Smoothing of Video Transmission Rates for an LTE Network

Khaled Shuaib and Farag Sallabi
Faculty of Information Technology, United Arab Emirates University
P.O Box 17551, Al-Ain, UAE
e-mail: {k.shuaib, f.sallabi}@uaeu.ac.ae

Abstract—Video smoothing techniques can be used to facilitate more effective transmission and to preserve better video quality. In this paper we develop a semi-optimal video smoothing approach to manage the transmission rates of MPEG-4 and H.264 video over a QoS-based wireless LTE network. The proposed technique utilizes a smoothing buffer with pre-defined thresholds to smooth the transmission rates while assuming minimal information about the video to be transmitted. The results obtained showed a significant improvements in smoothing transmission rate variability. In addition, we show a model for the wireless LTE channel and use it as a feedback to manage smoothing and regulate and map the transmission rates based on the availability of network resources.

Key words: Video, LTE, Wireless, Performance, QoS.

I. INTRODUCTION

3GPP [1, 2] is defining Long-Term Evolution (LTE), whose radio access is called Evolved UMTS Terrestrial Radio Access Network (E-UTRAN). LTE allows 3G operators to use new and wider spectrum (up to 20 MHz) while achieving higher data rates, lower latency and higher capacity to meet the increasing demand for enhanced broadband services by consumers. In general, LTE is being developed to satisfy the requirements that include: Downlink peak rates of more than 100Mbps and greater than 50Mbps in the uplink, low user latency, support for end-to-end QoS allowing different class of services and VOIP capacity deployment around three times that of UMTS. The main objective of a QoS-enabled network infrastructure is to ensure that the users get the desired experience they expect based on their service level agreement with the operator who owns and manages the network. On the other hand, from the operator point of view, applying QoS implies that it can optimize usage of limited network resources while satisfying customers.

LTE is supported by an evolved packet core (EPC) as part of the evolved packet system (EPS), which is designed by 3GPP to provide interoperability and seamless service continuity with existing mobile networks, supporting the market usage of any IP-based services including those with end-to-end QoS requirements. The evolved radio access network for LTE consists of NodeB that interfaces with the user equipment (UE). This node hosts the physical, MAC, radio link control, and packet data control protocol layers and offers several functionalities such as radio resource management, security, admission control, scheduling, and QoS support. The LTE physical layer provides shared channels to the higher layers using a 1ms transmission time interval (TTI), a frame of 10ms long and a subcarrier spacing of 15 kHz. LTE relies on hybrid automatic repeat request (HARQ) for rapid adaptation to channel variations and uses the concept of a physical resource block (PRB), which is a block of 12 subcarriers in one slot for bandwidth allocation. Each user is allocated a number of PRBs in the time–frequency grid which defines its bit rate.

In 3GPP the selected QoS approach is based on a combination of the resource and policy based admission control, differentiated services and measurement based admission control. In an LTE architecture, to support and maintain the QoS of in-progress sessions in a cell, it is important to admit a new radio bearer only if all the existing sessions and the new bearer can be guaranteed the desired QoS based on their requirements. QoS subscription information may be used together with policy rules such as: service-based, subscription-based or pre-defined internal policies, to derive the authorized QoS to be enforced for a service data flow [2]. In LTE, EPS bearers are of two kinds. When a guaranteed bit rate (GBR) value is configured to be permanently associated with an EPS bearer by an admission control function that exists at NodeB, this EPS bearer is referred to as a GBR bearer. Otherwise, an EPS bearer is referred to as a Non-GBR bearer. The GBR value is managed by the scheduling scheme at NodeB for the allocation of needed number of PRBs to achieve the desired bit rate. Each EPS bearer (GBR and Non-GBR) is associated with the following
bearer level QoS parameters: QoS class identifier (QCI) and allocation and retention priority (ARP). A QCI is a scalar value configured on the operator owned NodeB and used as a reference to access node specific parameters that control bearer level packet forwarding QoS policy. A one-to-one mapping of standardized QCI values to standardized characteristics is captured in [5].

