaluate voice quality of a network. Of the two, the bottom MOS value is recommended, given that the average MOS value is easy to conceal problems and will not benefit network optimization.

The MOS value of enhanced voice services (EVS) is higher than that of adaptive multi-rate wideband (AMR-WB) because of the advantages of EVS. Therefore, it is recommended that the P10 (10%) target MOS value be set for EVS and AMR-WB separately. The suggestions are as follows:

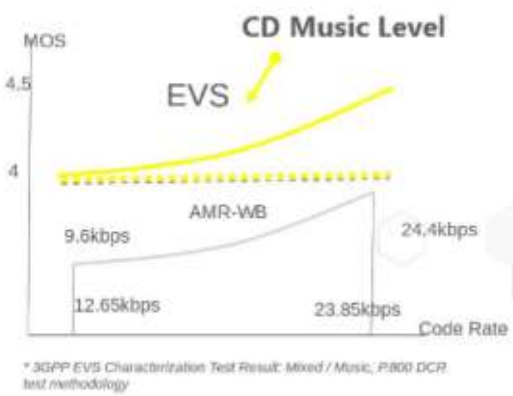
KQI	Early Stage	Mature Stage	Ultimate Experience
AMR-WB 10% MOS	> 2.5	> 3.0	> 3.5
EVS P10 MOS	> 3.0	> 3.5	> 4.0

Voice call MOS value is affected by the codec, E2E transmission latency, jitter, and packet loss rate.

Codec

VoLTE supports the following voice codecs: EVS, AMR-WB, and adaptive multi-rate narrowband (AMR-NB). A higher bit rate offers better user experience.

Codec	Bandwidth (KHz)	Bit Rate (kbit/s)
EVS	20	5.9–128
AMR-WB	7	6.6–23.85
AMR-NB	3.4	4.75–12.2



Driven by HD voice experience requirements, the VoLTE voice codec is changing to high bit rate and the EVS codec. It is recommended that AMR-WB 23.85 kbit/s or EVS 24.4 kbit/s be used as the network construction criteria.

E2E latency

The following factors affect E2E latency: UE voice coding and decoding latency, air interface transmission latency, core network processing latency, and transport network transmission latency. Transmission latency over the air interface fluctuates greatly, which is affected by the eNodeB scheduling wait latency, retransmission of error packets over the air interface, and segmentation.

Packet loss and jitter

The main factors that cause packet loss and jitter are signal quality over the air interface, eNodeB load, and packet loss or jitter on the transport network. Poor air-interface signal quality may increase the packet error rate, which results in more packet retransmissions and segmentation. As a result, the number of lost packets and jitters increases.

According to the preceding analysis, voice quality (namely experience) is determined by voice codec and air interface quality.

2.4.2. VoLTE Experience Requirements on RF Performance

RF performance directly affects voice experience. VoLTE services, in particular, are symmetric in the uplink and downlink, but the uplink coverage is likely to be a bottleneck. To solve this problem, features such as robust header compression (ROHC), uplink radio link control (RLC) segment, and uplink TTI bundling can be enabled on the RAN side to reduce uplink requirements and improve the VoLTE experience of cell edge users.

VoLTE coverage is limited in the uplink and therefore the valid coverage area is determined by the uplink coverage area. Uplink coverage requires a large enough uplink SINR so that the base station can correctly demodulate voice contents. According to simulation results, the uplink SINR is –6 to –3 dB when the MOS is greater than 2.5 to 3.5 for a probability of 90%.

Assuming a 20 MHz bandwidth on the 1800 MHz band, 2T2R antenna configuration of the base station, a total power of 40 W, and the 23.85 kbit/s VoLTE codec, the theoretical RSRP required by UEs at the uplink coverage edge is calculated as follows:

Item	Formula	Value	No.
eNodeB reference signal (RS) power	Total power – 10 x log (12 x Total number of RBs) (The total number of RBs is 100.)	15.2 dBm	A
UE equivalent isotropically radiated power (EIRP)	23 – 10 x log (12 x Number of RBs) (The number of RBs is 3.)	7.44 dBm	B
Uplink SINR	Base station performance, MOS simulation value of 2.5–3.5	–6 to –3 dB	C
Difference between uplink and downlink frequency bands	33.9 x log (Downlink frequency/Uplink frequency) (The downlink frequency and uplink frequency are 1850 MHz and 1755 MHz, respectively.)	0.78	D

