Adaptive Layered Segment Algorithm for Media Delivery over 4G LTE Wireless Cellular Networks

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Abstract—Advances in wireless communications and mobile networking have dramatically increased the popularity of media services for mobile users, with video delivery at their fingertips. Delivering high perceptual quality video over 4G LTE wireless cellular networks is challenging due to the changing channel quality and the variations in the importance of one source packet to the next for the end-user’s quality of experience. Therefore, new video transmission paradigms are required that make media-aware scheduling and resource allocation decisions at the level of the codec and the radio to optimize perceived video quality and minimize server/ network load and decoder’s buffer requirements. This paper proposes a layered Segment Algorithm for media delivery over 4G LTE Wireless Networks, which aims to address the problem of multi-user radio resource management and users decoding capabilities with various buffer requirements. Simulations demonstrate that the proposed Layered Segment Algorithm adapts effectively to bandwidth variations, packet loss and buffer requirements. Moreover it achieves superior performance over conventional schemes that require proportional media content scheduling on the network paths according to their available bandwidth.

Index Terms—4G LTE Networks, Media-aware networking, Signal processing for transmission, Transmission and networking

I. INTRODUCTION

Wireless communications and mobile networking have dramatically increased the popularity of media-centric services for mobile users. 3GPP Long Term Evolution (LTE) is the name given to the new standard developed by 3GPP to cope with future market requirements. LTE is the next step in the evolution of 3G systems and the goal is to provide levels of quality similar to those of current wired networks. Nowadays clients are demanding continuous delivery of increasingly higher quality videos over the Internet, in both wired and wireless networks. Due to its real-time nature, video streaming typically has bandwidth, delay and loss requirements. However, for wireless video delivery in LTE, especially delay-bounded real-time video streaming, higher data rate could lead to higher packet loss rate, thus degrading the user-perceived video quality [1] - [5]. Moreover the delivery of video streaming has the properties of real-time, continuity, and data dependency. Continuity demands that all video frames must be rendered in a predefined order. Generally, compressed video has two types of data dependency: inter-frame dependency demands that before decoding a frame, all frames it refers to must have already been decoded; intra-frame dependency demands that it’s preferable to decode a frame after all packets belonging to this frame have already been received, otherwise decoding errors and mosaic-like rendering effects may be resulted. [6] - [8]. Recently the Dynamic Adaptive Streaming over HTTP (DASH) specifications is available from 3GPP and MPEG. 3GP-DASH specification provides a normative definition of a Media Presentation, a normative definition of the formats of a Segment with a Segment defined as an integral data unit of a media presentation., a normative definition of the delivery protocol used for the delivery of Segments, namely HTTP/1.1, and an informative description on how a DASH client may use the provided information to establish a streaming service for the user. [9] - [11].

In LTE, the scheduling entity is placed at the base station (BS) which is responsible for the multi-user radio resource management (RRM) of time, frequency and antenna resources. The generic characteristics of wireless networks are time-varying and their performance is generally inferior to those of wired networks. Therefore, it is still a challenging problem to efficiently provide a video delivery service of high quality over 4G LTE Wireless Networks [12], [13]. To address these challenges, extensive research has been conducted [14]-[19]. In [14], the authors propose a conventional cross-layer packetization and retransmission technique based on a joint application layer adaptive prioritized scheduling and a MAC-layer retransmission strategy. The cross-layer problem is posed with a distortion minimization objective and existing joint source-channel coding theory is modified to account for delay constrained transmission.

