Provision of Multiuser 360º Video Streaming over MEC-enabled 5G networks

Miguel Loff Barreto
Instituto de Telecomunicações
Instituto Superior Técnico, Universidade de Lisboa
Lisbon, Portugal
miguel.loff@tecnico.ulisboa.pt

***Abstract*—Data traffic on the next generation of wireless networks (5G) is expected to be dominated by challenging video applications, such as 360º video streaming. In order for 5G networks to handle this and other demanding applications while meeting the users' Quality of Experience (QoE) requirements, it is necessary to characterize the network side factors, such as the deployment of Multi-Access Edge Computing (MEC) infrastructure and the used video delivery scheme, jointly with the 360º video features. To this end, the purpose of this thesis is to assess and compare the performance of different multiuser 360º video streaming solutions proposed in the literature. Since the high bandwidth requirements of 360º video delivery strongly limit the number of supported simultaneous streaming sessions, where users are watching different videos with different viewports, it is important to optimize the resource allocation in order to maximize the number of users that can satisfactorily stream 360º videos. Furthermore, this thesis aims at assessing the performance gains that can be achieved by the deployment of MEC infrastructure in the network. It is found that using a tiles-based streaming scheme increases the number of very satisfied users by 50% over the traditional viewport-independent scheme. Furthermore, it is found that deploying MEC servers on the edge of the network unlocks the possibility of using a viewport-rendering scheme, which allows a further three-fold capacity gain over the tiles-based scheme, while virtually eliminating non-satisfied users.**

***Index Terms*—360º video streaming, viewport adaptive streaming, 5G, QoE, MPEG-DASH**

I. Introduction

Roughly every ten years, there is a revolution in mobile telecommunications technology. With each of these revolutions, the paradigm of what is possible to achieve with a wireless network shifts, unlocking completely new use cases.

The next generation of wireless communications systems, the 5th Generation (5G), aims at keeping up with the ever-growing global mobile data traffic, while delivering a leap forward in data rates, latency, connectivity, and more. It is forecast that, in 2026, there will be 3.5 billion 5G subscriptions, accounting for 40% of all mobile service subscriptions, and that 5G data traffic will account for 122 exabytes per month [1]. This growth is going to be driven by challenging new use cases unlocked by the 5G networks, such as mobile broadcasting, remote surgery and Augmented Reality (AR).

Virtual Reality (VR) and AR applications are particularly interesting use cases for 5G networks, having the potential to emerge as one of the early opportunities for mobile network operators. This market will be worth $292 billion by 2025 [2]. Globally, AR and VR traffic will grow nearly 12 times from 22 petabytes per month in 2017 to 254 petabytes in 2022 [3].

One of the most well developed and adopted VR/AR applications is 360º video, and therefore it is likely that this application will be one of the first ones to gain popularity. This is an application where users watch videos such as sports, live shows, documentaries and more, with the freedom to look around the entire 360º sphere around the point of view of the camera. In order to this, they use a Head-Mounted Display (HMD) device, which allows them to look around, with the video accompanying this motion. The current availability of omnidirectional cameras make it easy to produce personalized 360º videos and social media platforms like Facebook and YouTube already allow users to publish these videos [4]. Currently, this application poses significant challenges for wireless networks since it requires the transfer and storage of large amounts of data and demand significant computing resources. HMD devices have limited communication and storage resources and finite processing capabilities. Therefore, this and other VR/AR applications are dependent on the rollout of 5G networks, which, with their more ambitious data rate and latency targets, will enable new schemes for the delivery of the content and expand the capabilities of the network to support it. However, it is still necessary to develop these schemes in order to cope with the ultra-high data rate and ultra-low latency requirements of 360º video in order to be able to support more than a few simultaneous users and therefore justify the investment in infrastructure. It is also necessary to study to which extent do the improvements enabled by 5G influence the users' Quality of Experience (QoE).

Several delivery schemes have been proposed in the literature with the aim of reducing the necessary bandwidth necessary for 360º video streaming. In this paper, it is studied how these schemes compare against each other and in which ways the use of 5G networks improves the streaming experience for users. It also studied how one of the most promising new technologies of 5G, Multi-Access Edge Computing (MEC), may be beneficial for these applications.

Section II presents an overview of 5G. Section III presents the background on 360º video streaming. Section IV describes the simulator implementation. Section V presents the main results and analysis. Section VI provides the conclusions.

II. 5G Overview

5G is the new standard of mobile access technology, aiming at being a leap forward in terms of data rates, latency, reliability, capacity and availability. 5G represents a major step in the capabilities that are expected from mobile networks, as it will enable services like Internet of Things (IoT) on a massive scale, data rates of up to 20 Gbps and a capacity increase of up to 1000 times [5]. 5G networks enable new use cases for consumers, enterprises, homes, and public domains. The 360º video streaming application pertains to the consumer segment, which will be enhanced by the available higher data rate and lower latency.

The requirements of 5G are driven by the International Telecommunications-2020 (IMT-2020). IMT-2020 is aimed at supporting a wide range of use cases in three broad usage scenarios: Enhanced Mobile Broadband (eMBB), Ultra-Reliable and Low Latency Communications (URLLC) and Massive Machine Type Communications (mMTC). The eMBB scenario addresses the human-centric use cases for access to multimedia content, services and data, supporting improved performance and an increasingly seamless usage experience. The peak download data rate is as high as 20 Gbps. URLLC addresses stringent requirements for capabilities such as throughput, latency and availability for delay-sensitive applications such as intelligent transport systems, vehicle-to-everything, remote medical surgery, smart grids, among others. mMTC is a family of applications that consist of a very large number of devices typically transmitting a relatively low volume of non-delay-sensitive data. These applications may be IoT or Machine-to-Machine [6].

A. 5G Physical Layer

The physical layer in 5G has several differences from 4G. Unlike in 4G, where only a sub-carrier spacing of 15 KHz is used, 5G NR enables flexibility of waveforms with scalable Subcarrier Spacing (SCS), according to $\Delta f = 2^{\mu} \times 15$ KHz, where μ is an integer between 0 and 4, yielding spacings between 15 KHz and 240 KHz. The choice of μ has to do with factors such as cell size, latency requirement and carrier frequency. Wider subcarrier spacing is beneficial in situations where latency is critical, while narrower subcarrier spacing is suitable for outdoor coverage [8]. In 5G, downlink and uplink transmissions continue to be organized into 10 ms long frames, as in 4G. Each frame consists of 10 subframes of 1 ms each. The number of slots per subframe varies according to μ, as illustrated in Figure 1. Each slot contains 14 Orthogonal Frequency Division Multiplexing (OFDM) symbols [9]. The 5G resource grid is formed by one subframe in the time domain and a carrier bandwidth in the frequency domain. A Resource Element (RE) is its smallest unit, and is constituted by the combination of one subcarrier in the frequency domain and one OFDM symbol in the time domain. REs are grouped into Physical Resource Blocks (PRBs). Each PRB consists of 12 subcarriers. The number of PRBs varies depending on the used numerology and bandwidth.

