Simulation-based Performance of Transport Protocols Using MPEG-4 Traffics over 4G Network

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Abstract— Four Generation (4G) networks are the boiling connection over the airwaves since it can support the far geographic distance over the world, especially are to the popularity of the wireless devices. The Long-Term Evolution (LTE) is often called the 4G. Nowadays, telecommunications, video traffic has burgeoned under the development of LTE, the truly underlying access technology of 4G networks. During LTE deployment, different transport protocols are the most recommended and widely studied such as Transmission Control Protocol (TCP), User Datagram Protocol (UDP) and Datagram Congestion Control Protocol (DCCP), which can perform differently on 4G networks depends on the network’s scenario as well as parameters setting. Although the LTE deployment is rapidly pace, there is a lack of performance evaluation of its protocols. Therefore, an extensive analysis is needed to evaluate the performance of various protocols for high end applications like multimedia applications. Supporting these multimedia applications with varied quality of service (QoS) requirements while maximizing the resource utilization is a challenging task.

To achieve the objective of this study, this work tries to answer the following questions: are the old protocol suit with this new technology; which one has the best performance and; which one has the greatest effect on throughput, delay, packet loss and packet delivery ratio. In this study, our analysis and compare are on the performance of different the Internet transport protocols for streaming video via using simulation environment. Through the Network Simulation 3 (NS3), the performance of transporting an MPEG4 video stream, including throughput, delay, packet delivery ratio and packet loss are analyzed at the base station through TCP, UDP, and DCCP protocols over 4G-LTE technology. The results show that DCCP has a better throughput, and a lesser delay, but at the same time it has more packet losses than UDP and TCP. Based on the results, DCCP is recommended as a transport protocol for real-time video.


I. INTRODUCTION

Wireless communications networks become an everyday commodity due to its flexibility of access and usage for providing high rate data anytime and anywhere [1]. These types of networking have been developed from being a costly technology for a few chosen individuals to today’s ubiquitous systems, which utilized by a larger part of the population of the world. Wireless technologies can be divided into five generations. First Generation (1G) was the analog radio systems. Second Generation (2G) was the first digital wireless systems. Third Generation (3G) was the first wireless systems, handling broadband data. While the Fourth Generation (4G) is often a Long-Term Evolution (LTE) [2]. As a result of the advancements in wireless communication the network traffic has been increasing, especially as the number of nodes increased. The LTE delivered higher data rates and met the burgeoning data demand [3].

In today’s telecommunications, the video traffic has burgeoned under the development of LTE, the truly underlying access technology of 4G networks [2][4]. During LTE deployment, three transport layer protocols are the most recommended and widely studied, they are UDP, TCP and DCCP. The data, delivering rate are significantly influenced by the performance of the transport protocol that is used in the scenario of wireless networking [5]. Providing applications of multimedia, for example real time video, video and audio streaming and video conferencing that is nowadays on the Internet. These applications are deemed to be challenged and demanded over wireless networks for the multimedia application (video). Hence, wireless networks have lack of QoS and limited bandwidth support for the air interface restrains wide deployment of these multimedia applications. The performance of an application can be varied significantly when different transport protocols are employed. Usually, UDP is used to transport of multimedia applications. However, the performance of UDP is not satisfactory due to many constraints, for example, lack of congestion control mechanism [6].

Although the LTE deployment is rapidly pace, there is a lack of performance evaluation of its protocols. Therefore, an extensive analysis is needed to evaluate the performance of various protocols for high end applications like multimedia applications. The problematic behavior of the three protocols in multimedia applications entails highlighting the pros and cons of their performance [7]. Existing comparison studies [8][9][10][11] did not consider the performance of the TCP, UDP and DCCP protocols in transferring video under LTE access environment. The best performing protocol in video transferring even does not characterize because of the conflicted conclusions. Therefore, providing an integral analyzing study about the performance of the three protocols in the LTE environment will help developers and researchers in choosing the proper protocol to be used in video traffic.

In this paper, we are analyzing and comparing the Internet transport protocols, which are used for streaming video (i.e.,
MPEG-4 over LTE infrastructure technology. Moreover, to show the strength and weakness of TCP, UDP and DCCP, by simulating it in the latest NS3 repository and gives the intensive results based on simulations each protocol spread. Hence, we expect through our result to see that which protocol will perform better than others, especially, when there are payloads through MPEG-4 over slandered LTE stations. In particular, transport layer protocols are provided in Section II. While in Section III, presents the Fourth Generation (4G) / Long Term Evolution (LTE). The preceding works are described in the next section. Simulation setup and evaluation metrics are described in Section V. Section VI shows the results with discussions. Finally, the conclusion is presented in Section VII.