The rest of the paper is organized as follow: Section 2 discusses video smoothing and section 3 presents the system model for the smoothing technique. Section 4 shows the proposed LTE wireless channel model and the obtained results and provides performance analysis. The paper is then concluded in section 5.

II. VIDEO SMOOTHING

Transmission of video over a wireless network is challenging due to the variability of video, bandwidth limitation and characteristics of wireless channels. When a video stream is encoded as VBR, bit allocation and distribution is varied depending on the complexity and motion of each scene. This is done to obtain an optimal video quality while not consuming more than needed resources. The video variability is very hard to measure and depends on the chosen encoding parameters of the video clip, mainly the mean encoding bit rate (CER) and the peak encoding bit rate (PER). The greater is the difference between these two parameters, the greater is the assumed variability in the video stream which results in great frame size variability. To mitigate for this variability for the transmission of video over a network, and for better provisioning of network resources certain measures are needed. To achieve this, traffic classifiers and conditioners or what are also called traffic shapers or-smoothers have been proposed by many researches previously [6, 7, 8]. Where traffic classifiers can be used to prioritize traffic types, the main concept applied by traffic shapers is to use one or more shaping buffers to control and adapt the rate at which the traffic is being sent over a wireless channel. Fig. 1 shows a network framework of how video can be transported over a wired/wireless network with smoothing of video done at the edge node just before the air interface i.e. NodeB in the case of LTE.

Two main extreme techniques of video smoothing have been mainly used in the literature: basic smoothing and optimal smoothing. In basic smoothing, video is transmitted at the average rate of N none overlapping successive frames. In this technique, the larger is N the less variability there is, but the larger is the delay. In optimal smoothing which has a greater complexity, the transmitting bit rate is minimized while guaranteeing an upper bound on delay and no over/under flows of the decoder buffer. This is achieved by using piecewise constant bit rate segments which are as long as possible. Optimal smoothing is only suitable for pre-encoded video since it computes the transmission rate schedule offline.
transmissions or through video analysis for pre-encoded streams. Another way to choose $\alpha$ would be based on the maximum allowed smoothing buffer delay. This will be defined in the next section where the video smoothing algorithm performance evaluations are conducted.

![Figure 2: Video Transmission Rates Representations](image)

### III. SYSTEM MODEL

The proposed video smoothing approach was implemented using a simulation program written in Java. The simulation program is based on a client/server paradigm. At the server, the video frames are generated every 33 ms, i.e. every frame period ($T_f$), and sent to NodeB over an assumed constant delay wired network using instantaneous video frame rates. At NodeB video frames are received and saved at a FIFO synchronized queue serving as a smoothing buffer. Data is then read from the queue based on $R$ utilizing an RTP frame length of 30 ms for transmission to the client which acted as a sink. Several MPEG-4 and H.264 video traces were used in the simulation. These video traces and their statistics were obtained from [9, 10, 11] and summarized in Table 1. The video traces were chosen to represent various video types (sport, movie, news). Corresponding MPEG-4 and H.264 traces used were best chosen for similar mean frame peak signal to noise ratio (PSNR) so that a better comparison can be made. All traces were encoded as 30 frames per second VBR CIF 352x288 with a group of picture (GOP) defined as G16B3 i.e. IBBBPBBBPBBBPBBBP [10,11].

In the proposed scheme, minimal information about the video traces was assumed to be known for applying smoothing, making it suitable for both pre-encoded and real-time video. There are two modes for the proposed smoothing algorithm, the conformant wireless channel mode and the variable wireless channel mode. In the first, we assume that the available wireless channel bandwidth can always meet the demanded transmission rate based on the agreed upon GBR. In the second, we use a wireless channel model with feedback to the smoothing buffer being provided on the available channel bandwidth so that the transmission rate is adjusted accordingly when needed. The LTE proposed channel model is discussed in the next section.