Item	Formula	Value	No.
Uplink interference	Thermal noise + Noise coefficient + Uplink interference margin = (–174 + 10 x log [15000]) + 2.3 + (6 – 10)	–124 to –120 dBm	E
Receiver sensitivity	A – B + C – D + E	–123 to –116 dBm	F
Body loss	-	3 dB	G
UE loss	-	6 dB	H
RSRP	F + G + H	–114 to –107 dBm	

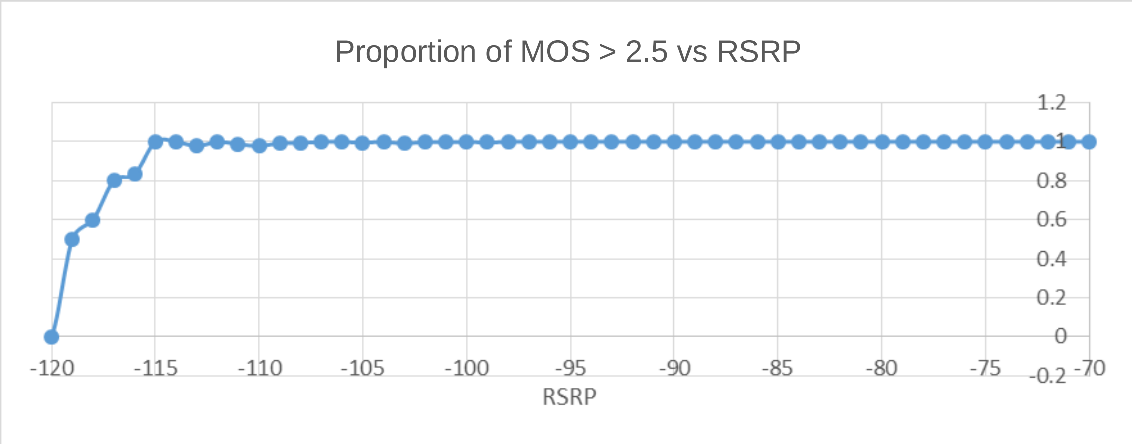
The uplink interference margin is related to load. In urban areas where load is high, it is recommended that the uplink interference margin be set to 10 dB (with the corresponding RSRP of about –110 dBm). In rural areas where load is light, the uplink interference margin can be set to 6 dB (with the corresponding RSRP of about –116 dBm).

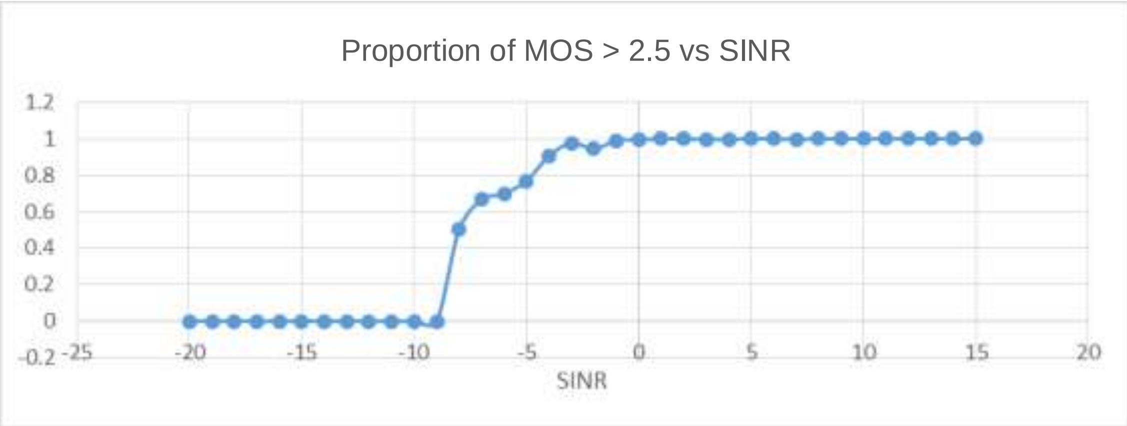
Bottom MOS 10%	RSRP (Light Load, < 30%) with Interference Margin of 6	RSRP (High Load, > 50%) with Interference Margin of 10
2.5	–114	–110
3.0	–112	–108
3.5	–111	–107