II TRANSPORT LAYER PROTOCOLS

The services and features of TCP, UDP, and DCCP transport layer protocols are shown in Table 1. Each has its own features and relevance for particular application under specific environments.

<table>
<thead>
<tr>
<th>Features and Services</th>
<th>TCP</th>
<th>UDP</th>
<th>DCCP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Reliable</td>
<td>X</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Connection Oriented</td>
<td>X</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Congestion Control</td>
<td>X</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Sequence Number</td>
<td>X</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Header Size</td>
<td>X</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Support of ECN</td>
<td>X</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Error Checking</td>
<td>X</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Streaming</td>
<td>X</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Method of transfer</td>
<td>Segment</td>
<td>Datagram</td>
<td>Datagram</td>
</tr>
<tr>
<td>Power Consumption</td>
<td>High</td>
<td>Low</td>
<td>Medium</td>
</tr>
<tr>
<td>Packet size</td>
<td>8 bytes</td>
<td>16 bytes</td>
<td>12 or 16 bytes</td>
</tr>
</tbody>
</table>

1. Transmission Control Protocol

TCP is an Internet Protocol Suite (IPS) core protocol that functions well when two end-systems at a higher level interact. However, the stream of bytes provides packet reliability through TCP [12] whereas this protocol also performs some management tasks, such as controlling rate and message during the regulating of traffic congestion and communication. TCP acts as a transport layer that hides the underlying systems administration points of interest from corresponding provisions [13]. One of the best cases of TCP applications is the web browser [8]. So, other common main applications include web server, e-mail, and file transfer. In Fig 1, the TCP functioning layer has been shown.

2. User Datagram Protocol

UDP is a transport layer protocol available for using with the IP network layer protocol [14]. However, the protocol does not have the ability for the handshaking mechanism to guarantee packet reliability, data integrity and packet ordering. UDP is a connection-less protocol working on transport layer [15]. The header size of UDP protocol is 8 bytes, including the field source port address, destination port address, length and checksum. All fields are of 16 bits, i.e., 2 bytes each. It is unreliable due to the lack of acknowledgement in the data transfer. Thus, an application software running over UDP should deal precisely with the issues of E2E communications which a connection-oriented protocol would have managed. These issues may be any of the retransmission for consistently delivery, flow control, packetization and reassembly, and congestion control etc. It is fast due to no connection establishment and tear down phase [16]. Therefore, it is more suited to small applications which do not need reliable connection.

The most common use of UDP is in the Domain Name System (DNS) services. To get the IP address for a requested URL from DNS. Other application layer protocols which use UDP as a carrier protocol on transport layer are Dynamic Host Configuration Protocol (DHCP) [17], Routing Information Protocol (RIP) [18] and Voice over IP (VoIP) [19]. At transport layer, the UDP is located in Fig 2.

3. Datagram Congestion Control Protocol

The DCCP is a convention of the transport layer with dependable association setup, blockage control, and characteristic transaction competence [20]. However, the primary configuration goal and broadening over the conventional UDP is the affirmation of blockage control for datagram streams. At that point, DCCP has a scheduled outline that divides the focal part purpose of the convention from the use of the blockage control instrument. DCCP is envisioned for multimedia functions, for example streaming media that can be assisted from manipulation over the adjustments between reliable and delay in order delivery. TCP may not be suitable for these kind of applications because congestion control and reliability in-order delivery can result in arbitrarily long delays.
The UDP protocol can avoid long delays, nevertheless for congestion control the governing application will have to deal on its own. DCCP has provided a built-in congestion control mechanism, including ECN support, for unreliable datagram flows, to avoid the arbitrary delays related to TCP. DCCP characteristic is a connection quality on whose value the two end connector makes an agreement. Several advantages of a DCCP association are coordinated by feature. For example, congestion control mechanism is used in the two half-connections. The endpoints attain the arrangement in the course of option of exchange negotiations in DCCP headers.

According to RFC 793, DCCP implementations follow TCP’s general principle of robustness, i.e. “Be conservative in what you do, while be liberal in what you accept from others.” DCCP is a transport layer protocol that deploys unicast, bidirectional connections of congestion-controlled and unreliable datagrams. Fig 3 locates the DCCP in OSI.