### TABLE 1: VIDEO TRACES USED

<table>
<thead>
<tr>
<th>Video Trace</th>
<th>Mean Frame Bit Rate (bps)</th>
<th>Peak Frame Bit Rate (bps)</th>
<th>Number of Frames</th>
<th>Mean Frame PSNR (db)</th>
</tr>
</thead>
<tbody>
<tr>
<td>H.264-Tokyo Olympics</td>
<td>144267.6</td>
<td>4182960</td>
<td>133125</td>
<td>35.557</td>
</tr>
<tr>
<td>MPEG4-Tokyo Olympics</td>
<td>278896.9</td>
<td>3358080</td>
<td>133125</td>
<td>34.829</td>
</tr>
<tr>
<td>H.264-Silence of the Lambs</td>
<td>68898.9</td>
<td>3168720</td>
<td>53997</td>
<td>37.598</td>
</tr>
<tr>
<td>MPEG4-Silence of the Lambs</td>
<td>194176.9</td>
<td>2592480</td>
<td>53953</td>
<td>37.116</td>
</tr>
<tr>
<td>H.264-NBC 12 News</td>
<td>197427.6</td>
<td>3393840</td>
<td>49521</td>
<td>33.131</td>
</tr>
<tr>
<td>MPEG4-NBC 12 News</td>
<td>420051.4</td>
<td>4248240</td>
<td>49521</td>
<td>33.375</td>
</tr>
</tbody>
</table>

The steps of the smoothing scheme algorithm are outlined in Fig. 3. T1 (first buffer threshold) in the algorithm is chosen to avoid any buffer underflow as R is chosen to guarantee that the maximum amount to be transmitted is no more than the current content of the buffer (B). T2 (second buffer threshold) is chosen to keep the startup delay down to a desired value, to maximize the transmission rate at the CER$_t$, to minimize the transmission at the PER, to avoid any buffer overflow and to keep the maximum buffering delay (D in seconds) below a certain desired limit. D in this case can be expressed by:

$$D = \left(\frac{T_f}{CER_t} * 8\right) + T_f$$

By limiting the maximum smoothing buffer delay to D, one can calculate the expected CER$_t$ and therefore the needed value of $\alpha$ based on CER$_t = CER (1+\alpha)$ as was indicated earlier.

The proposed algorithm is considered to be semi-optimal for two reasons: 1) the maximum buffering delay is not linked to the playback time of the video frames 2) the instantaneous amount of net credit/debt might not be zero i.e. the number of transmitted bytes at any instant of time is not conformant to the agreed upon average transmission rate. However, this should have been agreed upon before the start of any transmission and a final billing statement can be produced once the transmission is over to make for any needed adjustments. On the other hand, and for a risk of introducing additional delay i.e. larger maximum buffer fullness, step 3c can be modified to guarantee that R will not be above the agreed upon CER$_t$ unless there is available net credit. In this case R will be tied to the
number of bytes available as credit and step 3c will be modified as in Fig. 4. In the next section, results are obtained to investigate the performance of the proposed algorithm.

1. Choose two thresholds \( T_1 \) and \( T_2 \) where 
\[
T_1 = \left( \frac{\text{CER}_t}{8} \right) \times T_f; \\
T_2 = \left( \frac{\text{PER}}{8} \right) \times T_f;
\]

2. Pre-fill the buffer with video data until \( T_1 \) before starting to transmit any data over the network.

3. Choose a transmission rate, \( R \), based on the buffer fullness (\( B \)) in bytes as follow:
   a) If \( B < T_1 \), then 
   \[
   R = \left( \frac{B}{T_f} \right) \times 8 \text{ bps}; \\
   \text{Credit} = \text{Credit} + \left\lfloor \left( \frac{\text{CER}_t - R}{8} \right) \text{ bytes} \right\rfloor;
   \]
   b) Else if \( T_1 \leq B \leq T_2 \) then 
   \[
   R = \text{CER}_t \text{ bps};
   \]
   c) Else (i.e. \( B > T_2 \)), then 
   \[
   R = \max \left( \frac{\text{CER}_t}{8}, \left( \frac{B - \text{CER}_t}{T_f} \right) \times 8 \right) \text{ bps}; \\
   \text{Debt} = \text{Debt} + \left\lfloor \left( \frac{R - \text{CER}_t}{8} \right) \text{ bytes} \right\rfloor;
   \]

4. Read a video frame into the buffer every \( T_f \)

5. Transmit an RTP frame every RTP frame period

6. Go back to 3 and repeat until there are no video frames left to transmit.

7. When done, generate a final billing statement. This will be based on the agreed upon mean transmission rate and any difference between the gained credit and owed debt from step 3.