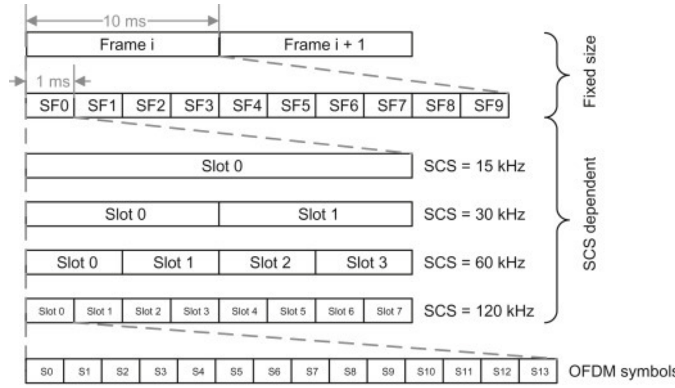

Fig. 1. 3GPP NR frame structure [10].

B. 5G Performance

In 5G, the approximate data rate can be calculated in Mbps by

$$\text{rate} = 10^{-6} \cdot \sum_{j=1}^{J} \left(v_{Layers}^{(j)} \cdot Q_m^{(j)} \cdot f^{(j)} \cdot R_{max} \cdot \right. \left. \cdot \frac{N_{PRB}^{BW(j),\mu} \cdot 12}{T_s^{\mu}} \cdot \left(1 - OH^{(j)}\right) \right), \quad (1)$$

where J is the number of aggregated component carriers in a band or band combination, R_{max} is the maximum coding rate and, for the j^{th} Component Carrier (CC), $v_{Layers}^{(j)}$ is the maximum number of layers, $Q_m^{(j)}$ is the maximum modulation order, $f^{(j)}$ is the scaling factor, μ is the numerology, T_s^{μ} is the average OFDM symbol duration for numerology μ, $N_{PRB}^{BW(j),\mu}$ is the maximum PRB allocation in bandwidth $BW^{(j)}$ with numerology μ and $OH^{(j)}$ is the overhead [11].

The Modulation Coding Scheme (MCS) defines the used coding rate, which is the proportion of useful bits in relation to total transmitted bits which can be carried by one RE. The MCS defines two aspects: the type of modulation used and the code rate. When the channel quality is poorer, a lower coding rate is used, meaning more redundant bits are be transmitted. In order to select the appropriate MCS, the radio channel quality has to be estimated. This is achieved with the Channel Quality Indicator (CQI), which is a metric periodically transmitted from the UE to the base station. The higher the value of the CQI, the higher the achievable efficiency of the transmission, allowing for more useful bits to be transmitted with each PRB. The achievable MCS for each CQI, considering a poor channel, is shown in Table I.

C. Multi-Access Edge Computing in 5G

MEC is a key technology in 5G, defined by the European Telecommunications Standards Institute (ETSI) as a technology that provides an IT service environment and cloud computing capabilities at the edge of the mobile network, within the Radio Access Network (RAN) and in close proximity to mobile subscribers. It can host compute-intensive applications,

TABLE I
EFFICIENCIES FOR EACH CQI (POOR CHANNEL) [12].

CQI Index	Modulation	Code rate ($\times$1024)	Spectral efficiency
0	(out of range)		
1	QPSK	78	0.1523
2		120	0.2344
3		193	0.3770
4		308	0.6016
5		449	0.8770
6		602	1.1758
7	16-QAM	378	1.4766
8		490	1.9141
9		616	2.4063
10	64-QAM	466	2.7305
11		567	3.3223
12		666	3.9023
13		772	4.5234
14		873	5.1152
15		948	5.5547

process large data before sending to the cloud and offer context-aware services. As such, MEC enables a wide variety of applications where a real-time response is strictly required [13]. Among the factors that have been driving the demand for MEC is low-latency computing, as this is a fundamental metric for network performance and is required by many applications, such as 360º video streaming.

III. 360º VIDEO STREAMING

One of the main applications of VR is 360º video which, supported by the growing number of existing VR headsets and availability of omnidirectional cameras, has created significant interest in this type of content. A 360º video is typically presented to the user by means of a Head-Mounted Display (HMD). The HMD is equipped with sensors that detect which direction the user is facing and it uses this information to deliver the corresponding region of the 360º video, known as the viewport [14]. 360º video streaming presents a series of new challenges when compared to 2D videos. Among these challenges are high bandwidth requirements, large storage requirements, low motion-to-photon latency, complex view adaptation and understanding what influences the user's QoE [4]. These challenges make the current video delivery architectures unsuitable for the 360º video case. Critically, the transmission bandwidth needs to be reduced and the network infrastructure needs to be optimized in order to reduce the end-to-end latency and manage the resource allocation.

A. MPEG-DASH Framework

After the 360º video is acquired, stitched, projected and encoded, it needs to be transmitted. Due to fluctuations in the channel's quality and availability, delays and loss of data may happen, impairing the viewing experience. In order to solve this problem, Adaptive Bitrate (ABR) techniques are used in order to adapt the transmitted video quality to the user's available bandwidth. One popular standard to incorporate this technique is the MPEG-DASH protocol [15]. The MPEG-DASH stream is divided into temporal segments, encoded with multiple bitrates or spatial resolutions. The MPEG-DASH standard serves as the foundation for 360º video streaming, by leveraging the DASH video framework for bitrate adaptation, allowing the client to select which bitrates and resolutions it requests and to switch between qualities depending on the available resources.

B. 360º Video Streaming Schemes

360º video streaming poses specific challenged that MPEG-DASH alone does not address. Therefore, standards such as the Omnidirectional MediA Format (OMAF) were developed specifically for VR. OMAF includes three different streaming schemes: viewport-independent streaming, viewport-dependent streaming and tile-based streaming. Each streaming scheme presents a trade-off between the required minimum bandwidth to stream the 360º video content and the required maximum latency for such stream to be transmitted in time for the video to be correctly displayed.

Viewport-independent streaming streams the entire scene in uniform quality. The bitrate adaptation of the DASH client is made in similar manner to a 2D video, meaning that the representation requests for upcoming segments are made only according to the available bitrate. While the deployment for this approach is simple, a large part of the bandwidth is wasted on content that is not shown to the user. Tiles-based streaming is a type of viewport-dependent streaming where, in addition to being temporally segmented, 360º video can further be spatially partitioned into rectangular tiles in order to independently adjust the quality of each. The client requests the tiles corresponding to the viewing direction with a high quality, and the others with a low quality, or not at all. Viewport-only is a scheme that is not included in OMAF. This is a solution where the client sends a request to the server for the exact viewport to display to the user. Using this scheme, only the bandwidth required to transmit the pixels inside of the viewport is necessary. However, this comes at the expense of the strict latency requirements and the highest complexity at the server of all the presented schemes, since it needs to process a personalized viewport for each connected user.

