Macrothink Institute™

Network Protocols and Algorithms
ISSN 1943-3581
2017, Vol. 9, No. 3-4

Intelligent Algorithm for Enhancing MPEG-DASH QoE in eMBMS

Miran Taha[1,2], Jose M. Jimenez[1], Alejandro Canovas[1], Jaime Lloret[1]

[1]Integrated Management Coastal Research Institute, Polytechnic University of Valencia.

C/ Paranimf nº 1, Grao de Gandía – Gandia, Valencia, Spain

[2]Department of Computer Science, University of Sulaimani,

Tasluja Street, n 1, 46001, Kurdistan region, Iraq

E-mail: miab2@doctor.upv.es, jojiher@dcom.upv.es, alcasol@posgrado.upv.es, jlloret@dcom.upv.es

Received: September 10, 2017 Accepted: December 10, 2017 Published: December 31, 2017

DOI: 10.5296/npa.v9i3-4.12573 URL: http://dx.doi.org/10.5296/npa.v9i3-4.12573

Abstract

Multimedia streaming is the most demanding and bandwidth hungry application in today's world of Internet. MPEG-DASH as a video technology standard is designed for live or on-demand streams in Internet to deliver better quality content with the fewest dropouts and least possible buffering. Hybrid architecture of DASH and eMBMS have attracted a great attention from the telecommunication industry and multimedia services. It is employed in response to the immense demand of multimedia traffic. However, handover and limited resources are highly affected when they lose segments of the adaptive video streaming while the users are receiving the eMBMS service. It is a problem for service and network providers when they deliver the service because it produces an adverse impact on Quality of Experience (QoE). In this paper, we show a case study to evaluate MPEG-DASH QoE in eMBMS service, by measuring the Quality of Service (QoS) metrics that influence the QoE. We observe the objective metrics like stalling (number, duration and place), buffer length and accumulative video time to assess the QoE. A smart algorithm is developed to predict the rate of the lost segments in multicast adaptive video streaming. The algorithm uses an estimation decision to recover the lost segments from one of the content providers. According to the obtained results based on our proposed algorithm, the user perceived latency is highly decreased in comparison to the traditional approach of MPEG-DASH multicast and unicast when there is high number of users.

Keywords: eMBMS, LTE, MPEG-DASH, Proxy caching, QoE

1. Introduction

Mobile data traffic is growing day by day due to the increase of user subscriptions and growth in the volume of data [1]. Therefore, Global Mobile Suppliers Association (GSA) highlights that 790 operators invest in Long-Term Evolution (LTE) in 201 countries, which launched 581 LTE, LTE-Advanced or LTE-Advanced Pro commercially in 186 countries [2]. The mobile networks always evolve to provide better services, higher frequency ranges, and a higher Quality of Service (QoS). In this way, the operators can have more agile and flexible networks, which allow them to face new challenges in the future. The Ericsson report [1] indicates that mobile operators are introducing Gigabit LTE networks. These Gigabit LTE networks will allow operators to achieve greater coverage and speed data for users. Consequently, Version 9 of LTE is called enhanced Multimedia Broadcast Multicast Services (eMBMS) [3]. The use of MBMS service in networks has the purpose of broadcasting services of broadcast and multicast though a cellular network and allows operators to deliver content simultaneously to multiple end users. In mobile networks, transmitting video through employing adaptive technique can provide the user a video service with better Quality of Experience (QoE) [4]. By using standards for adaptive streaming over HTTP such as Dynamic Adaptive Streaming over HTTP (DASH) [5], the DASH user can playback on-demand and live streaming video with great quality even though the network is quite saturated.

Therefore, eMBMS provides broadcast multimedia services through the LTE network, which combines multicast and unicast. In eMBMS service, it is not usual to employ the application layer protocol based on Transmission Control Protocol (TCP) when transmitting video streaming over the multicast channel to a high amount of users. This happens because TCP has high resource usage and it becomes inefficient for network services. File Delivery over Unidirectional Transport (FLUTE) is used to multicast of the videos [6]. In eMBMS, HTTP is used to recover these lost segments over unicast channel in order to integrate the playback process. Therefore, this mechanism improves the quality of service (QoS) and enhances the Quality of Experience (QoE) [7].

In mobile wireless networks, the bottleneck in the wireless channel and the limited available resources of the system are impacted highly when the segments of video multicast streaming are dropped. Moreover, it is need to retrieve those lost segments through the unicast channel which increases the arrival time of segments and depletes the application's buffer. This creates a trouble when rendering the video at the client side producing a buffer underrun and stalls when displaying the video.

As a result, a number of researches are investigating how to improve QoE in LTE networks when the MBMS service is used. Tara Ali in [8] proposed a rate adaptive solution for multimedia streaming in an interference scenario based HetNet. The solution is based on power control in a macro Femto cell through the allocation of an appropriate number of resource blocks to the base layer of the video streaming traffic. The approach is used to guarantee the QoS of the video streaming in mobile networks. Therefore, a QoE-aware radio resource management (RRM) framework is proposed in [9], which allows the network operator to further enhance the video capacity. Their results demonstrated that there is a significant potential to optimize the video capacity through QoE awareness both at the application level and radio access network (RAN) level.

The aim of our work is to evaluate the performance of QoE of adaptive video transmission in LTE networks [10] by providing a case study for observing the objective metrics in order to provide a dataset of experiments. Therefore, we develop a smart algorithm to predict the rate of segments that are lost in the multicast channel. The algorithm brings the lost segments to be closer to the clients and it provides estimation decision to recover the lost segments through a unicast channel with minimum delay

The rest of paper is structured as follows. Section 2 presents the background of MPEG-DASH and the most relevant works related to our study. The hybrid architecture of MPEG-DASH and eMBMS in LTE networks are presented in section 3. In section 4, we describe the proposed algorithm. Section 5 shows the QoE effective metrics and performance evaluation according to proposed algorithm. Section 6 presents conclusion and future work.

2. Related Work

This section presents a background on MPEG-DASH and states the current problems of multicast and unicast video streaming in LTE networks.

In adaptive video streaming technique, the source video at the content provider side is encoded with multiple quality rates, which is represented by different representation layers and each representation is divided into many chunks sizes [11]. Each chunk size has a period of duration of video playback such as on-demand movies or lives streaming events. When the clients request to receive the service from the content provider, the application player is available on the client side and it would be ready to select the video chunks according to current network availability, client's CPU usage and memory usage. While client's application starts to playback, the video chunks are requested by the client to maintain the desired buffer level. In this way, if the client's downlink drops off, the client buffer becomes empty and causes an interruption of the video playback.

In order to avoid stalling or rebuffering, it demands chunks sizes with lower representation. Consequently, the user may have a quality switching between different representations [12]. The quality switching helps to avoid video interruption playback. Therefore, there are many companies that have their own proprietary implemented adaptive video streaming system such as Apple HLS, Adobe HDS, Microsoft (MSS) and MPEG-DASH ISO/IEC standard [13]. They are using the same mechanism of adaptive video streaming system, but with different characteristics.

There are previous works about LTE networks on MBMS, such as the one presented by Lecompte and Gabin [14], which that describes the relevant use cases for eMBMS in terms of service. They gave a tutorial on eMBMS, in particular highlighting the evolution over MBMS. The scope comprises the radio access, core network, and service layer.

Lohmar et al. explained in [15], that Ericsson showed demonstrations of LTE Broadcast with eMBMs in international trades. These demos showed the solution’s potential to create new business models for telcos and ensure consistent QoS; even in very densely populated places like sports venues.