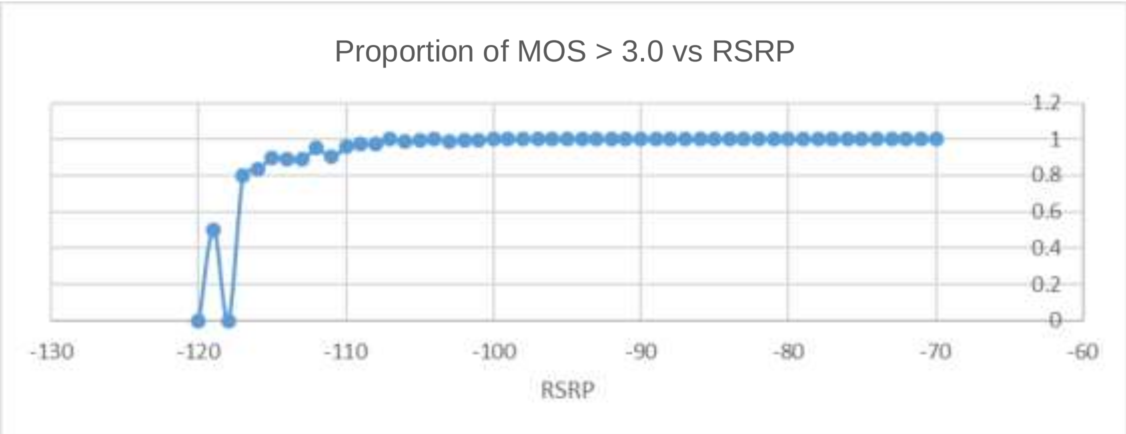
The preceding table lists the theoretical coverage capability requirements. The following describes MOS correlation with RSRP and SINR based on live network test results.

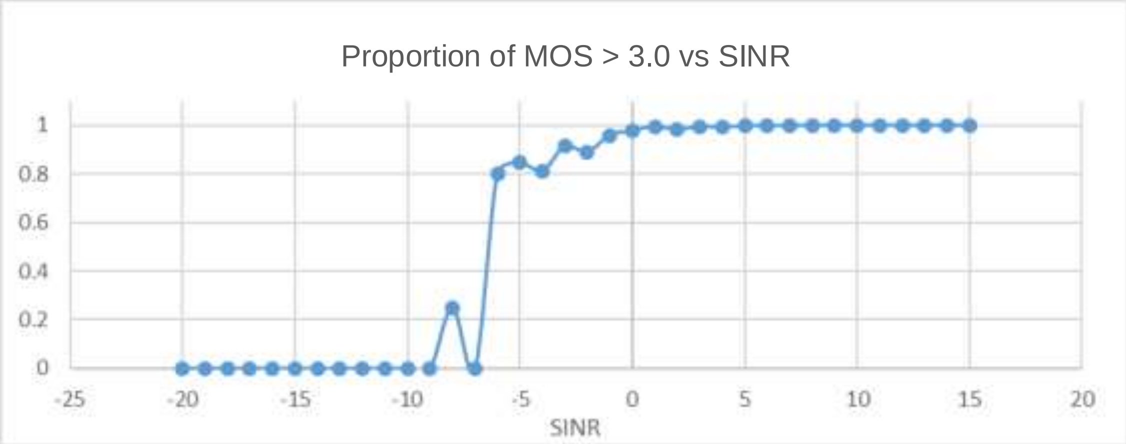
A large number of DTs on VoLTE 23.85 kbit/s HD voice calls were performed in a field of urban areas on a China Unicom live network. In the DTs, RS power was 15.2 dBm, load was below 30%, and RoHC and TTI bundling were enabled. RSRP, SINR, and MOS at each location during the DTs or fixed-point tests were recorded, and required RSRP and SINR were calculated when the P10 MOS value was greater than 2.5, 3.0, and 3.5 respectively.