C. 360º Video Streaming QoE

QoE is defined as "the degree of delight or annoyance of the user of an application or service. It results from the fulfillment of his or her expectations with respect to the utility and/or enjoyment of the application or service in the light of the user's personality and current state" [16]. Therefore, it becomes clear that measuring QoE is a very subjective matter. Because of this, service providers need good QoE models in order allow them to understand how to design an application and allocate resources in order to keep users satisfied. The existing methods can be divided into two categories: subjective and objective. Irregardless of which modelling technique is used, all QoE models provide a prediction of the perceived subjective quality of a video through some function or mapping. The metric that is normally predicted is the Mean Opinion Score (MOS), which ranges from 1 ("Bad") to 5 ("Excellent"). Subjective quality assessment consists in subjective tests performed on groups of human observers, who provide quality ratings. The final quality score is obtained by averaging the ratings given by multiple subjects into a MOS. Objective quality assessment models can provide a real-time in-service quality assessment and replace a human test group with a computational model. The goal of these models is to predict the MOS that would be obtained with subjective experiments as closely as possible, by using objective metrics [17].

D. Related Work

Several solutions have been recently proposed in the literature that leverage the benefits brought by edge-assisted 5G networks in order to solve some of the challenges of 360º video streaming. In [18], a viewport rendering solution at the edge of a mobile network is proposed, with the aim of reducing the bandwidth and latency required by 360º video streaming. In order to assess the role and value of edge computing to reduce latency and traffic at the radio access network, the authors deploy the 4G LTE test environment. A MEC server is deployed between the base station and the network core. In [19], a mobile edge assisted streaming system for 360º video is proposed in order to reduce the bandwidth and transmission delay, focusing on a live streaming scenario. This system, called MELiveOV, offloads some of the processing tasks from the device to the edge computing enabled 5G base stations. In [20], a scheme to optimize the network bandwidth using motion-predicted-based multicast to serve concurrent viewers is proposed, taking into account that the users are watching the same video and that most follow similar motion patterns in order to focus on the areas of interest. Therefore, the possibility of sending only a partial viewport to every user in a multicast manner emerges.

IV. Simulator Implementation

In order to assess and compare the viability and performances of the different 360º video streaming schemes, a previously developed simulator for LTE networks and 2D video was used as a starting point for the development of a new network simulator in MATLAB[1] [21]. The idea of this simulator is to replicate the state of a 5G network during a 360º video streaming session of a certain duration, with a certain number of connected users, and in which a certain streaming scheme is employed. The simulator then calculates the experienced conditions by each user throughout the session, and inputs this information into a QoE model. The users' satisfaction computed by this model is then used as a metric to evaluate the performance of each method. Another objective of this simulator is to analyze the performance gains enabled by deploying MEC server infrastructure on the edge

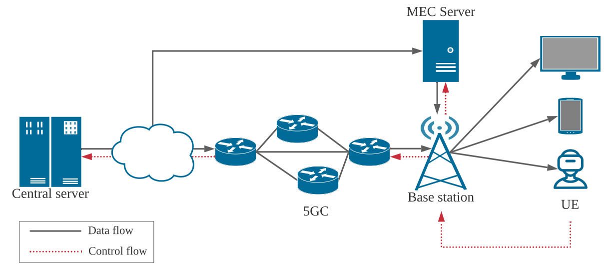

Fig. 2. Network topology.

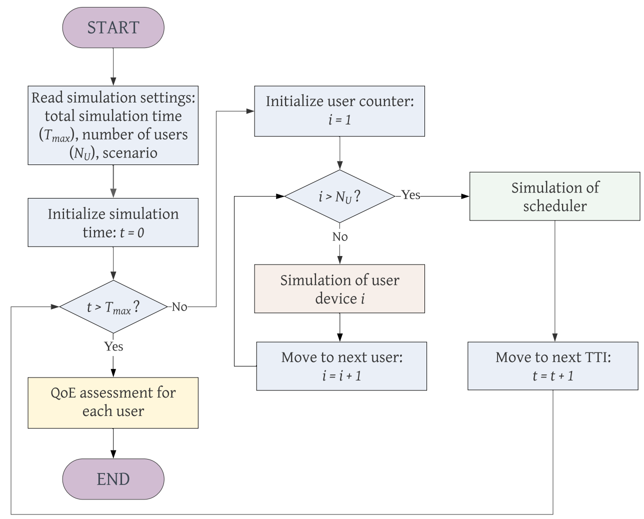

Fig. 3. General simulator workflow.

of the network. Figure 2 illustrates the implemented network topology. When a MEC server is not deployed, the client needs to fetch the requested segments from the central server, through the core network. With a MEC server, assuming the relevant segments are cached and/or assuming no backhaul limitations, the client can fetch the requested segments from the edge of the network, resulting in much smaller end-to-end latencies.

The simulator was developed in a modular form, in order to allow the reconfiguration of the various blocks and the implementation of different algorithms. The general workflow is presented in Figure 3.

First, the simulation settings are imported. The main settings include the total simulation time, which is equivalent to the length of the streaming session to be simulated, the number of connected users, which delivery scenario is being simulated, the 5G numerology configuration, among others. After the settings are imported, the simulator enters its main *for* loop. The number of iterations of this loop is equivalent to the number of TTIs one wishes to simulate, which is given by the total simulation time and the duration of each TTI, which depends on the used numerology. For each iteration, the state

[1]Simulator code available: https://github.com/miguel-loff/360_streaming_simulator

of each user's device is simulated: the viewport orientation is updated, the buffer level is updated, the rate adaptation algorithm may request new segments to the server and select their respective quality, and the CQI may be reported to the base station. Following the simulation of all users' devices, the resource scheduler is simulated. In each TTI, the scheduling algorithm distributes the available PRBs among the users that need to receive data. When the simulator reaches the predefined simulation time, the QoE is assessed for each user.

A. User Throughput

Although the simulator does not consider packet loss during the streaming session, it assumes that the channel quality experienced by the users varies throughout the simulation time. The channel quality, which is estimated by the UE and reported to the base station by means of the CQI, defines how many bits can be transmitted with each PRB. To calculate the instantaneous throughput, (1) is used.