Nguyen et al. in [16] evaluated and validated the performance of eMBMS following the Third Generation Partnership Program (3GPP) standard (release 10) implemented in the context of the OpenAirInterface SDR platforms. The obtained results showed that eMBMS performance in the OpenAirInterface satisfies the requirement of 3GPP standard in terms of Block Error Rate (BLER).

Stephanakis et al. [17] proposed a cache-accelerated approach in order to reduce the backhaul traffic, that associates distributed cache deployed at eMBMS Gateways per service area. Also, they presented a simple algorithm that determines the optimal cache size in a mixture of chain ON/OFF modeled services.

In [18], Pande et al. detailed some challenges in delivery of multimedia content over 4G networks for several application scenarios. They classified mobile video applications into four broad categories and analyze the current trends, issues, and opportunities of mobile video delivery in cellular networks.

We can also find works related to adaptive streaming, QoE and DASH. For example, related to adaptive algorithms, in [19], Taha et al. proposed an SDN-based TCP throughput management algorithm to provide fairness system for competing users over a wireless network. This wireless network is used by adaptive video streaming providers to take advantage of the HTTP protocol for deliver video contents to the applications of the end users.

Garcia et al [20] specified the aspects that are most annoying to Polimedia platform users and they provided some recommendations to improve the QoE of Polimedia users. Polimedia was developed by Polytechnic University of Valencia to provide multimedia to the university community. It uses techniques MPEG DASH adaptive streaming over HTTP.

Stockhammer and Luby [21] focused on DASH application as an enabler to address a significant portion of mobile video streaming focusing on overload situations, variable bandwidth and power consumption issues. Also, the applicability of DASH formats for multicast distribution in MBMS is shown. Finally, the usage of DASH in hybrid unicast/multicast is introduced to provide a flexible extension to support different business and delivery models.

In [22], Guo et al. proposed a hybrid transmission strategy for DASH (HTD) in LTE network, which is considered both unicast and multicast modes for DASH. The optimization problem is formulated as a Mixed Binary Integer Programming (MBIP) problem, and a two-level greedy algorithm is proposed, which could improve the QoE of wireless DASH users, and save the wireless resources in LTE network. Their simulation results demonstrated that their scheme achieves better performance than traditional single transmission mode in the literature.

Belda et al in [23] describes how File Delivery over Unidirectional Transport (FLUTE) and Dynamic Adaptive Streaming over Hypertext Transfer Protocol (DASH) can be used to provide mobile video streaming services over broadcast wireless networks. In their proposal, the protocol FLUTE is adapted to provide video streaming services in crowded environments. Moreover, their paper shows that FLUTE and DASH can be seamlessly integrated into a hybrid broadcast/unicast streaming technology, providing flexibility to trade off PSNR and bandwidth depending on the conditions of the mobile network.

Our approach proposed in this paper is different from other existing approaches, we provide smart algorithm in proxy caching points in order to reduce the rate of lost segments of adaptive multicast streaming in LTE networks. Moreover, the algorithm detects the network behavior and video characteristics in order to specify the lost segment to be recovered in the transmission with minimum latency. This leads to improve QoE in mobile users.

3. eMBMSArchitecture and proxy caching

In telecommunication, LTE is designed for high-speed wireless communication. eMBMS provides broadcast multimedia services through the LTE network, which has unicast physical downlink shared channels (PDSCH) and physical multicast channels (PMCH) services in the same LTE frame [14]. The network components are included to provide eMBMS operation as shown in Figure 1. They are the following ones:

- Broadcast Multicast Service Center (BMSC): This point provides direct interface for Content Provider and the Evolved Packet Core. It supplies eMBMS service with scheduling. This entity announces services to mobile users and plays the role of traffic shaping, authorizing request between provider and user equipment, users, allocates bearer service, and establishes and terminates MBMS bearer resources. It imposes SYNC protocol to synchronize the transmitted data among the Evolved Terrestrial Radio Access Network (E-UTRAN) and terminates the synchronization (SYNC) protocol over the M1 interface. The SYNC protocol keeps a specific header to IP packets, which is providing time stamps and session information.

- Multimedia Broadcast Multicast Services Gateway (MBMS GW): This component is placed between BMSC and all E-UTRAN Node Bs (Evolved Packet Core - eNBs). The principal function of this gateway is to send IP multicast to all eNBs that are using the eMBMS service. Therefore, eMBMS Gateway is responsible for MBMS session (Session Start/Stop) toward eNBs.

- Multimedia Broadcast Multicast Service Single Frequency Network (MBSFN): In this area a group of cells are coordinated to achieve a MBSFN transmission, which is a simultaneous transmission technique that sends identical waveforms at the same time from multiple cells. The group of cells transmission is seen as a single transmission by the user equipment.

- Multi-cell/multicast Coordinating Entity (MCE): This entity is admitted to control and allocation of the radio resources throughout the MBSFN. The MCE can be residing as a part of a network element such as part of eNB.
- Mobility Management Entity (MME): This entity is responsible for mobility and session management procedures in the Evolved Packet Core (EPC). This entity keeps location information at the tracking area level for each user and is involved in choosing appropriate gateway during the initial registration process. The MME communicates with the users' equipment as termination point via Non-Access Stratum (NAS) signaling and responsible for authentication communicates by interacting with the Home Subscriber Server (HSS).

The most interesting and important characteristic of LTE MBMS is a synchronous LTE network, which allows broadcast over a Single Frequency Network (MBSFN). It is shown in Figure 1.

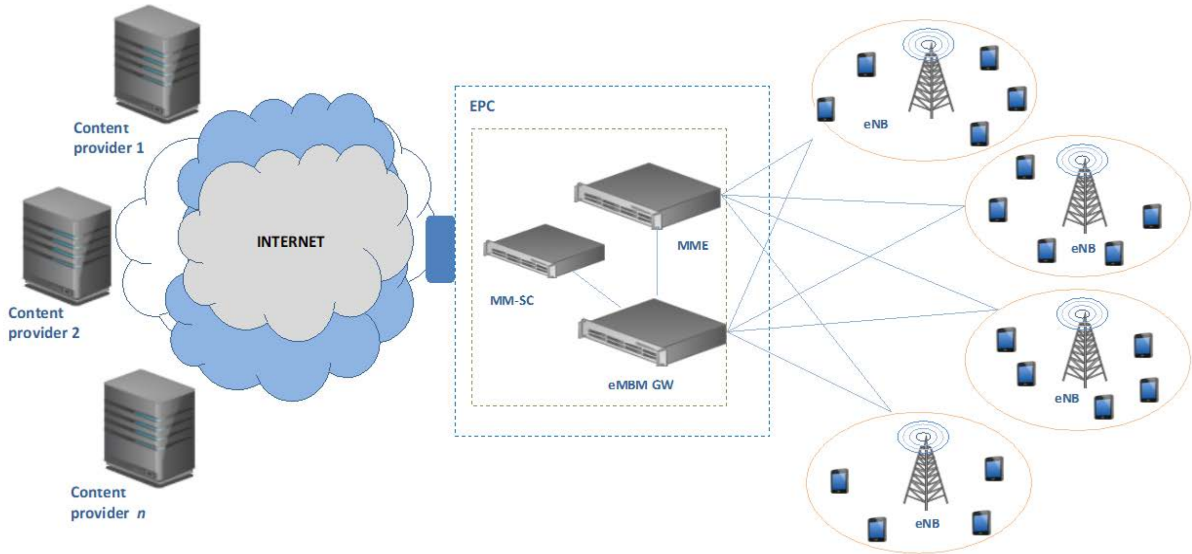

Figure 1. Architecture of eMBMS-LTE.