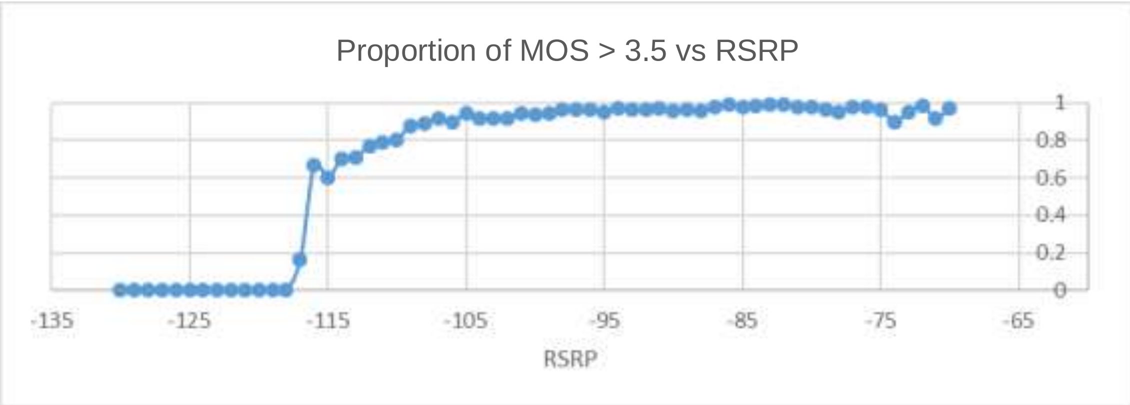
According to the analysis, when 90% of MOS values are greater than 2.5, RSRP is about –116 dBm and SINR is greater than –4 dB; When 90% of MOS values are greater than 3.0, RSRP is about –112 dBm and SINR is greater than –3 dB; When 90% of MOS values are greater than 3.5, RSRP is about –107 dBm and SINR is greater than 2 dB.

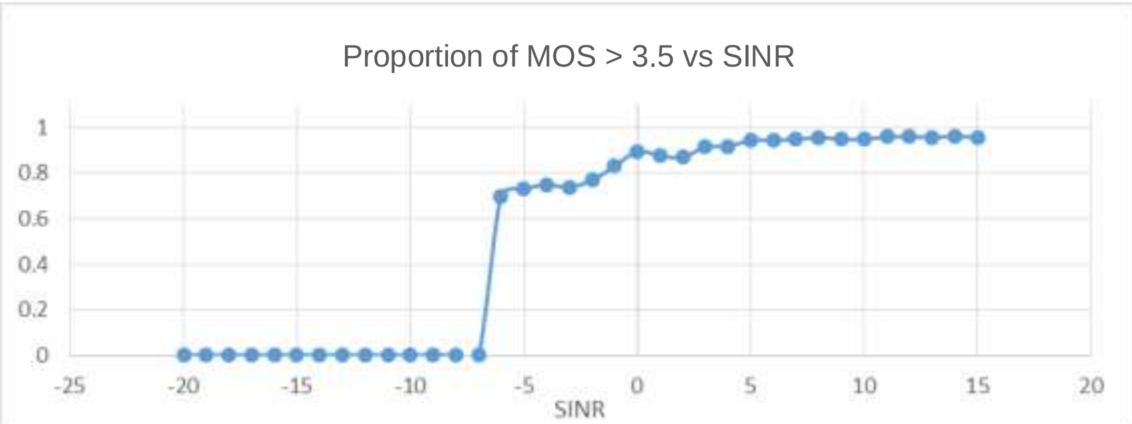

Proportion of MOS > 2.5 vs RSRP
1.2
1
0.8
0.6
0.4
0.2
0
-0.2
-120
-115
-110
-105
-100
-95
-90
-85
-80
-75
-70
RSRP


Proportion of MOS > 2.5 vs SINR
1.2
1
0.8
0.6
0.4
0.2
0
-0.2
-25
-20
-15
-10
-5
0
5
10
15
20
SINR


Proportion of MOS > 3.0 vs RSRP
1.2
1
0.8
0.6
0.4
0.2
0
-130
-120
-110
-100
-90
-80
-70
-60
RSRP







The load was light during DTs. Considering the increase in interference caused by subsequent load increases, a 2 to 4 dB interference margin was reserved during planning. Therefore, it is recommended that when the P10 MOS is greater than 2.5, 3.0, and 3.5, the RSRP be –112 dBm, –108 dBm, and –103 dBm, respectively, and the SINR be greater than –4 dB, –3 dB, and 2 dB, respectively.