The modulation order and coding rate depend on the reported CQI. A higher CQI will allow the base station to use a higher modulation order and code rate, resulting in a higher throughput. The values are used according to Table I. All the users report their CQIs to the base station every 5 ms. The used CQI value dataset, obtained through a simulation containing a single base station and 200 mobile users randomly roaming around the base station, was taken from [21].

B. Rate Adaptation Algorithm

After the initial buffering stage, in order for MPEG-DASH to work, there needs to be an algorithm that selects the quality of the subsequent segments according to the available bandwidth. The client also needs to determine when to request a new segment (i.e., when to call the rate adaptation algorithm). The client should not call the rate adaptation algorithm before a previously requested segment has finished downloading as it would cause conflicts of information. There is also the question of buffer overflow. Since the client's buffer capacity is not infinite, it does not make sense to request a new segment at a time when it would exceed the buffer's limit. Therefore, the rate adaptation algorithm should only be called to request the next segment when the client's buffer falls below a certain threshold, α. In addition to this, it is essential to consider that the timing in which the next segment request is made directly impacts the streaming session quality. Setting the threshold α to a higher level, which means the client would request the next segment sooner in relation to the instant the current segment finishes playing, means the client has more time to download the new segment before the buffer is depleted. However, when dealing with a viewport-dependent streaming scheme such as the tiles, partial tiles or viewport only schemes, it is important to remember that the high quality tiles or viewports are chosen at the time of the request. Since the users move their head during the time between the segment request and the segment playback, this means that requesting the segment sooner will result in a higher mismatch between the requested direction and the actual direction the user is viewing when the segment is played back. Therefore, the threshold α presents a trade-off between the risk of stalling and the risk of delivering an inaccurate viewing direction.

The implemented algorithm is an adaptation of the QoE-Enhanced Adaptation Algorithm over DASH (QAAD), presented in [22]. The rate adaptation algorithm consists of two parts: the throughput estimation and the bitrate selection. The throughput estimation is based on a segment-based scheme [23]. The estimated channel throughput is updated every time a new segment finishes downloading taking into account the size in bits of the downloaded segment and the time it took to download it, from the moment it was requested to the moment it finished downloading. In order to avoid big fluctuations, a weighted moving average scheme is used, with a weight factor of $\omega = 0.3$.

C. Video Sequences

In order for the simulator to fulfill the objective of realistically simulating the behavior of a 5G network in a 360º video streaming scenario, real bitrates need to be considered for the available video quality levels. Furthermore, in order for the simulator to also fulfill the objective of comparing different 360º streaming delivery schemes, the bitrates for the various schemes need to be comparable, as it is important that the video quality yielded by a given quality level in one scheme is comparable to the one yielded by the same level in another scheme. The following delivery schemes were studied [24]:

- Monolithic Equirectangular (MonoEqui) - It is based on the viewport-independent streaming scheme, using the equirectangular projection. The entire frame is encoded with the original resolution and with some target Quantization Parameter (QP). The target QP is varied to obtain different rate-distortion points and thus obtain different target quality levels and bitrates;
- Tiles OMAF Spatial Resolution (OMAF-SRes) - It is based on the tiles-based streaming scheme, using the equirectangular projection. Tiles are encoded with a high and low spatial resolution. The selection of the high spatial resolution tiles is made based on the coverage of the viewport. The low spatial resolution tiles have half the resolution of the high resolution tiles;
- Tiles OMAF Spatial Resolution Partial Delivery (OMAF-SRes-Partial) - This strategy is the same as OMAF-SRes, however only the tiles not covering the viewport (i.e., the low spatial resolution tiles) are not streamed;
- Viewport Only (Viewport-Only) - It is based on the viewport-only streaming scheme. The server keeps the original equirectangular video and encodes the requested viewports on demand. The transmitted frame has the resolution of the viewport and is encoded with some target QP in order to obtain different target quality levels;
- Viewport Only with Margin (Viewport-Only-Margin) - It modifies the Viewport-Only scheme in order to include a margin around the sent viewport. This way, if the user moves their head inside one segment, there is an area of tolerance before the viewport starts to lose coverage.

Each scheme is tested using three test sequences taken from the JVET dataset [25], *ChairliftRide*, *SkateboardInLot* and *KiteFlite*. All videos use the equirectangular projection, with a spatial resolution of 7680×3840 pixels, a temporal resolution of 30 frames/s and a duration of 10 seconds. In order to estimate the bitrate for each scheme and quality level, the V-PSNR metric is used in order to make sure that the V-PSNR difference between quality levels is the same and that the same quality level between different schemes has the same V-PSNR.

D. Scheduling Algorithm

After the rate adaptation algorithm makes the decision to fetch a new segment of a certain quality level, the client makes this request to the server. When the data arrives from the server, the base station needs to decide how to best distribute the available resources (i.e., PRBs) every Transmission Time Interval (TTI) among all the clients that are requesting data. It does so by means of a scheduling algorithm. Resource allocation for each user is based on the comparison of a per-PRB metric [26]. The k-th RB is allocated to the j-th user if its metric $m_{j,k}$ is the largest out of all the users, i.e., if it satisfies

$$m_{j,k} = \max_{i}\{m_{i,k}\}. \tag{2}$$

After comparing the behavior of several scheduling algorithms, the decision was made to implement the Proportional Fair (PF) algorithm [27]. This algorithm falls under a category known as channel-aware schedulers [28]. This type of scheduler takes advantage of the periodically sent CQI to estimate the maximum achievable throughput for the i-th user, t-th TTI and k-th RB, i.e. $r_k^i(t)$. The PF algorithm provides a good compromise between throughput and fairness among users. While it tries to maximize the total throughput, at the same time it tries to provide a minimum QoS to all users. The metric is defined as

$$m_{i,k}^{PF} = \frac{r_k^i(t)}{r^{(avg)} + \sum_{k=1}^{K} \delta_{ik} r_k^i(t)}, \tag{3}$$

where $r^{(avg)}$ is the averaged throughput experienced by the user up to the current TTI. The sum over all K PRBs in the denominator accounts for PRBs that have already been allocated to user i in the current TTI. δ_{ik} equals one if user i has been allocated PRB k, and equals zero otherwise. The effect of this is that, when one PRB is allocated to a user, the metric decreases, giving other users a higher chance to be served. With the PF scheduling algorithm, even users with bad channel conditions are served within a certain amount of time.

E. Latency

In order for the simulator to better reflect what happens in a real scenario, it is necessary to take into account the end-to-end latency that exists between the client requesting data and that data arriving at the base station. When a user makes a request, the scheduling algorithm waits L TTIs before it starts including that user into the PF algorithm. The larger the value of the value of L, the higher the risk of the client depleting the buffer. In the case of partial delivery scenarios such as OMAF-SRes-Partial or Viewport-Only, it also increases the risk of a high viewing direction mismatch because the user moved their head too much in the time between making the request and playing the corresponding segment.