Therefore, according to eMBMS service, residing cache services toward the network edge, in both EPC caching and RAN caching, closer to users equipment makes it possible to simultaneously reduce the overload network traffic and reduce user-perceived latency. It is shown in Figure 2. This feature optimized the network operating costs because it connects eNBs to EPC [24].

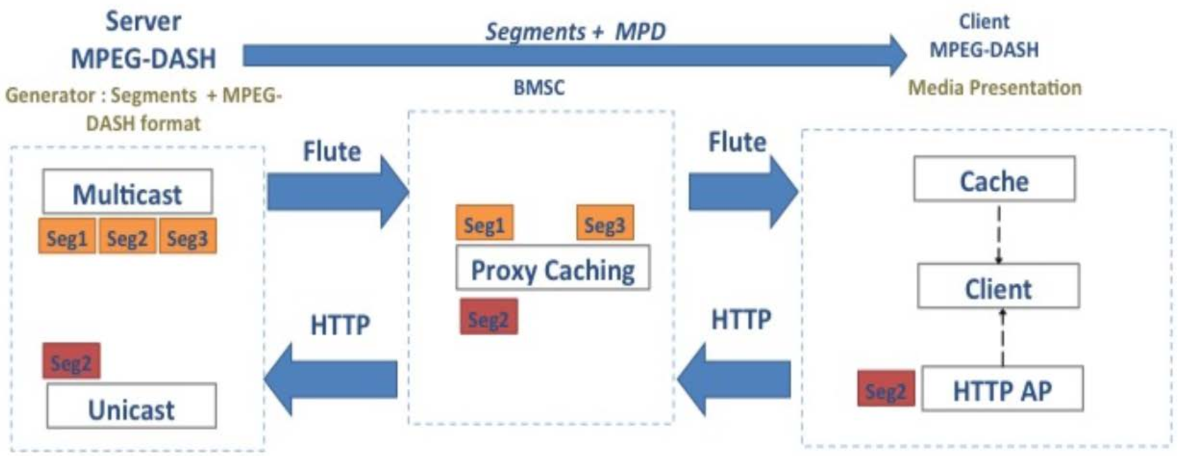

Figure 2. eMBMS-LTE and Proxy Caching.

4. Proposal Algorithm

In this section, we propose an algorithm to estimate the lost and recovered segments for adaptive video streaming application in eMBMS service. The algorithm is based on the information gathered from the network parameters and the characteristics of the video. MPEG-DASH client is a handset terminal that downloads the MPD file. The file contains the video representations quality and the location of the segments. When the stream is established, the file is sent to all of eMBMS users. Then, the client starts requesting the video segments, which are transmitted though the eMBMS channel. Therefore, when the user from one area is moved to different areas during the transmission, the user's player keeps requesting the segments that are defined in the MPD. Since these video segments are not reached to client's buffer through the eMBMS channel of the new area, they are retrieved via HTTP.

According to Figure 3, when the algorithm starts running the "proxy server" listening mode, it becomes active. It stays in this mode as long as there is no connection request from a video client web application. As soon as there is a request, the "proxy server" goes out and checks that the client-server connection conditions are optimal for the transmission of the video. If this is not the case, the connection is rejected. If the conditions are matched, the connection is established and the management system for the transmission of the video in the web application is started.

The management system consists of mainly of two phases: control of video segments and loss estimation of said segments. In the segment control phase, the "proxy server" sends each video segment to the client once the transmission has started. These segments are obtained by TCP and unicast from the "MPEG-DASH server". In the loss estimation phase, the management system is responsible for predicting the loss of video segments during transmission to request them in advance from the "MPEG-DASH server" and thus avoid delays and video stops during the transmission through the unicast channel. The control phase that is responsible for transmitting each video segment to the clients. In the case that one of the video segments is lost during the transmission, it checks if it is in cache in order to transmit it again. If it is not available in the cache, it requests it from the “MPEG-DASH server" with the lowest congestion, which sends it back to the client. In the segment loss estimation phase, which is based on the studies carried out, we can estimate the loss of video segments based on the bandwidth, the characteristics of the video and the accumulated time in the transmission.

The estimation result can determine one of the following possible cases:

- It is estimated that there will be some error in the transmission and a segment of video will be lost (Positive)
- It is estimated that there will be no error in the transmission and that therefore the video segment will not be lost (Negative)
- It is estimated that there will be some error in the transmission and however it does not occur and therefore there is no loss of segment (False Positive)
- It is estimated that there will be no error in the transmission and nevertheless if it happens, a loss of a video segment will occur (False Negative)

For each one of the above cases, the management system acts as follows:

- Positive: A request is sent to the MPEG-DASH server to estimate the segment that will be lost in future. In this case, the segment is searched in the cache to be re-sent to the client.
- Negative: In this case, the system does not perform any type of operation since it is estimated that the corresponding video segment will not be lost and this is indeed the case.
- False Positive: In this case the proxy server asks the DASH server for the segment that is estimated to be lost, but it is not lost. Then, said segment will be eliminated and the event will be saved in the register so that, in this way, it feedbacks the system and improves the estimation results.
- False Negative: In this case it is not estimated that the corresponding segment will be lost, but nevertheless if it is lost. Then, from the control phase, the request of said segment will be carried out to the less congested "MPEG-DASH server". Then, the event will be saved in the register so that, in this way, it feedbacks the system and improve the estimation results.

Finally, it may be that the number of lost video segments is very high. This case would be detected in what is the management system, in the phase of transmission control of segments. To solve it, the variation of parameters such as the bandwidth and the characteristics of the transmitted video would be carried out.