There are several updated research results [15-17] on the cross-layer optimization to improve the video transmission quality. The authors in [15] considered intra refreshing rates and lower layer access strategy as the key factors that can
optimize multimedia transmission quality. In [16], a mixed
integer nonlinear programming (MINLP) method is used for
solving the quality optimizations problem. In [17], a theoretic
solution for wireless resource management is proposed for
reducing multimedia transmission delay. In [18], a layered
video transmission scheme over MIMO is proposed and
termed adaptive channel selection. It periodically switches
each bit stream among multiple antennas. The bit stream
switching matches the ordering of SNR strength for the sub-
channels and enables prioritized delivery and unequal error
protection. In [19], the authors consider the problem of QoS-
aware LTE OFDMA scheduling and propose a cross-layer
system for real-time video delivery to maximize the video
quality subject to the application-layer delay constraint. A
weighted round robin algorithm provides application level
QoS guarantees in terms of fairness and delay constraints.
It should be emphasized that none of the methods mentioned
above fully address the problem of random packet losses,
time-varying network conditions, users decoding capabilities
and buffer requirements for efficient media delivery over 4G
LTE Wireless Networks.

In this paper, we propose a very efficient Layered Segment
Algorithm for media delivery over 4G LTE Wireless
Networks, which aims to address the problem of multi-user
radio resource management, network load, users decoding
capabilities and buffer requirements. According to the Layered
Segment Algorithm, the media content is divided into variable
media segments each of which is encoded at various bitrates
and quality and can be decoded independently. Layered
Segments contain media data and/or metadata to access,
decode and present the included media content. Moreover in
the proposed approach the LTE scheduler can make use of
information defined in Adaptive Layered Segment algorithm
for more efficient video delivery. Simulations with internet
network traces demonstrate that the proposed Layered Segment Algorithm adapts effectively to bandwidth variations,
packet loss and buffer requirements. Moreover they show that
the proposed Layered Segment Algorithm provides superior
performance over conventional schemes that achieve
proportional media content scheduling on the network paths
according to their available bandwidth. It should be noted
that the performance of the proposed technique can be also
affected by the characteristics of media content. Thus, we
chose different video sequences, sequences from the video
database [20], [21], based on common test conditions listed in
[22].

The paper is organized as follows. Section II presents the
proposed Layered Segment Algorithm and specifically focuses
on the 4G Wireless Network resources and client-end
requirements. Experimental results that illustrate and validate
the accuracy of the proposed Layered Segment Algorithm are
presented in Section III. Finally, conclusions and future
research are discussed in Section IV.

II. LAYERED SEGMENT ALGORITHM

The video quality consumed by the end-user is impacted by
two main sources of distortion: source distortion due to lossy
compression and channel distortion due to channel-induced
impairments such as packet losses, errors, and delays. There
is always a tradeoff between these two types of distortions in
the framework of a rate-adaptive codec and a link-adaptive
transmission medium. For a given channel capacity, improving reliability through channel coding comes at the
expense of reducing source rate. On the other hand, for a static
channel code design, increasing the modulation order to
support higher source rates reduces error flexibility. Consequently we propose a very efficient Layered Segment
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The segment formats are defined for two encapsulation
methods MPEG-2 TS and ISO base Media File Format
(ISOBMFF). It should be noted that the MPEG DASH
standard could be defined the formats of these segments and
the format of data and/or metadata as the media description
representation (MPD), which is an XML document describing
the characteristics of the media available at the server [10],
[11]. Moreover the ISOBMFF is used as the basis for many
codec encapsulation formats. In addition to continuous media,
as well as metadata can be stored in a file conforming to
ISOBMFF. Files structured according to the ISOBMFF can be
used for local media file playback, progressive downloading
of a remote file, segments for Dynamic Adaptive Streaming
over HTTP (DASH), containers for content to be streamed and
its packetization instructions, and recording of received real-
time media streams. [11]. The latest ISOBMFF specification
specifies six types of Stream Access Points (SAPs) for use
with DASH. The first two SAP types (types 1 and 2),
correspond to IDR pictures in H.264/AVC The third SAP type
(type 3) corresponds to open-GOP random access points.

The objective of this work is to enhance video perceptual
quality and make more efficient use of network capacity by
developing adaptive layered segment algorithm for media
delivery over 4G LTE Wireless. At the application layer,
hybrid motion-compensated video coding is usually adopted
for transmission over lossy channels.