The amount of end-to-end latency present is impacted by several factors and depends heavily on the implemented network. In the context of the simulator, in order to study the impact of the presence of MEC infrastructure on the users' QoE, the latency value, L, is set as 1 ms or 10 ms, depending on whether the existence MEC servers is being considered or not [29].

F. Viewport Trajectory

Except in the case of the MonoEqui scheme, the video that is sent from the server to the client has an area where the quality is higher. When the client makes the request for a new segment, it sends, along with this request, information on the requested tiles or viewports. This means that, due to the latency and to previously buffered video that still hasn't been played back, by the time the user sees the segment, they may have moved their head from the orientation it was in at the time of the request, causing the actual viewport and the requested viewport to be misaligned. In order to calculate the quality impact due to this, in each TTI, the angular difference between the actual viewport and that segment's requested viewport is calculated. Then, using the information in [24], the angular difference is used to estimate the corresponding quality impact. The quality impact, expressed as a gain in dB, is then used to calculate the adjusted V-PSNR by subtracting it to the original V-PSNR. It is then possible to map this adjusted V-PSNR to the corresponding adjusted quality level, using the video's RD curve.

It is also necessary to have information about the viewport trajectory along the streaming session. Therefore, it was decided to use a real head tracking dataset from [30]. This dataset is composed of 48 users. The participants are free to look around the video and their head motions are recorded during the viewing of these videos.

G. QoE Model

In order to assess whether the users are satisfied or not with the streaming session, a model that estimates the QoE of each user must be used. It was decided to use a parametric QoE model based on the work presented in [31]. Although this model was designed to calculate the QoE for 2D video, it was adapted and used since there is a lack of formal research into parametric QoE models for 360º videos. The QoE for user i is calculated as

$$\mathrm{QOE}_i = 5.67 \times \frac{\overline{q_i}}{q_{max}} - 6.72 \times \frac{\hat{q}_i}{q_{max}} + 0.17 - 4.95 \times F_i, \tag{4}$$

where $\overline{q_i}$ is the average served quality level, $\hat{q}_i$ is the standard deviation of the served quality level, both normalized to the

TABLE II
PERCEIVED QUALITY BY USER i WITH A GIVEN GLOBAL QOE.

QoE	Perceived quality
$4 \leq \mathrm{QoE}_i \leq 5$	Excellent
$3 \leq \mathrm{QoE}_i < 4$	Good
$2 \leq \mathrm{QoE}_i < 3$	Fair
$1 \leq \mathrm{QoE}_i < 2$	Poor
$0 \leq \mathrm{QoE}_i < 1$	Bad

number of available quality levels in the server, q_{max}, and F_i models the influence of the rebuffering events and is calculated as

$$F_i = \frac{7}{8} \times \max\left(\frac{\ln(\phi_i)}{6} + 1, 0\right) + \frac{1}{8} \times \frac{\min(\psi_i, 15)}{15}, \quad (5)$$

where ϕ_i is the freeze frequency and ψ_i is the average freeze duration. The model was adapted to take the initial buffering delay into account, by including it in the calculation of the mean rebuffering duration. The model also has to be adapted to take the quality impact due to the viewport mismatch into account. Therefore, it was decided that the $\overline{q_i}$ and $\hat{q}_i$ should be calculated not based on the served video quality levels, but on the adjusted quality levels calculated from taking the quality impact of the viewport mismatch into account. For the OMAF-SRes-Partial and Viewport-Only schemes, it was decided that a blank area in the viewport should impact the QoE at the level of the stalling event. Therefore, whenever the blank area surpasses 10%, it counts towards the freeze frequency, ϕ_i, and average freeze duration, ψ_i.

With this model, it is possible to calculate the global QoE at the end of the streaming session. By looking at the global QoEs of different streaming schemes, it is possible to evaluate and compare the corresponding users' satisfaction. Table II relates the QoE with the perceived quality.

V. SIMULATIONS AND RESULTS

A. Simulation Parameters

In order to study how the different streaming schemes compare against each other, it is necessary to run the developed simulator several times, changing the used scheme each time. However, in order to make an accurate comparison, all the other simulation parameters need to be the same, so that any changes in the outcome are due only to the used streaming scheme. For this end, each simulation consists in a streaming session with a duration of 3 minutes. During this period, a fixed number of users are simultaneously streaming 360º video using one of the aforementioned streaming schemes with a fixed end-to-end latency. Each user is streaming one of the three available test sequences. The allocation of each test sequence to each user is made randomly at the beginning of the simulation using a uniform distribution. The test sequences are available in the server in 1 second segments, except in the case of the Viewport-Only scheme, where it is considered that each segment has 40 ms, which is the duration of a frame in a 25 frames/s video.

The Monolithic scheme requests 5 segments initially and 5 segments after a stall, all with the minimum quality available. The other schemes request 1 segment initially and 1 segment after a stall. Each user starts requesting their initial segments at a random time between 0 and 200 ms. After receiving the initial segments with minimum quality, the DASH client selects the quality of the subsequent segments using the QAAD algorithm. This algorithm has two parameters, μ and σ, that need to be set. The marginal buffer length, μ, defines the minimum length the buffer needs to have in order for the requested quality level to be increased. The minimal buffer length, σ, defines the critical buffer length that the client always tries to keep, triggering a quality decrease if that condition is not met. Considering that the buffer threshold that triggers the request of the next segment, α, influences the range of values the buffer can have, it makes sense to set μ and σ dynamically according to this value. Therefore, μ is set to 80% of α and σ is set to 20% of α.

Each user is allocated one of the available CQI profiles. This allocation is made as a random permutation, meaning that each user is randomly assigned one CQI profile without repetitions. The viewport trajectory dataset is allocated to each user in the same way.

It was decided that the simulator would use a network configuration with 2x2 MIMO, 15 KHz SCS, and 20 MHz bandwidth. This means that there are 106 PRBs available per TTI, and each TTI is 1 ms in length [7]. The TTI is the basic time unit of the simulator. As such, with a 3 minute simulation, 180000 TTIs are simulated.

B. Simulation Outputs

In order to assess the number of satisfied and non-satisfied users, the main output of the simulation is the final QoE of each user. The term "final QoE" refers to the overall QoE level a user has at the end of the streaming session. The used QoE model has two main factors contributing to the final value: metrics related to the radio conditions of the channel, such as average served quality, stall frequency, among others, and metrics related to the viewport mismatch due to the user's head movement, such as the average adjusted quality. These two factors are independent, meaning that, for example, the user can have a poor final QoE because the video was not served correctly due to the channel conditions, or because the video was correctly served but the head movement caused a high level of viewport mismatch throughout the streaming session. Since it is important to evaluate the impact that each of the main factors has on the final QoE, this metric is computed in two steps: first, only the metrics related to the radio conditions are taken into account, producing the QoE_{radio} metric. Then, the metrics related to the viewport mismatch are taken into account on top of the radio metrics, producing the QoE_{final} metric. This way, it is possible to assess to which extent each of the two factors contribute to the final QoE.