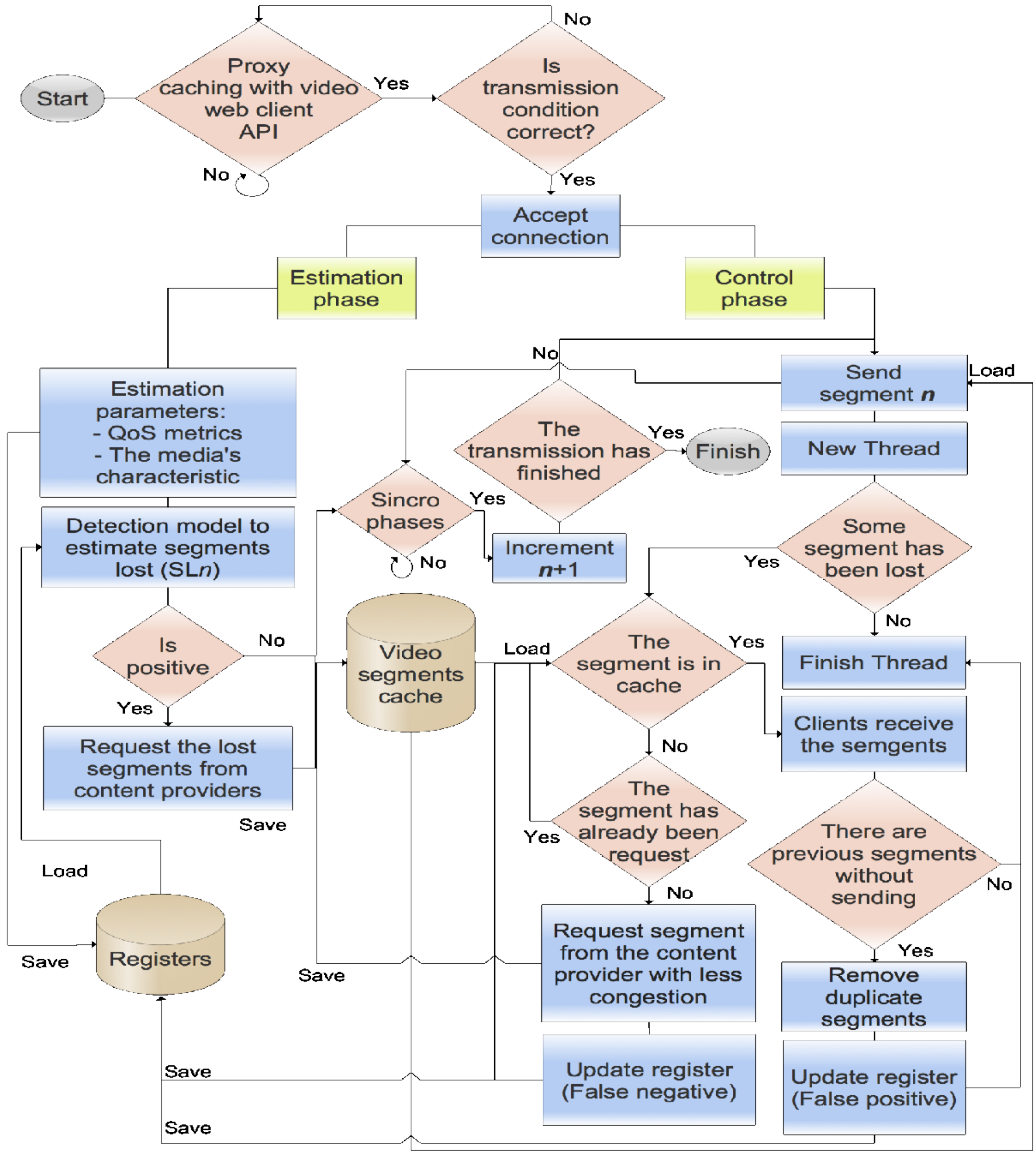

Figure 3. Proposed algorithm.

5. Experimental setup and performance evaluation

In order to evaluate the proposed algorithm, we provide an experimental study for evaluating QoE according to observation of the objective metrics, such as stalls, quality oscillation and buffer length and accumulative video time.

A) Calculation of QoE influence metrics

Buffer size: the client's application creates a buffer length for Y duration in seconds, Y seconds duration at beginning based on video segment length, and maximum availability of buffer size related to QoS factors which Buf_sl denoted as buffer size of segment length, where $Buf\_sl_i < Buf\_sl_{i+1}$, $i = 1$ to M, where M is the maximum number of segment length. The buffer at time T_0 starts to receive data and stores them in the buffer queue before being played back at time T_n.

Quality oscillation: The number of quality switch is described as oscillation of the video session. It depends on the number of bits flow through the channel per second and the instability of throughput. High value of frequent switching leads to decrease the QoE as described in the following formula.

$$\sigma^2 = \sum_{i=0}^{n} g(si) \qquad \begin{cases} 1 & if \quad i = 1 \\ 1 & if \quad f(s_{\,i-1}) \neq f(s_{\,i}) \\ 0 & else \end{cases} \tag{1}$$

s_i ... $segment_{\,i}$, $i \in [0 - N]$ ... $Segment\ Index$ f_{si} ... $bitrate\ of\ segment_{\,i}$

Accumulative video time: it is the exceeded video time over the default video length. It is the insufficient bandwidth caused due to the application's buffer is empty and the stalls are occurred during the playback the video. The accumulative video time can be included the startup delay of the video until the end of the video. It is expressed by

$$accumulative\ time = Initial\ delay + \gamma \tag{2}$$

Where $\gamma = \sum_{i=0}^{n} Zi$

B) Experimental system and material description

In order to measure the QoE in eMBMS, we provide a network system testbed that presents the scenario of eMBMS in LTE (shown in Figure 2). In real life scenarios, adaptive video streaming service is used for mobiles devices. Although, these devices access to the service provider throughout LTE network connection point, most users receive multimedia service over cellular network, so different scenarios appear such as mobile user at home, pedestrian, railroad, etc. We have considered these scenarios in order to apply the tests and obtain the dataset. The topology of the network scenario is depicted in Figure 4. We provided a real testbed setup, which covered fives modules.

The main server is MPEG-DASH web server, which is hosted on Linux Ubuntu. The server provided encoding video and adaptive video streaming. Network shaper is a hardware device which is equipped with Ubuntu operating system. The shaper provides shaping QoS parameters of the downstream and upstream channels by prioritizing network resources and guaranteeing certain bandwidth based on some predefined policy rules. It uses concepts of traffic classification, policy rules, queue disciplines and QoS. This is done in order to shape and control the network's uplink and downlink, delay, jitter and packet loss ratio. Proxy caching server is an intermediary device between the users and the content provider. When a user accesses a website, proxies interpret and respond to requests on behalf of the original server.

LTE network emulator emulates the LTE connection according the QoS parameters. We selected the BigBuckBunny video [25] in order to cover sufficiently the targets of real-life medias and applications in order to be able to produce noticeably the degradation in the process of assessment. Therefore, in order to provide an accurate assessment, 300 seconds from the raw video is encoded. This length is selected to generate higher number of segments and receive better assessment from the evaluation users. Long sequence time may fatigue the participants and short sequence time may present inaccurate results.

Generally, each video was encoded into 6 representations as shown in Table 1 with x264 encoder to cover different quality levels. The test sequences are segmented with GPAC's MP4Box with a segment length of 2 seconds.

Table 1. Characteristic of adaptive video streaming.

Quality level code	Resolution	Aspect	Bitrate (kbps)
1	384x288	4:3	500
2	512x384	4:3	700
3	1280x720	16:9	1000
4	1920x1080	16:9	1500
5	2048x1152	16:9	2500
6	3840x2160	16:9	5000

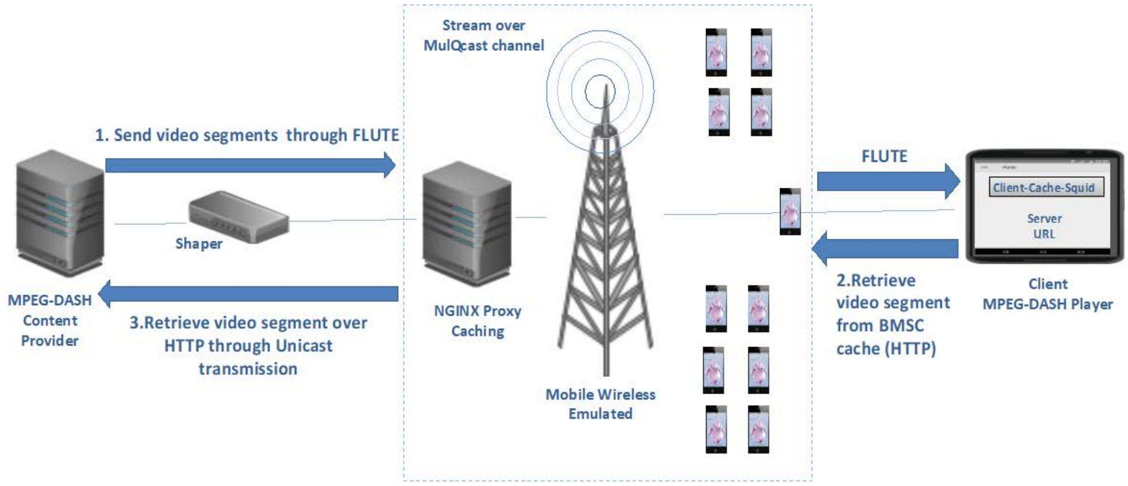

Figure 4. The testbed scenario.