Adaptive layered segment at time instance i for k different
users can be defined as follows

\[
\text{Segment} = \arg\max_{\kappa, i} \sum_{\kappa=1}^{K} r_{k}(i) R_{\kappa}(i) \Delta_{\kappa}(i) B_{k}(i)
\] (1)

where
\[ r_k(i) = r_1(1), r_2(2), r_3(3), \ldots \text{denotes the accessible data rate inside, the available rate region } C_k(i) \text{ defined by the LTE scheduler.} \]

Hence, \( r_k(i) \leq C_k(i) \). LTE's scheduler can make use of information defined in adaptive Layered segment to adaptive optimization of radio resource allocation among the different users' terminals.

\[ R_k(i) = R_1(1), R_2(2), R_3(3), \ldots \] denotes the initial data rates of each user allocated by LTE scheduler.

The main task of the scheduler is to allocate radio resources to all user terminals (UTs) in a multi-user scenario. Radio resources are elements from the time, frequency, and spatial domain of the LTE system.

\[ \Delta_k(i) = \Delta_1(1), \Delta_2(2), \Delta_3(3), \ldots \] denotes the priority indicator which signals delay- or throughput-or packet loss awareness of the user.

\[ B_k(i) = B_1(1), B_2(2), B_3(3), \ldots \] denotes the buffer fullness at each time instant and based on predefined thresholds which depends on of each user terminal-capabilities. Therefore the limitation in buffer storage capacity is considered, which means that clients will continue access layered media-segment as long as the channel characteristics and buffer fullness allows for it.

### III. EXPERIMENTAL RESULTS

Computer simulations using video sequences were performed to evaluate the performance of the proposed adaptive layered segment algorithm. The JSVM reference software (H.264/AVC JM 15.1) [24] is used to encode and decode the layered segment media. The “riverbed” and “pedestrian” sequences from the video database [20], [21] are adopted for performance analysis, each sequence is transmitted by all users in a cell simultaneously. For these users, the first 1000 frames of the video sequences are coded at a frame rate of 25 frames/s, and each Intra (-I) frame is followed by several Predicted (-P) frames. Specifically, simulations have been performed for video sequences with different motion characteristics according to the VCEG common conditions [23] for -I and -P coding. In this simulation, the structure of Group of Pictures (GOP) is “IPPPPPPPP...” i.e., all frames except the first frame (I-) are encoded as -P frame.

Assume that the achievable layered segment media of data rates is 0.5 Mbps to 3 Mbps. The layered segment media model used in the simulation is based on an encoded the “riverbed” and “pedestrian” video streams with three operating point for each video sequence, as depicted in Table 1.

One of the main reasons that influence the Quality of Experience (QoE) of the users of a video service is the appearance of break or interruption of a video delivery due to minimum available resources at the network load and or the user terminals buffer limitations, respectively.

<table>
<thead>
<tr>
<th>Video trace</th>
<th>Quantity (avg. dB-Mbps)</th>
<th>Layered media segment (Byte)</th>
</tr>
</thead>
<tbody>
<tr>
<td>River-bed</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Q1 (LOW)</td>
<td>34.2 dB</td>
<td>Segment (A) : 20.8 KByte</td>
</tr>
<tr>
<td></td>
<td>0.5 Mbps</td>
<td>Segment (B) : 20.8 KByte</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Segment (C) : 20.8 KByte</td>
</tr>
<tr>
<td>Q2 (MEDIUM)</td>
<td>36.2 dB</td>
<td>Segment (A) : 62.5 KByte</td>
</tr>
<tr>
<td></td>
<td>1.5Mbps</td>
<td>Segment (B) : 62.5 KByte</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Segment (C) : 62.5 KByte</td>
</tr>
<tr>
<td>Q3 (HIGH)</td>
<td>38.2 dB</td>
<td>Segment (A) : 104.16 KByte</td>
</tr>
<tr>
<td></td>
<td>2.5Mbps</td>
<td>Segment (B) : 104.16 KByte</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Segment (C) : 104.16 KByte</td>
</tr>
<tr>
<td>Pedestrian</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Q1 (LOW)</td>
<td>34.9 dB</td>
<td>Segment (A) : 41.6 KByte</td>
</tr>
<tr>
<td></td>
<td>1 Mbps</td>
<td>Segment (B) : 41.6 KByte</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Segment (C) : 41.6 KByte</td>
</tr>
<tr>
<td>Q2 (MEDIUM)</td>
<td>37.1 dB</td>
<td>Segment (A) : 83.3 KByte</td>
</tr>
<tr>
<td></td>
<td>2 Mbps</td>
<td>Segment (B) : 83.3 KByte</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Segment (C) : 83.3 KByte</td>
</tr>
<tr>
<td>Q3 (HIGH)</td>
<td>39.4 dB</td>
<td>Segment (A) : 125 KByte</td>
</tr>
<tr>
<td></td>
<td>3Mbps</td>
<td>Segment (B) : 125 KByte</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Segment (C) : 125 KByte</td>
</tr>
</tbody>
</table>