In this section we assume that the wireless channel bandwidth can accommodate the needed transmission rate determined by the smoothing algorithm as in Fig. 3. To look at the performance of the proposed algorithm, several experiments were conducted. Table 2 shows the general performance results when \( \text{CER}_t \) was set to be \( \text{CER} \). As can be seen from Table 2, when the smoothing technique is applied the value of \( R \) is roughly 50% of the time around the \( \text{CER}_t \) except for the H.264 NBC 12 News which is 67%. Adding up the percentages of time when \( R \) is either at \( R_1 \) or \( \text{CER} \) reflects the percentage of time where transmission is \( \leq \text{GBR} \) in an LTE network. The results also reveal that a \( \text{CER}_t \) greater than the \( \text{CER} \) is needed to minimize the \( \text{(Debt} - \text{Credit}) \) value for optimal overall transmission based on a traffic contract.

To look at the effect of smoothing on the rate variability we used the variability definition from [9, 10] given by
\[
V = \frac{\text{Standard Deviation of Transmission Rates}}{\text{Mean Transmission Rate}}.
\]

Figure 3: The proposed smoothing algorithm

If \( (\text{Credit} - \text{Debt}) > 0 \) 
\[
R = \max \left( \frac{\text{CER}_t}{8}, \left( \frac{\text{PER}}{8} \right) \times T_f \right) \text{ bps};
\]
\[
\text{Debt} = \text{Debt} + \left\lfloor \left( \frac{R - \text{CER}_t}{8} \right) \text{ bytes} \right\rfloor;
\]
\[
\text{Credit} = \text{Credit} - \left\lfloor \left( \frac{R - \text{CER}_t}{8} \right) \text{ bytes} \right\rfloor;
\]
Else \( R = \text{CER}_t \);

Figure 4: The proposed smoothing algorithm with R based on the availability of credit

IV. LTE CHANNEL MODELING AND PERFORMANCE ANALYSIS

A. Conformant Wireless Channel

<table>
<thead>
<tr>
<th>Video Trace</th>
<th>(Debt – Credit) bytes per second</th>
<th>Percentage at R1, CER, R2</th>
<th>Maximum B in bytes</th>
<th>CER_t Obtained / CER</th>
</tr>
</thead>
<tbody>
<tr>
<td>H.264-Tokyo Olympics</td>
<td>198.2</td>
<td>34.4, 52.6, 13.0</td>
<td>34514</td>
<td>1.0116</td>
</tr>
<tr>
<td>MPEG4-Tokyo Olympics</td>
<td>406.9</td>
<td>29.0, 48.9, 22.1</td>
<td>28142</td>
<td>1.0116</td>
</tr>
<tr>
<td>H.264-Silence of the Lambs</td>
<td>124.7</td>
<td>34.2, 52.7, 13.1</td>
<td>24986</td>
<td>1.0149</td>
</tr>
<tr>
<td>MPEG4-Silence of the Lambs</td>
<td>166.7</td>
<td>29.2, 50.9, 19.9</td>
<td>20910</td>
<td>1.010</td>
</tr>
<tr>
<td>H.264-NBC 12 News</td>
<td>285.8</td>
<td>23.5, 67.0, 9.5</td>
<td>26461</td>
<td>1.0118</td>
</tr>
<tr>
<td>MPEG4-NBC 12 News</td>
<td>524.3</td>
<td>28.3, 53.8, 17.9</td>
<td>32784</td>
<td>1.010</td>
</tr>
</tbody>
</table>