C) Experimental analysis

DASH client request MPD file from the main server and it retrieves the MPD file on to the device. After that, the client's application reads the MPD. Then, the application starts requesting the segmentations of the video that are streamed through the eMBMS channel. However, when the user moves from one area to another, where the segments of different representation are sent, the application keeps requesting the video segments of adaptable quality that is defined in the MPD. Since these segments are lost in the multicast channel due to the high rate of packet loss, they are retrieved using unicast connection via HTTP protocol.

When high number of the segments is lost during the transmission and the request time of unicast connection has a high delay, the application's buffer takes time to fill the buffer queue as fast as possible. As a result, number of stalls to render the video is increased. According to the case study, we provided a wide number of experiments to evaluate QoE

In the first experiment, we observe the objective metrics such as video stall (buffer underran) metric according to number, place, and duration, of the stalls. The second metric is the accumulative video time according to stalls, and in the third metric, the buffer length shows maximum and minimum length of buffer during the transmission. These metrics are highly relative to the evaluation of the QoE.

In this experiment, the network throughput of the multicast and unicast connections is configured to 350Kbps. In this case, many segments are lost and the user faced to highly degraded QoE with many stalls in different places of the video playback, as shown in Figure 5. High number of video stalling appeared in the observation of the experiment, the application player frozen the video 10 times. The maximum duration of stall reached 5 seconds and the sharp stalls occurred at the beginning of the play back until the end.

Network Protocols and Algorithms
ISSN 1943-3581
2017, Vol. 9, No. 3-4

Therefore, the video accumulative time in this experiment is highly increased to maximum range. It can be estimated to 80 seconds, since the video length equals to 300 seconds.

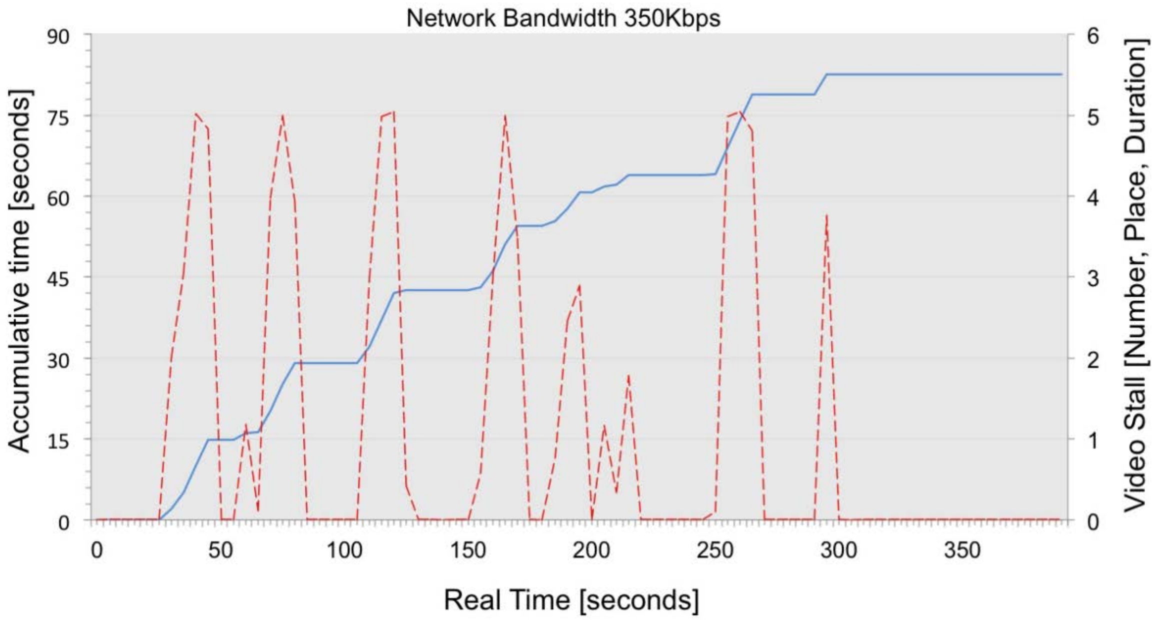

Figure 5. QoE evaluation when BW is 350Kbps

In the second experiment, the network throughput is configured to 500Kbp for both channels. At the beginning, the client application received a sequence of segments to playback and render. At 30th seconds its buffer suffers from filling the queue and the buffer stayed underrun for 5 seconds until next segments are received. Therefore, other video stalling occurred, which were respectively placed at 60th seconds, 100th seconds, and 220th seconds. The duration of these stalls respectively were 4 seconds, 1 second and 1 second. According to Figure 6, the video application's buffer in the 225th second receives the the video without interrupting the user's playback. Therefore, the accumulated video's time for the experiment is extended to 18 additional seconds. In this experiment, the QoE is enhanced by 25% compared to previous experiments.

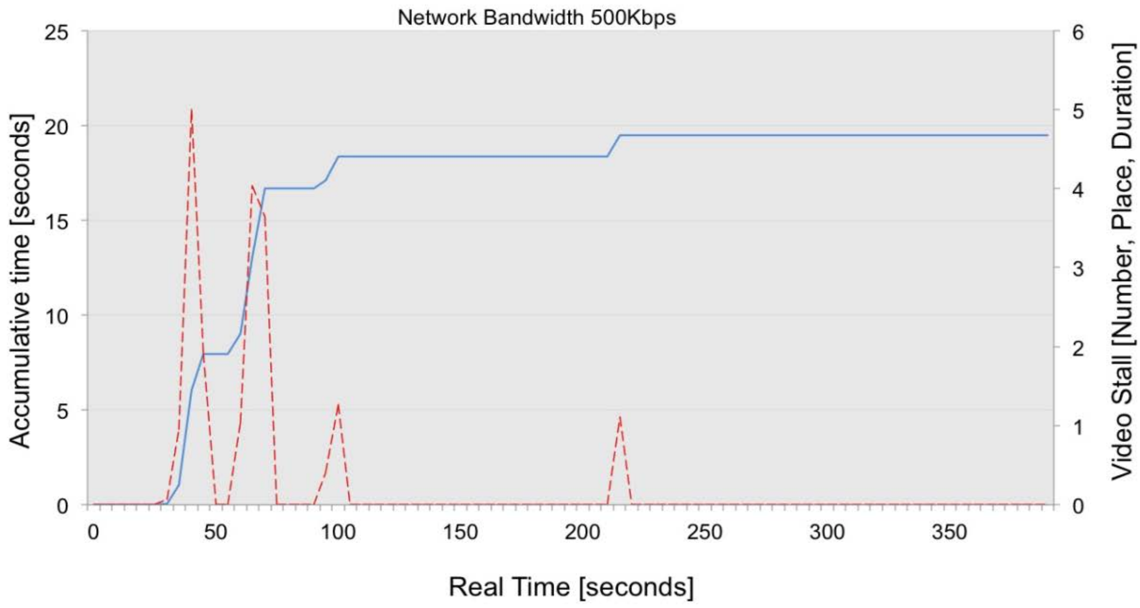

Figure 6. QoE evaluation when BW is 500Kbps

In the third experiment, the network throughput is configured to 550Kbps. It is shown in Figure 7. The number, duration and location of stalls are revealed at the beginning of the video playback. Therefore, in this experiment durations of the stalls are reduced to short range. Moreover, the sharpness stall is 0.8 seconds and the additional video time is 2.4 seconds. In this experiment, the QoE is increased by 82% compared to the second experiment and 93% compared to the first experiment.

In the fourth experiment, we test the size of the application's buffer according to the maximum and minimum size and the high and low fluctuation in the receiving data. In this experiment, the throughputs of multicast and unicast channels are configured to 350Kbps, 500Kbps and 550Kbps respectively.

As shown in Figures 8, 9 and 10, when the throughput has the minimum rate, the buffer size is degraded to a lower level. Moreover, the buffer oscillation in the period of the video playback is increased.