TABLE III
OPTIMAL α FOR EACH SCENARIO AND USER CAPACITY.

Scheme	Latency	
	10 ms	1 ms
MonoEqui	6000 ms	6000 ms
OMAF-SRes	1000 ms	1000 ms
OMAF-SRes-Partial	100 ms	100 ms
Viewport-Only	15 ms	3 ms
Viewport-Only-Margin	15 ms	4 ms

Performance is assessed in terms of the number of users that are satisfied and/or non-satisfied at the end of a streaming session, out of all the connected users. It is considered that a user is satisfied if their QoE_{final} is above a certain limit S ($QoE_{final} \geq S$) and non-satisfied if their QoE_{final} is below a certain limit NS ($QoE_{final} \leq NS$), where $S > NS$. For a given number of connected users, the percentage of satisfied users for that S is denoted by $\%_S$ and the percentage of non-satisfied users for that NS is denoted by $\%_{NS}$. Naturally, these values depend heavily on the chosen simulation parameters. However, if the parameters are kept constant, the relative performance can be used to make assessments.

C. Sensitivity Analysis

It is necessary to perform a sensitivity analysis to the buffer threshold that triggers the request of the next segment, α. Because this value is different for different end-to-end latency values, L, it is also necessary to perform the analysis taking this parameter into account. In order to do this, a series of values of α are simulated for each of the values of latency, L, 10 ms and 1 ms, which are equivalent to having MEC infrastructure deployed or not. This simulation is performed for various numbers of connected users, using the Monte Carlo method, in order to assess the performance limits of each streaming scenario. The methodology for determining the best α for each scenario is to assess the capacity for each value. The capacity is given by the worst value between the maximum number of connected users for which 90% have a $QoE_{final} \geq$ 3 (S = 3; $\%_S$ = 90%) and the maximum number of connected users for which 5% have a $QoE_{final} \leq$ 2 (NS = 2; $\%_{NS}$ = 5%). Table III presents the results.

D. Schemes Comparison

The different schemes are compared against each other regarding the number of satisfied and non-satisfied users. In order to do this, the best α value for each scheme is used.

1) Comparison of Satisfied Users (S = 3): Tables IV and V present for each scheme the number of satisfied users (S = 3), the total number of connected users and the capacity gains relative to MonoEqui, for S = 3 and $\%_S$ = 90%, 80% and 70%, for a latency of 10 ms and 1 ms, respectively. The number of satisfied users is calculated by multiplying the percentage by the number of connected users at that percentage. The scheme that supports the most users with a latency of 10 ms

TABLE IV
CAPACITY OF ALL SCHEMES. S = 3, $\%_S$ = 90%, 80%, 70%. L = 10 MS.

$\%_S$	Scheme (Satisfied users/Connected users/Gain)				
	MonoEqui	**OMAF-SRes**	**OMAFSRes-Partial**	**Viewport-Only**	**Viewport-Only-Margin**
90%	**43.71** 48.57 (0.00%)	**45.10** 50.11 (+3.18%)	**0.00** 0.00 (-100.00%)	**0.00** 0.00 (-100.00%)	**126.00** 140.00 (+188.26%)
80%	**40.90** 51.13 (0.00%)	**40.97** 51.21 (+0.17%)	**0.00** 0.00 (-100.00%)	**0.00** 0.00 (-100.00%)	**113.06** 141.33 (+176.43%)
70%	**36.67** 52.38 (0.00%)	**36.62** 52.31 (-0.14%)	**0.00** 0.00 (-100.00%)	**0.00** 0.00 (-100.00%)	**99.87** 142.67 (+172.35%)

TABLE V
CAPACITY OF ALL SCHEMES. S = 3, $\%_S$ = 90%, 80%, 70%. L = 1 MS.

$\%_S$	Scheme (Satisfied users/Connected users/Gain)				
	MonoEqui	**OMAF-SRes**	**OMAFSRes-Partial**	**Viewport-Only**	**Viewport-Only-Margin**
90%	**43.71** 48.57 (0.00%)	**45.10** 50.11 (+3.18%)	**0.00** 0.00 (-100.00%)	**0.00** 0.00 (-100.00%)	**118.29** 131.43 (+170.62%)
80%	**40.92** 51.15 (0.00%)	**40.97** 51.21 (+0.12%)	**0.00** 0.00 (-100.00%)	**128.26** 160.32 (+213.44%)	**106.78** 133.47 (+160.94%)
70%	**36.71** 52.44 (0.00%)	**36.62** 52.31 (-0.25%)	**0.00** 0.00 (-100.00%)	**113.26** 161.94 (+208.25%)	**94.86** 135.51 (+158.40%)

is Viewport-Only-Margin. For a latency of 1 ms, Viewport-Only supports more users than Viewport-Only-Margin for a percentage of satisfied users of 80% and 70%. The Viewport-Only scheme only has a number of satisfied users higher than zero for a latency of 1 ms and percentage of satisfied users of 80% and 70%. It is also possible to observe that the OMAF-SRes scheme generally has a higher number of satisfied users than MonoEqui, except when the percentage of satisfied users is 70%, for both values of latency. The OMAF-SRes-Partial has zero satisfied users for any percentage of satisfied users and latency values.

2) Comparison of Very Satisfied Users (S = 4): Tables VI and VII present for each scheme the number of very satisfied users (S = 4), the total number of connected users and the capacity gains relative to MonoEqui, for S = 4 and $\%_S$ = 90%, 80% and 70%, for a latency of 10 ms and 1 ms, respectively. The Viewport-Only-Margin scheme does not work with this high level of user satisfaction with a latency of 10 ms, except when the percentage of satisfied users is 70%. For S = 4, the performance gains of OMAF-SRes over MonoEqui are also higher across all percentages of satisfied users and latency values. The number of satisfied users with these two schemes are similar for both latency values. Also, the Viewport-Only scheme does not work with S = 4.

3) Comparison of Non-Satisfied Users (NS = 2): The comparison of the various schemes for NS = 2 is done as a function of the number of very satisfied users (S = 4). As such, Tables VIII and IX presents for each scheme the number of connected users for S = 4, $\%_S$ = 90%, 80% and

TABLE VI
CAPACITY OF ALL SCHEMES. S = 4, $\%_S$ = 90%, 80%, 70%. L = 10 MS.