Fig. 5 shows the results on variability for 6 video traces when transmitted using the proposed smoothing technique and without any smoothing applied i.e. each video frame is being transmitted using a transmission rate calculated based on the frame size. As can be seen from Fig. 5, improvement in rate variability is above 20% for all traces with more than 46% for the H.264 NBC 12 News clip. This is mainly due to the fact that \( R \) is at \( R_1 \) or \( \text{CER} \) the majority of the time. This can also be captured from Fig. 6 which shows a sample of 100 transmission rates for the H.264 Silence of the Lamb video trace.
Finally, we looked at the performance of the algorithm with step 3c in Fig. 3 being replaced with the one in Fig. 4. In this case, R was tied to the availability of credit when the buffer content was above T2. This case was run for the H.264 Silence of the Lamb clip and the results showed $V = 2.4$ and 172 buffer overflow instances indicating a poor performance. This supports the idea that immediate conformant to a service level is usually less effective than long term conformance.

B. Variable Wireless Channel

In this section we present a model of an LTE wireless channel and use it as an input to the smoothing algorithm.

![Figure 6: Effect of smoothing on transmission rates](image)

**LTE Channel Model**

In LTE PRBs can be modeled as a finite-state Markov channel (FSMC). The states are determined by partitioning the average received SNR range to $N+1$ intervals, where $N$ is the number of modulation schemes. Let $S = \{S_0, S_1, \ldots, S_K\}$ denote the state space of a stationary Markov chain with $K$ states. The state space $\{S_j\}$ contains $K$ different PRB states with corresponding bit per symbol rate (Constellation size). Let $\pi_i$ be the steady-state probability and $p_{ij}$ be the state transition probability, $i, j \in \{0, 1, \ldots, K-1\}$. Since a stationary Markov process has the property of time-invariant transition probabilities, the transition probability is independent of time and can be indicated as:

$$p_{ij} = \Pr(S_{n+1} = j|S_n = i), \quad n = 0, 1, \ldots, i$$

Due to slow fading, the transitions happen only between adjacent states, the probability of transition exceeding two states is zero; i.e.,

$$p_{ij} = 0, \quad |i-j| > 1, \quad i, j \in \{0, 1, 2, 3\}$$

Multipath propagation environment is best modeled by Rayleigh distribution. With additive Gaussian noise, the received instantaneous SNR $\gamma$ is distributed exponentially with probability density function as specified in [12]:

$$P(\gamma) = \frac{\bar{\gamma}}{\gamma} e^{-\frac{\gamma}{\bar{\gamma}}} \quad \gamma \geq 0$$

Where $\bar{\gamma} = E\{\gamma\}$ is the average received SNR.

Assume one-step transition in the model corresponds to the channel state transition after one sub-frame time period $T_f$ (1 ms). A received sub-frame is said to be in channel state $\pi_k, k = 0, 1, 2, 3$, if the SNR values in the sub-frame varies in the range $[\gamma_k, \gamma_{k+1}]$. Let $\gamma_0 < \gamma_1 < \gamma_2 < \gamma_3$ be the thresholds of the received SNR. Then the PRB is in state $\pi_k$ if the received SNR is $\gamma_k < \gamma < \gamma_{k+1}$. To avoid deep fade, no data are sent when $\gamma_0 \leq \gamma \leq \gamma_1$.

The following figure shows the K-state noisy wireless channel modeled by FSMC:

![Figure 7: K-state wireless channel FSMC model](image)

The steady-state probabilities of the channel states are given by [12]:

$$\pi_k = \int_{\gamma_k}^{\gamma_{k+1}} p(\gamma) d\gamma = e^{-\frac{\gamma_k}{\bar{\gamma}}} - e^{-\frac{\gamma_{k+1}}{\bar{\gamma}}}$$

Transitions are allowed between two adjacent states only, so the transitions states for the FSMC can be determined as in [12]:

$$p_{k,k+1} = \frac{N(\gamma_{k+1}) T_f}{\pi_k}, \quad k = 0, 1, 2, 3$$
where \( k = 0, 1, 2, 3 \)  \( (6) \)

Where \( N(.) \) is the level crossing function given by [12]:

\[
N(y) = \sqrt{\frac{2\pi}{y}} f_d e^{-\left(\frac{y}{2}\right)}
\]

Where \( f_d \) is the maximum Doppler frequency; \( f_d = \frac{v}{\lambda} \), \( v \) is the velocity and \( \lambda \) is the wavelength.