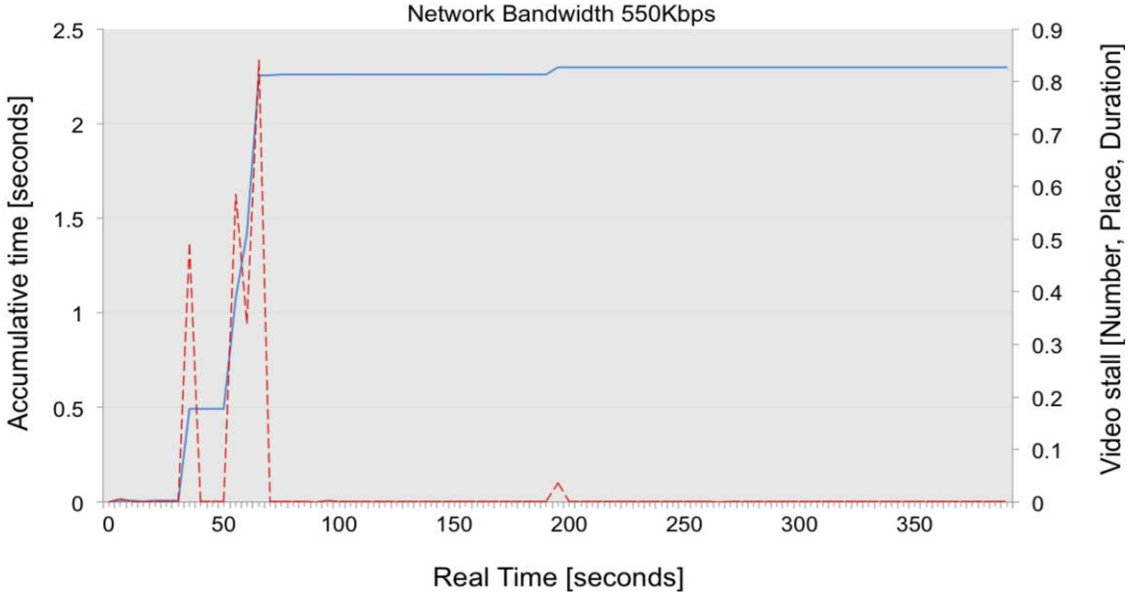

Figure 7. QoE evaluation when BW is 550Kbp

When the buffer is degraded to zero, as shown in the figures, the users are unable to continue watching the video. However, the segments contained less information of an I-frame. The application's buffer, becomes stable and away from stalling as shown at the beginning and end of the figures. In Figure 9 and 10, between 150 to 200 seconds, the application's buffer avoided to interrupt rendering the video because there were enough segments downloaded to the buffer.

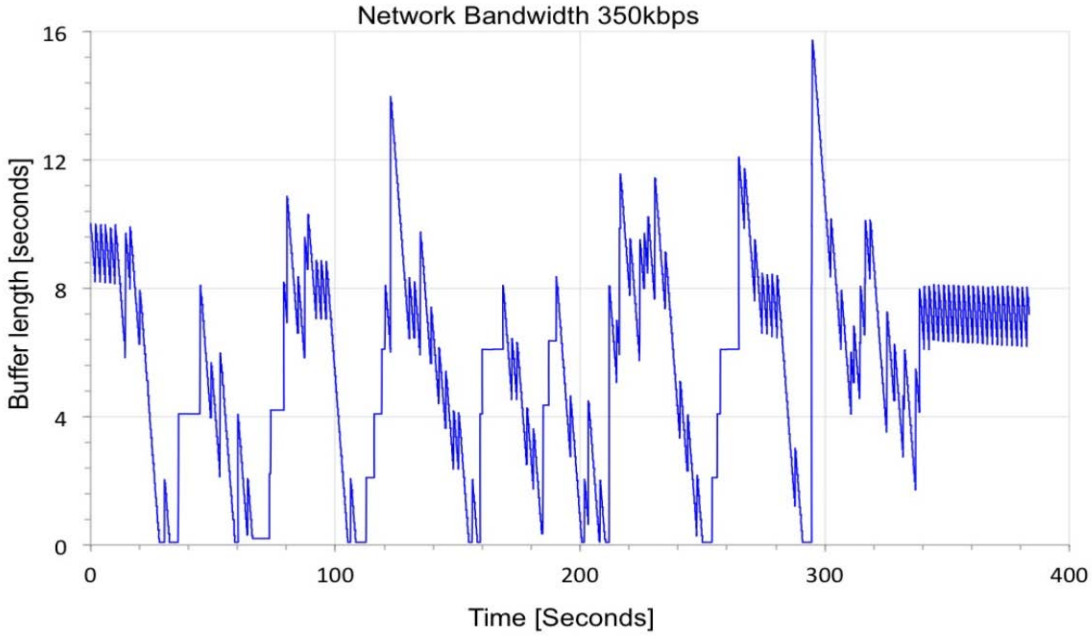

Figure 8. Buffer size when BW is 350Kbps.