III FOURTH GENERATION / LONG TERM EVOLUTION

Fourth Generation blankets over billions of supporters; more than 80% of the worldwide versatile business sector [2]. However, the number of worldwide subscribers, in 2008, utilizing High-Speed Packet Access (HSPA) networks surpassed 70 million [21]. So, HSPA is a 3G evolution of GSM that supports high-speed data transmission by means of WCDMA technology. The global use of HSPA technologies among clients and businesses have accelerated, representing continuous traffic growth for high-speed wireless networks worldwide, whereas extensive efforts are proceeding on the 3G Partnership Project (3GPP) to create a novel criterion for the development of GSM/HSPA technology towards a packet-optimized method known as LTE, with the intention of meeting the continuous demands in Internet traffic [22].

The main purpose of the LTE standard is to design plans for a new radio-access technology that can suitably handle higher data rates and is beneficial for low latency, and better spectral efficacy [5]. However, the spectral efficacy target of the LTE scheme is 3-4 times more than the existing HSPA scheme [23]. These uncompromising spectral efficacy targets need to push the technology enfolded by using advanced air-interface mechanisms. For example, low-PAPR orthogonal uplink multiple access based on the Multiple-Input Multiple-Output, Single-Carrier Frequency Division Multiple Access, inter-cell interference mitigation methods, low latency channel structure, multi-antenna technologies, and Single-Frequency Network broadcast [21].

IV RELATED WORK

Nowadays, most of the multimedia applications are utilized TCP, UDP and DCCP as a primal transport layer protocols. In literatures, several analyzing studies deal with these protocols under various environments. However, existing studies do not encircle their performance in LTE environment, in spite of its fundamental rule in accelerating video traffic in today’s telecommunications. The performance of protocols is an important part to be measured in the evaluation of any network environment.

Nor et al., [24] performance of standard and paced TCP when coexisting with DCCP over long and short delay link networks is investigated. The topology used six nodes. While evaluating the results used throughput, average delay, jitter and packet loss as a metric. The results show that packet pacing enhances the TCP flow when it comes to jitter as well as bandwidth for a long delay link, while the performance of standard TCP flow performs fairly with the paced TCP flow for a short delay link. Furthermore, it is shown that TCP pacing is only suitable for implementation over a network link with long propagation delay (e.g., wireless or satellite links). Whereas it is not recommended for using TCP pacing in normal short delay link.

According to Chughtai, Malik and Yousaf in [11] results in their simulation, performance of SCTP, DCCP and UDP protocols the transport of video traffic (MPEG-4) over WiMAX as underlying access technology was analyzed. While in [25] the performance analysis of TCP and UDP voice in a static wireless multi-hop network by simulation was investigated. Measurement is carried out in a network consisting of eight stations with IEEE 802.11b interfaces in a ring eight node topology. They analyzed the effect of collision avoidance mechanism, packet size, and hop count. While on the performance metrics using delay, jitter, and throughput as well.

Nosheen et al., [9] presented a comparison study to evaluate the Quality of Service (QoS) when sending MPEG-4 video considering transport protocols DCCP and SCTP over wireless local area network. They used three (3) nodes. Throughput is used as an evaluation metric. If the transfer files video is less than 4Mbps then the throughput of both DCCP and SCTP are almost 100% without packet loss, but if it becomes more than 5Mbps then the DCCP maintains its throughput also it shows minimum packet loss, and SCTP losses its performance. The DCCP protocol is better than SCTP protocol in the case of delay behavior. It can be concluded from this study that both DCCP and SCTP are achieving better throughput than UDP and the QoS for DCCP better satisfies than SCTP in case of transport video traffic.
Azad, Mahmood and Mahmood [10] addresses the shortcoming of not considering all metrics by using throughput, delay, packet loss, and jitter in assessing the performance of DCCP, TCP, and UDP protocols. Also, they used six nodes instead of three. The protocol behavior in video applications is evaluated using MPEG-4 video coding with clear topology for wired networks. In this paper, the result shown, DCCP protocol can be utilized as a transport layer protocol for video applications and assures superior QoS than others for transmitting video under congestion.

V SIMULATION SETUP AND PERFORMANCE METRICS

There are many network simulators that are used by the network researcher to implement and evaluate the simulation scenario like OPNET [26], OMNET++ [27], NS2 [28] or NS3 [29]. In this project, NS3.22, which was released in February 2015, is used to implement the scenario and evaluate the transport layer protocols (TCP, UDP and DCCP) for sending video streaming in LTE 4G environments.