**Video Smoothing with Channel Feedback**

The average available bandwidth of the wireless channel over the period of an RTP frame, 30 ms, as allocated by the scheduler at NodeB is fed back to the smoothing buffer and used to regulate the transmission rate. In this case \( R \) will not just depend on the fullness of the smoothing buffer, but will also depend on the available channel bandwidth. We assume that the average channel bandwidth (\( R_c \)) does not change within the length of an RTP frame i.e. 30 ms. Although this is a far fetch assumption, we use it to show the effect of channel condition feedback on smoothing. For this mode, step 3 in the algorithm of Fig. 3 is modified as shown in Fig. 8.

Choose a transmission rate, \( R \), based on the buffer fullness \( (B) \) in bytes and the average channel bandwidth as follow:

\[
\begin{align*}
\text{a)} \quad & \text{If } \{ B < T1 \}, \text{ then } \\
& R = \text{Min} \{ (B / T_f) \times 8 \text{ bps}, R_c \}; \\
& \text{Credit} = \text{Credit} + \lfloor (CER_t - R) / 8 \rfloor \text{ bytes}; \\
\text{b)} \quad & \text{Else If } \{ T1 \le B \le T2 \} \text{ then } \\
& R = \text{Min} \{ \text{CER}_t, R_c \} \text{ bps}; \\
\text{c)} \quad & \text{Else (i.e. } B > T2), \text{ then } \\
& R = \text{Min} \{ \text{Max} \lfloor \text{CER}_t, \text{Min} (\text{PER}, ((B- T2) / T_f) \times 8) \rfloor, R_c \} \text{ bps}; \\
& \text{Debt} = \text{Debt} + \lfloor (R - \text{CER}_t) / 8 \rfloor \text{ bytes};
\end{align*}
\]

![Figure 8: Smoothing with channel feedback](image)

Based on data generated from the proposed channel model for a system bandwidth of 10 MHz, the average bit rate per PRB is shown in Fig. 9. We assume that the scheduler at NodeB is to allocate an average wireless channel bandwidth calculated based on the average bit rate for one PRB over a number of TTIs, within an LTE frame of 10 ms. \( R_c \) is chosen to be as close as possible to the average encoding rate of the video clip being transmitted. On the other hand, the assumed maximum wireless channel bandwidth that can be allocated is based on the average bit rate of one PRB per every TTI within an LTE frame. For example, when transmitting the H.264 Silence of the Lambs clip which has a CER around 70 Kbps, one PRB is allocated every 4 TTIs within an LTE frame. Based on this, and the average PRB bit rate generated, the average scheduled channel bit rate will be around 80 Kbps and the maximum will be around 340 Kbps. The transmission rate variability results for the Silence of the Lambs H.264 and MPEG-4 are shown in Fig. 10. Although the transmission rate variability was reduced with channel feedback, several buffer overflows and excessive delays occurred due to the conservative assumption made on the maximum channel bit rate being allocated. This can be avoided given that a higher bit rate can be allocated by the scheduler where more than one PRB can be used per TTI. As part of our future work a scheduling algorithm will be developed and integrated to optimize the allocation of bandwidth while guaranteeing the desired video quality.

![Figure 9: Average PRB bit rate per TTI](image)

![Figure 10: Variability results with and without channel feedback](image)

**V. CONCLUSIONS**

In this paper a video smoothing technique is proposed for the transmission of video over an LTE network. The results obtained showed good improvements in
transmission rate variability while having no losses due to smoothing when the wireless channel was assumed to be conformant with the client requirements. The smoothing approach was mapped to be used in an LTE network where a guaranteed rate is assumed to be available in one mode and where a channel model was used to regulate the rate based on the average channel bandwidth in another mode. Scheduling of PRBs is crucial to optimize utilization of resources while satisfying the need of clients. Therefore, future work will focus on scheduling algorithms.

REFERENCES