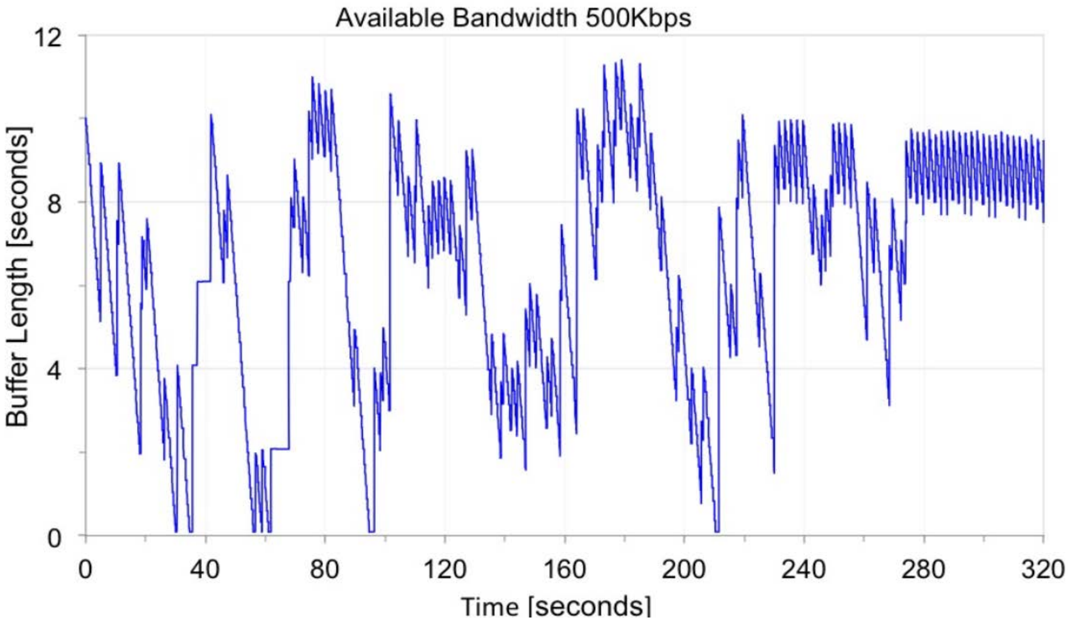

Figure 9. Buffer size when BW is 500Kbps

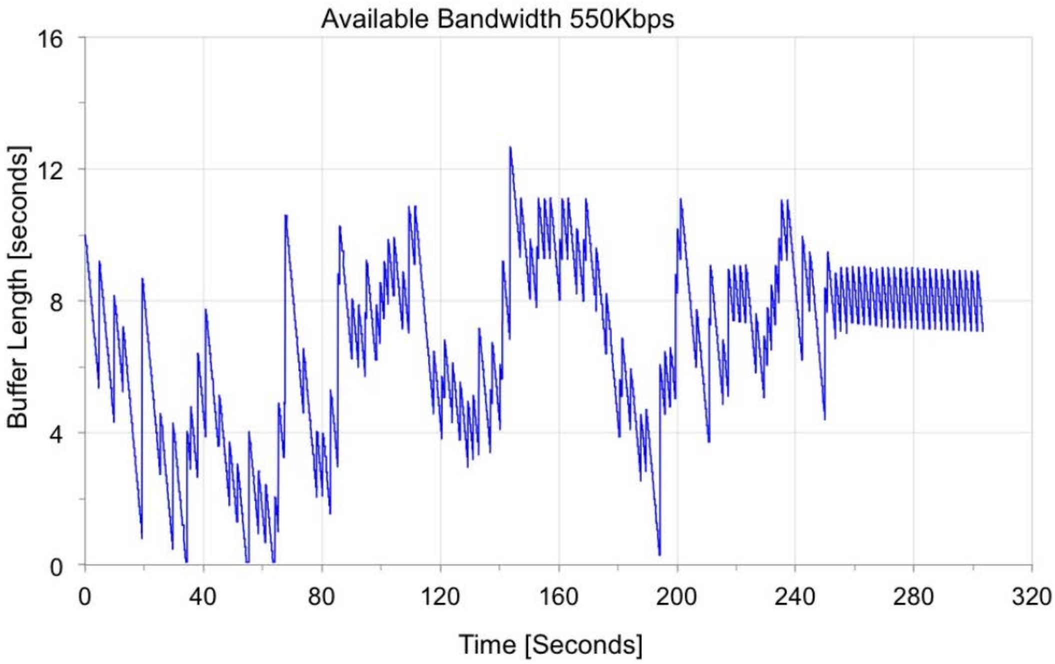

Figure 10. Buffer size when BW is 550Kbps

In the fifth experiment, we observe the number of segments that are lost and received in multicast transmission. In this experiment, the multicast channel throughput is configured to 520 Kbps and the unicast throughput to 5Mbps. In this scenario, number of the base stations is 10 and the number of the users is variable. In our scenario, the number of the mobile users is divided into different groups. The maximum number of users is receiving multicast service from the content provider is 395 users. However the minimum amount of the users is 75 users.

Therefore, Forward Error Correction (FEC) code rate, and modulation and coding scheme (MCS) of the multicasting groups, which correspond to a FLUTE session, are set to 0.85 and 590 respectively. As shown in Table 1, the server transmits 300 segments of lower quality (500Kbps) to mobile users. Figure 11 represents the relation between the video segment lost/received and the variety range of users. The red bar indicates the lost segments in the transmission and the blue bar indicates the segments that arrived to the clients. In this experiment, when there are 140 and 325 users, many segments are lost. However, for 375, 388, 395, and 72 users, we obtained the maximum range of delivered segments. Although, the number of lost of segments is highly related to channel capacity and rate of loss in the channel, this is caused to huge number of segments would be lost.

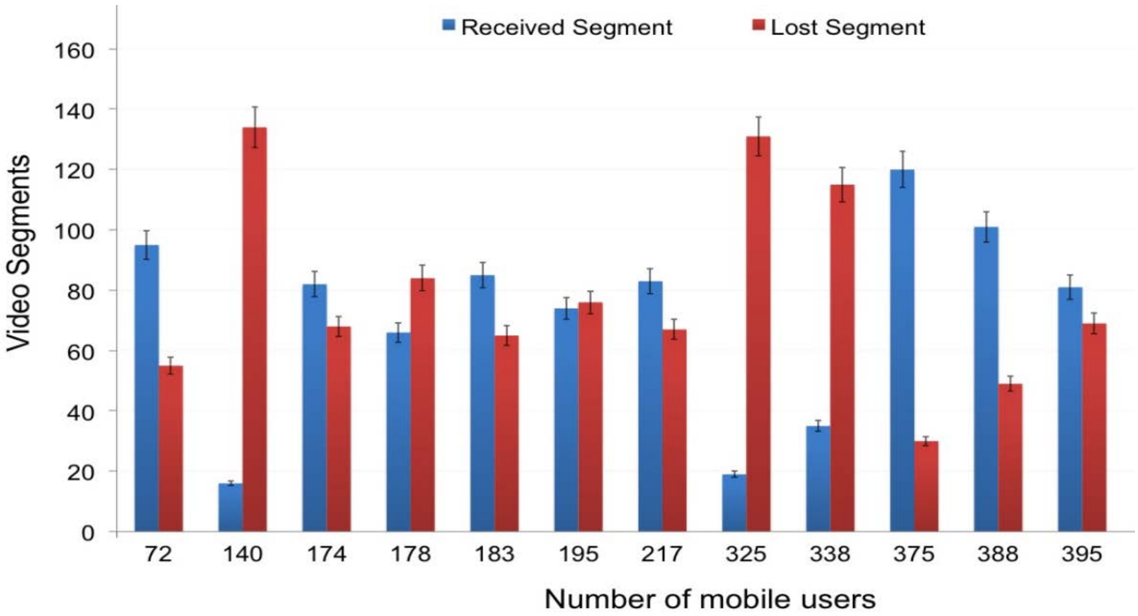

Figure 11. Segment lost/received in eMBMS

In last experiment, we carried out the performance evaluation of the proposed algorithm. In order to provide an accurate comparison between the traditional approach and the proposal, we repeated the experiments 10 times and we took mean value of the measures. This way helped us to understand deeply the relative parameters, which highly affected the lost and recover segments. Thus, when segments are lost in multicast channel these segments are recovered using HTTP protocol through the unicast channel as depicted in Figure 2. Therefore, many parameters are impacted by the arrival time of the delivering segments in unicast channel as explained in Section 4. Delay time of recovering segments through the unicast channel for both approaches is shown in Table 2. The proposed approach reduces the latency of arrival time of the segments. The considered approach provided better QoE in multicast and unicast when caches have huge rolls to bring the content to be closed to end-users.

Table 2. Comparison approaches.

Number of users	Lost segments from 300 segments	Time to recover Lost segments based on traditional algorithm (Sec.)	Time to recover Lost segments based on proposal algorithm (Sec.)
72	65	9	3
140	136	12	4
174	77	11	3
178	90	16	4
183	83	16	4
195	86	17	4
217	83	16	4
259	51	7	2
266	64	10	3
325	133	19	6
338	123	19	6
375	47	5	1

6. Conclusion and Future work

In this paper, we proposed an algorithm, which improved the QoE for mobile users where eMBMS is followed by multicast/unicast adaptive video streaming. The eMBMS takes advantage of single frequency network features and allows a flexible content delivery via unicast, multicast and broadband services. When high rate of the video segments of adaptive streaming are lost through the handover process, Internet congestion or by limited resources, the QoE of adaptive video streaming users become degraded and an unsatisfactory service reaches to mobile users. The proposed algorithm takes into account the QoS and the characteristic of media in order to estimate the number of segments that are going to be lost in eMBMS. According to the observed results, the algorithm highly improves QoE by reducing the latency time of delivering lost segments through the unicast channel to end-users.

In future work, we plan to provide our algorithm in a sophisticated system with complex scenario and take into account different models of deep learning.