1. Selection of Simulation

Network Simulation 3 (NS3) is a discrete event simulator, which can be used for the implementation of numerous applications. The NS3 project started in [30] as an open source. The code of NS3 is freely available to the research community and students to be used as the foundation for the implementation. In NS3, there are numerous external animators and tools. This simulation platform provides users with a single, integrated Graphical User Interface environment, data analysis, and visualization is also provided. It has become very famous simulator in the methodology of network and development of the LTE [31]. The development of the LTE module for NS3 has innovated out during the Google Summer of Code 2010. This module offers a primary implementation of LTE units, which consist of the proliferation models; PHY and MAC layers [32]. On the other hand; the proposed scenario permits the recreation of a few significant parts of LTE frameworks. Additionally, it provides a great foundation for further augmentations.

2. Traffic Model (video)

Moving Picture Expert Group (MPEG4) video coding standard is the most appropriate format for video communication over the Internet [22]. Utilized for low touch rates, MPEG4 empowers true pictures to exist together with PC created partners. MPEG4 makes conceivable the detachment of genuine from workstation created pictures for distinctive medications emerging from the interface with clients [33].

The primary role of the system is the competence for constant versatile encoding, which upgrades system use and empowers MPEG4 senders to be more receptive to changes in system conditions. MPEG4 creates feature in three separate edges, for example, (I, P, and B) that assist to encode unique parcels of the feature information in distinctive levels of value. The video and audio formats standardized as MPEG4 are apparently suited for streaming media since the quality of media achieved at lower bitrates and have become very popular in streaming media applications [34].

3. Environment and Parameters of Simulation

Table 2 shows the summary of the model parameters that was used for the simulation experiment.

![Table 2: The parameters of simulation scenario](image)

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Scenarios</th>
<th>Scenarios</th>
<th>Scenarios</th>
</tr>
</thead>
<tbody>
<tr>
<td>Protocol</td>
<td>TCP</td>
<td>UDP</td>
<td>DCCP</td>
</tr>
<tr>
<td>No. of nodes</td>
<td>10, 20, 30, 40</td>
<td>10, 20, 30, 40</td>
<td>10, 20, 30, 40</td>
</tr>
<tr>
<td>No. of packet</td>
<td>250000</td>
<td>250000</td>
<td>250000</td>
</tr>
<tr>
<td>Packet size</td>
<td>1000 byte</td>
<td>1000 byte</td>
<td>1000 byte</td>
</tr>
<tr>
<td>Connection channel</td>
<td>P2P</td>
<td>P2P</td>
<td>P2P</td>
</tr>
<tr>
<td>Net device type</td>
<td>LTE</td>
<td>LTE</td>
<td>LTE</td>
</tr>
<tr>
<td>Interval</td>
<td>100ms</td>
<td>100ms</td>
<td>100ms</td>
</tr>
<tr>
<td>Mobility model</td>
<td>CPM model</td>
<td>CPM model</td>
<td>CPM model</td>
</tr>
<tr>
<td>Channel data rate</td>
<td>10 Mb/s</td>
<td>10 Mb/s</td>
<td>10 Mb/s</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>MPEG-4 Characteristics</th>
</tr>
</thead>
<tbody>
<tr>
<td>Frame high</td>
</tr>
<tr>
<td>Frame width</td>
</tr>
<tr>
<td>Data rate</td>
</tr>
<tr>
<td>Total bitrate</td>
</tr>
<tr>
<td>Frame rate</td>
</tr>
</tbody>
</table>

4. Evaluation Metrics

The evaluation step is very important to evaluate the transport layer protocols for video transmission in LTE 4G networks by using a simulation scenario described in the previous sections. This performance evaluation will be done by the same metrics that are used by other researchers in literatures [9], [25], [35].

A. Throughput: It is defined as the amount of effective packet delivery over a communication channel. Usually, it is measured either data packets per second or bits per second. The performance is good when the throughput is high. The following formula is often used to calculate Throughput value:

\[
\text{Throughput} = \frac{\text{Number of Received Packets}}{\text{LastPacketSendTime} - \text{FirstPacketSendTime}} \quad (1)
\]

B. Packet loss: It is defined as the difference of the total number of packets sent by the source and the total number of packets received at the receiver. The performance is good when the packet loss is low. The following formula is often used to calculate packet loss value:

\[
\text{Packet Loss} = \sum \text{Packets Send} - \sum \text{Packets Received} \quad (2)
\]
C. **Delay**: It is defined as interval that packets experience when travelling across several networks from a single to multiple gadgets. The following formula is often used to calculate Delay value:

\[
\text{Delay} = T_r - T_s \ldots \ldots (3)
\]

In the equation, ‘Ts’ represents the time of transmitting the specific packet whereas ‘Tr’ stands for the reception time for the packet. The performance is good when the Delay is low.