Therefore, in order to evaluate the quality of the proposed method, Figure 1 depicts the probability (%) of the smooth video delivery occur during an adaptive layered segment session and the conventional scheme, where the initial pre-buffer at the user terminals (UTs) is set to 10 sec and the number of users in the simulation is increased from 10 to 120 user-terminals (UTs) in a cell for the riverbed (Fig.1 (a)) and pedestrian (Fig.1 (b)) video sequences, respectively.
It is clearly observed that for all the video sequences (Fig.1 (a) and (b) the proposed adaptive layered segment algorithm outperforms the conventional scheme. Specifically the probability of the smooth video delivery remains constant for the proposed approach as the number of user-terminals is increased in a cell. This is due to the layered segment algorithm which takes into consideration the accessible data rate inside the available rate region defined by the LTE scheduler the initial data rates of each user allocated by LTE scheduler with the priority indicator which signals delay- or throughput-or packet loss awareness of different users and the buffer fullness at each time instant based on predefined thresholds which depends on of each user terminal-capabilities in order to efficient delivery video over less than good variable channel conditions that have different QoE requirements at end-users terminals. It is worth pointing out that adaptive algorithm with the optimized results could also be used to improve the performance of 4G LTE Wireless Networks resource allocation. The experimental simulation results indicate a significant performance improvement of the proposed very efficient algorithm compared to conventional schemes that require proportional media content scheduling on the network paths according to their available bandwidth.

Future work will investigate the performance of the proposed very efficient adaptive layered segment algorithm with High Efficiency Video Coding (HEVC) suited to large-scale delay-sensitive real-time video delivery over 4G LTE Wireless Networks that have different QoS requirements. Future work will also estimate Quality of Experience (QoE) parameters (subjective assessment) from Quality of Service (QoS) parameters (objective assessment) with a high degree of accuracy for adaptive layered segment-aware HEVC [25], [26] over LTE [27] and LTE-A [28] radio access networks that can provide hundreds of Mbps up to 1 Gbps, as well as short latencies and thus enable innovative video services with a novel users’ Quality-of-Experience (QoE) perspective.

### IV. CONCLUSION

In this paper, we present a Adaptive layered Segment Algorithm for media delivery over 4G LTE Wireless Networks, which merges the accessible data rate inside the available rate region defined by the LTE scheduler the initial data rates of each user allocated by LTE scheduler with the priority indicator which signals delay- or throughput-or packet loss awareness of different users and the buffer fullness at each time instant based on predefined thresholds which depends on of each user terminal-capabilities in order to efficient delivery video over less than good variable channel conditions that have different QoE requirements at end-users terminals. It is worth pointing out that adaptive algorithm with the optimized results could also be used to improve the performance of 4G LTE Wireless Networks resource allocation. The experimental simulation results indicate a significant performance improvement of the proposed very efficient algorithm compared to conventional schemes that require proportional media content scheduling on the network paths according to their available bandwidth.

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### REFERENCES