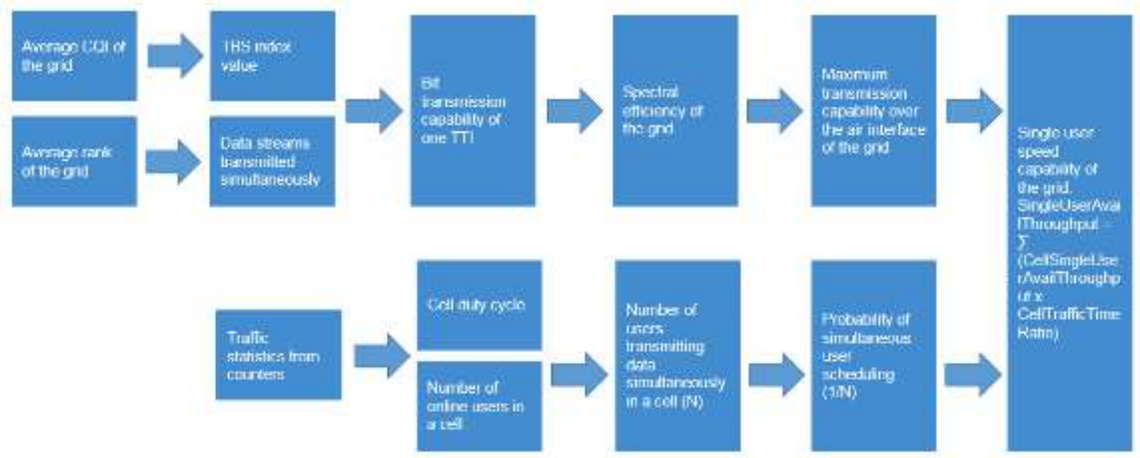
According to this principle, the planned RSRP needs to be adjusted accordingly if the configuration changes. For example, if the RS power increases, the planned RSRP must also increase. If the base station uses 4R configuration, the planned RSRP can decrease by about 3 dB.

3. Typical Methods for Service Experience Evaluation

3.1. Using MRs to Measure User Experience Distribution

MRs from UEs contain information such as CQI, rank, RSRP (of the serving cell and neighbouring cells), and Global Positioning System (GPS). Based on the CQI and rank, the data volume transmitted in each TTI at a location can be calculated. Based on mapping among the GPS and geographic grids (50 m x 50 m), the spectral efficiency and maximum transmission rate of a grid can be calculated. Since the transmission capability of a grid is shared by multiple users, the number of users transmitting data simultaneously in a cell must also be obtained in order to obtain the grid's user experienced rate.

The number of UEs that transmit data simultaneously in a cell can be obtained based on traffic statistics, including the number of UEs in RRC_CONNECTED state and the data transmission duty cycle.



Based on the MRs from GPS-enabled UEs, a fingerprint database that maps the GPS and RSRP data of each cell can be constructed on the network side. If the GPS function is not enabled on a UE, the UE can query the fingerprint database for grid mapping based on the cell RSRP in the MR. The UE can then participate in the calculation of grid-level experienced rate, improving the accuracy of experience evaluation.

MR 1	Latitude and Longitude P1	**Cell Name**	**Cell 1**	**Cell 2**	**Cell 3**	...
		RSRP	–85	–79	–95	...

MR 2	Latitude and Longitude P2	**Cell Name**	**Cell 1**	**Cell 2**	**Cell 3**	...
		RSRP	–90	–70	–100	...

...

MR *n*	Latitude and Longitude P*n*	**Cell Name**	**Cell 1**	**Cell 2**	**Cell 4**	...
		RSRP	–80	–77	–90	...

The following figure shows the experience data distribution in a city served by China Unicom. The experienced rate distribution in different areas is intuitively displayed, and poor experience areas can be identified.