$\%_S$	Scheme (Satisfied users/Connected users/Gain)				
	MonoEqui	OMAF-SRes	OMAF-SRes-Partial	Viewport-Only	Viewport-Only-Margin
90%	**27.00**	**40.50**	**0.00**	**0.00**	**0.00**
	30.00	45.00	0.00	0.00	0.00
	(0.00%)	(+50.00%)	(-100.00%)	(-100.00%)	(-100.00%)
80%	**24.89**	**40.19**	**0.00**	**0.00**	**0.00**
	31.11	50.24	0.00	0.00	0.00
	(0.00%)	(+61.47%)	(-100.00%)	(-100.00%)	(-100.00%)
70%	**22.55**	**36.02**	**0.00**	**0.00**	**98.23**
	32.22	51.46	0.00	0.00	140.33
	(0.00%)	(+59.73%)	(-100.00%)	(-100.00%)	(+335.61%)

TABLE VII
CAPACITY OF ALL SCHEMES. S = 4, $\%_S$ = 90%, 80%, 70%. L = 1 MS.

$\%_S$	Scheme (Satisfied users/Connected users/Gain)				
	MonoEqui	OMAF-SRes	OMAFSRes-Partial	Viewport-Only	Viewport-Only-Margin
90%	**25.50**	**40.80**	**0.00**	**0.00**	**108.00**
	28.33	45.33	0.00	0.00	120.00
	(0.00%)	(+60.00%)	(-100.00%)	(-100.00%)	(+323.53%)
80%	**29.34**	**40.29**	**0.00**	**0.00**	**105.18**
	36.67	50.36	0.00	0.00	131.48
	(0.00%)	(+37.32%)	(-100.00%)	(-100.00%)	(+258.49%)
70%	**22.44**	**36.10**	**0.00**	**0.00**	**93.33**
	32.05	51.57	0.00	0.00	133.33
	(0.00%)	(+60,87%)	(-100.00%)	(-100.00%)	(+315.91%)

TABLE VIII
NON-SATISFIED USERS OF S = 4, $\%_S$ = 90%, 80%, 70%. L = 10 MS.

$\%_S$	Scheme (Connected users/$\%_{NS}$/Non-satisfied users)				
	MonoEqui	OMAF-SRes	OMAFSRes-Partial	Viewport-Only	Viewport-Only-Margin
90%	30.00	45.00			
	1.00%	4.00%	N/A	N/A	N/A
	0.30	**1.8**			
80%	31.11	50.24			
	1.22%	10.21%	N/A	N/A	N/A
	0.38	**5.13**			
70%	32.22	51.46			140.33
	1.44%	21.43%	N/A	N/A	5.67%
	0.46	**11.03**			**7.96**

TABLE IX
NON-SATISFIED USERS OF S = 4, $\%_S$ = 90%, 80%, 70%. L = 1 MS.

$\%_S$	Scheme (Connected users/$\%_{NS}$/Non-satisfied users)				
	MonoEqui	OMAF-SRes	OMAFSRes-Partial	Viewport-Only	Viewport-Only-Margin
90%	28.33	45.33			120.00
	0.83%	3.73%	N/A	N/A	**0**
	0.24	**1.69**			0.00%
80%	36.67	50.36			131.48
	2.33%	10.35%	N/A	N/A	8.25%
	0.85	**5.21**			**10.85**
70%	32.05	51.57			133.33
	1.41%	21.6%	N/A	N/A	17.32%
	0.45	**11.14**			**23.09**

70%, the corresponding number of non-satisfied users, and the corresponding $\%_{NS}$ with NS = 2, for a latency of 10 ms and 1 ms, respectively. As it is possible to observe, some values cannot be computed because the corresponding capacity in terms of very satisfied users with S = 4 is zero. For a latency of 10 ms, the scheme that presents the lowest numbers of non-satisfied users is MonoEqui. For OMAF-SRes, the number of non-satisfied users is low for when the number of connected users is the one that supports S = 4, $\%_S$ = 90% and 80%. For 70%, the number of non-satisfied users is much higher for only 1.22 more connected users. For a latency of 1 ms, the number of non-satisfied users is comparable to the latency value of 10 ms for MonoEqui and OMAF-SRes. For Viewport-Only-Margin, although the number of non-satisfied users when the number of connected users is the one that supports S = 4, $\%_S$ = 70% is much higher, when $\%_S$ = 90%, the number of non-satisfied users is zero, which is the only case where this happens.

E. Results Summary and Analysis

Based on Tables IV to IX, this section presents the most relevant results and conclusions:

- In a scenario where MEC infrastructure is not deployed, the OMAF-SRes scheme is better than MonoEqui at satisfying users, especially when aiming at high satisfaction levels. It achieves 40.50 very satisfied users for $\%_S$ = 90%, which is 50% more than MonoEqui. The only drawback is that the corresponding number of non-satisfied users will be 1.69 instead of 0.24. However, it is considered that the relative improvement in very satisfied users is enough to justify this drawback.
- When considering the MonoEqui and OMAF-SRes schemes, the gains of deploying MEC infrastructure are very small, if existent at all. As previously discussed, a latency reduction in the order of 10 ms has a residual impact when the buffer threshold is in the order of hundreds of milliseconds.
- The OMAF-SRes-Partial scheme should not be considered since it does not support any satisfied or very satisfied users for any $\%_S$. As discussed previously, the fact that the video area is small and the segments are 1 second long makes this an unsuitable scheme.
- The Viewport-Only scheme does not work without MEC infrastructure. With MEC infrastructure deployed, it can only achieve 80% of satisfied users (S = 3). However, at this level, it is possible to achieve 128.26 satisfied users, which is the most out of any scenario. This corresponds to a total of 160.32 connected users and 11.7 non-satisfied users ($\%_{NS}$ = 7.3%).
- Although the Viewport-Only-Margin supports around 100 satisfied users (S = 3) for both latency values, when considering very satisfied users (S = 4), it only supports 70% of users when there is no MEC infrastructure, with 7.96 non-satisfied users. The only way for Viewport-Only-Margin to support 90% of users with an ”Excellent” perceived quality is to deploy MEC infrastructure. If MEC infrastructure is present, not only is it possible to support 108 very satisfied users, the highest value for S

= 4, it is also possible to achieve zero non-satisfied users, being the only configuration that achieves this.