Acknowledgement

This work has been partially supported by the Postdoctoral Scholarship "Contratos Postdoctorales UPV 2014 (PAID-10-14)" of the "Universitat Politècnica de València", by the "Programa para la Formación de Personal Investigador—(FPI-2015-S2-884)" of the "Universitat Politècnica de València", by the "Ministerio de Economía y Competitividad", through the "Convocatoria 2014. Proyectos I+D - Programa Estatal de Investigación Científica y Técnica de Excelencia" in the "Subprograma Estatal de Generación de Conocimiento", project TIN2014-57991-C3-1-P and through the "Convocatoria 2017 - Proyectos I+D+I - Programa Estatal de Investigación, Desarrollo e Innovación, convocatoria excelencia" (Project TIN2017-84802-C2-1-P).

References

[1] Ericsson Mobility Report. November 2017. Available at: https://www.ericsson.com/assets/local/mobility-report/documents/2017/ericsson-mobility-report-november-2017.pdf (last accessed November 20, 2017)

[2] GSA: Evolution to LTE Report – January 2017. Available at: https://gsacom.com/paper/gsa-evolution-lte-report-january-2017/. (last accessed November 20, 2017)

[3] Enhanced Television Services over 3GPP eMBMS. Available at: http://www.3gpp.org/news-events/3gpp-news/1905-embms_r14. (last accessed November 20, 2017)

[4] Definition of Quality of Experience (QoE). Available at: https://www.itu.int/md/T05-FG.IPTV-IL-0050 (last accessed November 20, 2017)

[5] ISO/IEC 23009-1:2014. Information technology -- Dynamic adaptive streaming over HTTP (DASH) -- Part 1: Media presentation description and segment formats. Available at: https://www.iso.org/standard/65274.html. (last accessed November 20, 2017)

[6] FLUTE - File Delivery over Unidirectional Transport. Available at: https://tools.ietf.org/html/rfc6726 (last accessed November 20, 2017)

[7] Jiasi Chen, Mung Chiang, Jeffrey Erman, Guangzhi Li, K. K. Ramakrishnan, and Rakesh K. Sinha. "Fair and optimal resource allocation for LTE multicast (eMBMS): Group partitioning and dynamics." In Computer Communications (INFOCOM), 2015 IEEE conference on, Kowloon, Hong Kong, 26 April - 1 May 2015, pp. 1266-1274, https://doi.org/10.1109/INFOCOM.2015.7218502

[8] Tara Ali Yahiya, "Adaptive video streaming rate control for HetNet based on interference scenarios" Wireless and Mobile Networking Conference (WMNC), 2017 10th IFIP, Valencia, Spain, 25-27 Sept. 2017, https://doi.org/10.1109/WMNC.2017.8248843

[9] Sarabjot Singh, , Ozgur Oyman, Apostolos Papathanassiou, Debdeep Chatterjee, and Jeffrey G. Andrews. "Video capacity and QoE enhancements over LTE." In Communications (ICC), 2012 IEEE International Conference on, pp. 7071-7076. IEEE, 2012, https://doi.org/10.1109/ICC.2012.6364806

[10] Tara Ali-Yahiya." Understanding LTE and its Performance. " Springer Science & Business Media, 2011, https://doi.org/ 10.1007/978-1-4419-6457-1

[11] Miran Taha. "A Novel CDN Testbed for Fast Deploying HTTP Adaptive Video Streaming." Proceedings of the 9th EAI International Conference on Mobile Multimedia Communications, ACM, Xi'an, Chine, pp. 65-71, June 2016.

[12] Michael Seufert, Sebastian Egger, Martin Slanina, Thomas Zinner, Tobias Hossfeld, and Phuoc Tran-Gia. "A survey on quality of experience of HTTP adaptive streaming." IEEE Communications Surveys & Tutorials, vol. 17, no. 1, 2015, pp. 469-492. https://doi.org/10.1109/COMST.2014.2360940

[13] Miran Taha, Jaime Lloret, Alejandro Canovas, Laura Garcia. "Survey of Transportation of Adaptive Multimedia Streaming service in Internet." Network Protocols and Algorithms, Vol. 9, Issue 1&2, pp. 85–125, 2017, https://doi.org/10.5296/npa.v9i1-2.12412

[14] Lecompte D., Gabin F., "Evolved multimedia broadcast/multicast service (eMBMS) in LTE-advanced: overview and Rel-11 enhancements". IEEE Communications Magazine, Vol. 50, Issue 11, Pp 68-74, November 2012, https://doi.org/10.1109/MCOM.2012.6353684

[15] Lohmar T., Slssingar M., Kenehan V., Puustinen S., "Delivering content with LTE Broadcast ". Ericsson Review, Vol. 1 , no. 11, Feb 2013, pp. 2-8.

[16] Nguyen N.-D., Knopp R., Nikaein N., Bonnet, C., "Implementation and Validation of Multimedia Broadcast Multicast Service for LTE/LTE-Advanced in OpenAirInterface Platform". IEEE 38th Conference on Local Computer Networks Workshops (LCN Workshops), October 21-24, 2013, pp. 70–76 , https://doi.org/10.1109/LCNW.2013.6758500

[17] Stephanakis I. M., Chochliouros I. P., Lymperopoulos G. L., Berberidis K., "Optimal Video Delivery in Mobile Networks Using a Cache-Accelerated Multi Area eMBMS Architecture", AIAI 2014 Workshops, IFIP AICT 437, September 19-21, 2014, Rhodes, Greece,pp. 13–23, https://doi.org/10.1007/978-3-662-44722-2_2

[18] Pande A., Ahuja V., Sivaraj R., Baik E., Mohapatra P., "Video Delivery Challenges and Opportunities in 4G Networks". IEEE MultiMedia, Vol. 20, Issue 3, July-Sept. 2013, Pp. 88-94, https://doi.org/10.1109/MMUL.2013.44

[19] Taha M., García L., Jimenez J. M., Lloret J., "SDN-based throughput allocation in wireless networks for heterogeneous adaptive video streaming applications", Wireless Communications and Mobile Computing Conference (IWCMC), 2017 13th, 26-30 June, 2017, Valencia, Spain, https://doi.org/10.1109/IWCMC.2017.7986416

[20] García L., Lloret J., Turro C., Taha M., "QoE assesment of MPEG-DASH in polimedia elearning System", 2016 Intl. Conference on Advances in Computing, Communications and Informatics (ICACCI), Sept. 21-24, 2016, Jaipur, India,, https://doi.org/10.1109/ICACCI.2016.7732194

[21] Stockhammer T., Luby M. G.,"Dash in mobile networks and services". Visual Communications and Image Processing (VCIP), 2012 IEEE, San Diego, CA, USA, 27-30 November, 2012, https://doi.org/10.1109/VCIP.2012.6410826

[22] Guo J., Gong X., Liang J., Zhang S., Zhao M., Wang W., Li Z., "A Hybrid Transmission Approach for DASH over MBMS in LTE Network". Global Communications Conference (GLOBECOM), 2015 IEEE, 6-10 December 2015, https://doi.org/10.1109/GLOCOM.2015.7417297

[23] Belda R., de Fez I., Fraile F., Arce P., Guerri J. C., "Hybrid FLUTE/DASH video delivery over mobile wireless networks", Transactions on Emerging Telecommunications Technologies, Vol. 25, Issue 11, November 2014, pp. 1070–1082, https://doi.org/10.1002/ett.2804

[24] Xiaofei Wang, Min Chen, Tarik Taleb, Adlen Ksentini and Victor C. M. Leung," Cache in the air: exploiting content caching and delivery techniques for 5G systems", IEEE Communications Magazine, Vol. 52, No 2, 19 February 2014, pp. 131 – 139, https://doi.org/10.1109/MCOM.2014.6736753

[25] https://media.xiph.org, (last accessed November 20, 2017)

Copyright Disclaimer