3.2. VoLTE Experience Evaluation Method

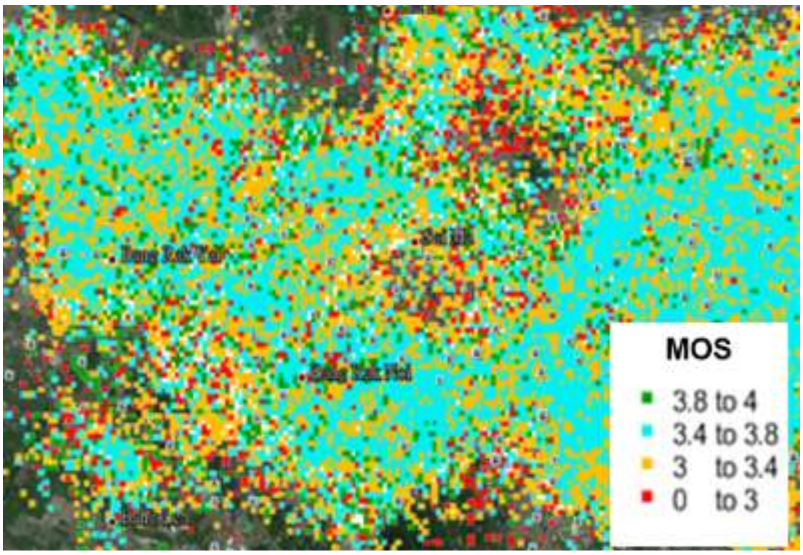
VoLTE service evaluation involves basic KPIs and voice quality.

Basic KPIs can be evaluated by observing network KPIs. Call success rate and call drop rate correspond to the QCI1 E-RAB setup success rate and abnormal release rate on the RAN side respectively. Call setup delay on the IP multimedia core network subsystem (IMS) is observed.

Voice quality is evaluated by the MOS value. To obtain the MOS value, perceptual evaluation of speech quality (PESQ) standardized in ITU-T Recommendation P.862 or POLQA standardized in ITU-T Recommendation P.863 is used to perform tests with assistance of special instruments. Specifically, VoLTE calls are tested in the network planning area by means of outdoor DTs or indoor fixed-point tests. The target UE is connected to a professional voice quality test instrument (such as DSLA) which outputs the MOS value. This evaluation method requires professional test personnel to perform a large number of tests on roads or indoors to obtain the voice quality of a network. This method features high accuracy but huge labour input and insufficient traversal.

As an alternative of DT-based voice quality evaluation, the packet loss rate, packet error rate, and jitter of uplink and downlink voice packets on the base station can be measured (periodically by the base station and then mapped to MOS values using algorithms). Statistics are collected by segment based on the MOS 0–5 range to obtain the distribution of different MOS values in a cell. This method can evaluate the voice quality of a network from a macro perspective without performing DTs, but the specific location of problems that compromise voice quality cannot be determined.

The grid-based MOS evaluation method combines the advantages of the preceding two methods. In this method, a network is divided into multiple 50 m x 50 m grids, with the RSRP and SINR of each location on the network obtained based on the MRs of UEs and then mapped to the grids. The voice quality in a grid is estimated based on mapping among the MOS value, RSRP, and SINR determined in section 2.4.2 "VoLTE Experience Requirements on RF Performance." This method can be used to obtain the voice quality of a network from a macro perspective and identify specific areas where voice quality problems exist, thereby providing accurate input for network optimization.



4. Summary

In the 5G era, network deployment and optimisation must take both signal coverage and service experience satisfaction rates into consideration.

The mapping between services and air interface indicators described in this document can guide network planning and optimisation. Based on the target services carried on a network, experience requirements can be analysed, and experience objectives can be set accordingly. Eventually, air interface KPIs such as RSRP, SINR, and network load are obtained to guide network construction.

The service experience evaluation methods described in this document can help operators identify problems with user experience and perform targeted network optimisation, thereby providing better user experience assurance.

GSMA HEAD OFFICE
Floor 2
The Walbrook Building
25 Walbrook
London
EC4N 8AF
United Kingdom
www.gsma.com