VI. Conclusions

This work aimed at studying how different 360º video streaming schemes compared against each other in terms of their performance and resulting user satisfaction. Another goal was to study how Multi-Access Edge Computing (MEC) may be beneficial for these types of applications. It was found that, when no MEC infrastructure is deployed, the OMAF-SRes is a better streaming scheme than the commonly used MonoEqui, supporting 40.50 very satisfied users while keeping the number of non-satisfied users fairly low. When the 5G network is leveraged by MEC servers, making a remote viewport rendering solution viable, 120 very satisfied users can be supported with Viewport-Only-Margin, which is three times better than what OMAF-SRes can achieve without MEC, while achieving zero non-satisfied users. While it is true that Viewport-Only and Viewport-Only-Margin can support high numbers of satisfied users, it is important to remember that today's paradigm of high quality multimedia streaming applications means that users are extremely demanding in terms of the quality of experience they receive from the services they pay for. Furthermore, users are particularly sensitive to 360º video, suffering from motion sickness and other effects when the quality is below standard. Therefore, mobile network operators and other stakeholders must aim to deliver QoEs higher than 3. Taking this into account, it is concluded that there is a clear advantage to using MEC infrastructure in 5G networks in order to leverage 360º video streaming.

For future work, it would be interesting to perform subjective studies on the perceived quality of 360º videos using the different delivery schemes in order to develop more accurate QoE models. The simulator could be further developed in order to include new capabilities such as coexisting with other latency-critical services that compete in terms of prioritization with the 360º video delivery or having users give up from their streaming session if their QoE stays too low for more than a certain amount of time. It would also be interesting to study how mobile network operators can operationalize the deployment of MEC infrastructure in terms of location and number of servers in order to take advantage of the performance gains while keeping their Capital Expenditures (CAPEX) and Operational Expenditures (OPEX) at feasible levels.

References

[1] Ericsson. "Ericsson Mobility Report". Ericsson, November 2020.
[2] Huawei. "Preparing for a Cloud AR/VR Future". Huawei.
[3] Cisco. "Cisco Visual Networking Index: Global Mobile Data Traffic Forecast Update, 2017-2022". Cisco, February 2019.
[4] M. Zink, R. Sitaraman and K. Nahrstedt. "Scalable 360° Video Stream Delivery: Challenges, Solutions, and Opportunities". Proceedings of the IEEE, vol. 107, no. 4, pp. 639-650, April 2019.
[5] H. Holma, A. Toskala and T. Nakamura. "5G technology: 3GPP new radio". 1st edition, Wiley, 2020.
[6] M. Shafi, W. Shuai and M. U. Younus "5G: A Tutorial Overview of Standards, Trials, Challenges, Deployment, and Practice". IEEE Journal on Selected Areas in Communications, vol. 35, no.6, pp. 1201-1221, June 2017.
[7] 3GPP. "Base Station (BS) radio transmission and reception". 3GPP, June 2019.
[8] J. Campos. "Understanding the 5G NR Physical Layer". Keysight Technologies, November 2017.
[9] 3GPP. "Physical channels and modulation". 3GPP, June 2019.
[10] S. Ahmadi. "5G NR: Architecture, Technology, Implementation, and Operation of 3GPP New Radio Standards". Elsevier Science, Cambridge, 2019.
[11] 3GPP, "User Equipment (UE) radio access capabilities". 3GPP, October 2018.
[12] 3GPP. "Physical layer procedures for data". 3GPP, October 2018.
[13] N. Abbas et al. "Mobile Edge Computing: A Survey". IEEE Internet of Things Journal, vol. 5, no. 1, pp. 450-465, February 2018.
[14] A. Yaqoob, T. Bi and G. Muntean. "A Survey on Adaptive 360º Video Streaming: Solutions, Challenges and Opportunities". 2019 Data Compression Conference (DCC), March 2019.
[15] I. Sodagar. "The MPEG-DASH Standard for Multimedia Streaming Over the Internet". IEEE MultiMedia, vol. 18, no. 4, pp. 62-67, April 2011.
[16] K. Brunnström et al. "Qualinet White Paper on Definitions of Quality of Experience". 2013.
[17] Z. Akhtar and T. H. Falk. "Audio-Visual Multimedia Quality Assessment: A Comprehensive Survey". IEEE Access, vol. 5, pp. 21090-21117, September 2017.
[18] S. Mangiante et al. "VR is on the Edge: How to Deliver 360° Videos in Mobile Networks". Proceedings of the Workshop on Virtual Reality and Augmented Reality Network - VR/AR Network '17, August 2017.
[19] X. Hu et al. "Mobile Edge Assisted Live Streaming System for Omnidirectional Video". Mobile Information Systems, vol. 2019, pp. 1-15, May 2019.
[20] Y. Bao et al. "Motion-Prediction-Based Multicast for 360-Degree Video Transmissions". 2017 14th Annual IEEE International Conference on Sensing, Communication, and Networking (SECON), San Diego, USA, June 2017. IEEE.
[21] F. Rodrigues et al. "QoE-Aware Scheduling Algorithm for Adaptive HTTP Video Delivery in Wireless Networks". Wireless Communications and Mobile Computing, vol. 2018, pp. e9736360, September 2018.
[22] D. Suh, I. Jang and S. Pack. "QoE-enhanced adaptation algorithm over DASH for multimedia streaming". The International Conference on Information Networking 2014 (ICOIN2014), Phuket, Thailand, February 2014. IEEE.
[23] T. Thang et al. "Adaptive streaming of audiovisual content using MPEG DASH". IEEE Transactions on Consumer Electronics, vol. 58, no. 1, pp. 78-85, February 2012.
[24] M. Luís. "Viewport Adaptive Streaming for Omnidirectional Video Delivery". MSc Thesis, Instituto Superior Técnico, Lisbon, Portugal, June 2020.
[25] J. Boyce et al. "JVET-H1030: JVET common test conditions and evaluation procedures for 360º video". JVET, July 2018.
[26] F. Capozzi et al. "Downlink Packet Scheduling in LTE Cellular Networks: Key Design Issues and a Survey". IEEE Communications Surveys & Tutorials, vol. 15, no. 2, pp. 678-700, 2013.
[27] M. K. Muller, S. Schwarz and M. Rupp. "QoS investigation of proportional fair scheduling in LTE networks". 2013 IFIP Wireless Days (WD), Valencia, Spain, November 2013. IEEE.
[28] I. Sousa, M. P. Queluz and A. Rodrigues. "A survey on QoE-oriented wireless resources scheduling". Journal of Network and Computer Applications, vol. 158, pp. 102594, May 2020.
[29] G. Brown. "New Transport Network Architectures for 5G RAN". Fujitsu.
[30] C. Wu et al. "A Dataset for Exploring User Behaviors in VR Spherical Video Streaming". Proceedings of the 8th ACM on Multimedia Systems Conference, Taipei, Taiwan, June 2017. ACM.
[31] S. Petrangeli et al. "QoE-Driven Rate Adaptation Heuristic for Fair Adaptive Video Streaming". ACM Transactions on Multimedia Computing, Communications, and Applications, vol. 12, no. 2, pp. 1-24, March 2016.
[32] M. D. Byrne. "How Many Times Should a Stochastic Model Be Run? An Approach Based on Confidence Intervals".