D. **Packet Delivery Ratio (PDR)**: It refers to the number of packets effectively delivered to an endpoint as compared to the amount of packets that has been sent out by the sender. The performance is good when the PDR is high. The following formula is often used to calculate PDR value:

\[
PDR = \frac{\sum \text{Total No. of Received Packets}}{\sum \text{Total No. of send Packets}} \ldots \ldots (4)
\]

VI EVALUATION OF RESULTS

The following simulations are done to measure the performance of the TCP, UDP, and DCCP. The parameters used and their performance in these are explained below. To study the impact of the number of nodes on performance of the network, the maximum number of nodes was varied as 10, 20, 30, 40 and 50. The network was simulated for packet size was 1000 bytes. Figures 4-7 shows the number of nodes impacted for TCP, UDP, and DCCP regarding the various performance metrics.

The throughput in the network refers to the rate of successful message delivery over a communication channel. Fig 4 showed that the DCCP protocol has a good throughput in the environment of the LTE network of 10 nodes. The scenario here supposes all the ten nodes are sending the MPEG-4 video file at a same time to the base station node.

Also, we can show the continuity of the DCCP protocol in its good throughput in the environment of the LTE network if the number of nodes increase to 20, 30, 40 and 50 nodes. The results confirm the stability of this protocol even with an increasing number of video files sending when increasing in a number of nodes than the behavior of other protocols. Furthermore, here as the number of nodes increases the throughput of complete network will get improved. The consistent growth of graph shows that the network is capable to handle all these nodes number. To get the peak performance, there is no bottleneck up to this limit of node numbers. As the nodes increase the throughput grows too high.

The DCCP protocol has good throughput than TCP and UDP protocols because it has two mechanisms employed inside it, which are: congestion control (controlling the packets sent to the network when it became greater than available network capacity) and flow control (controlling the traffic size when the sender being sent up to the limit it receives response from the receiver).

![Fig 4. The results of comparing the throughput of TCP/UDP/DCCP protocols with different node density](image)

As shown in the Fig 5, the DCCP protocol has better results than TCP and UDP because the delay time is less than the other protocols. TCP protocol needs at the beginning more time to establish the connection. Also, this establishment of connection affects the number of nodes. Because of when the number of nodes is increased definitely the time delay also increases. This increase happens more in the wireless than wire because the layer two in the wireless needs acknowledgement (ACK) the RTS/CTS as well as layer three (ACK). Besides, a wireless network uses media share not like wire. Compared to all three scenarios for Average Delay time for TCP, UDP, DCCP, the UDP protocol shows consistently more delay due to connecting less flow of the data over the network.

TCP protocol experiences the largest delay due to the congestion control mechanism of the TCP. The reliability feature of TCP consumes time so cost of unreliability in UDP and DCCP is in terms of time. And DCCP outperforms these both conventional connection-less and connection-oriented protocols in case of delay. Comparative analysis of TCP, UCP and DCCP protocols for 10-50 node scenario shows the DCCP protocol is the best protocol regarding to delay time.

![Fig 5. The results of comparing the delay of TCP/UDP/DCCP protocols with different node density](image)
In case of packets with incomplete success to transmit and receive, then packet loss is happening in the end and the video stream becomes interrupted. It can mean much slower upload and download speeds, pauses with streaming media or poor quality VoIP audio. Moreover the packet loss happens in the wireless network more than wire network because of sharing media among nodes. The result is shown in the Fig 6, the TCP protocol has good result while the DCCP protocol has the worst. This result for 10 nodes broadcast file video to the server at the same time. Also, there is no big difference when increase the nodes to 20, 30, 40, 50 as well. We can show here in this figure; the amount of loss packets increases as the number of nodes increased. This is due of the base station became the bottleneck which is affected with the number of nodes in our network topology since all the nodes send packets at a same time to one base station.

The packet delivery ratio is the rate of packets arrived at the receiver node in comparison to the total number of packets sent from the sender node. PDR for TCP socket varies, i.e., minimum 94 % to 99% approximately, which is quite good result for any network. The TCP protocol uses (ACK) while establishes the connection that is why it has good Packet Delivery Ratio. Also, the result shows the UDP protocol has good PDR than DCCP. Hence, as shown in Fig 7, the DCCP protocol is the worse if we measure PDR, i.e., it is about 75%-85%. This result would be different if we remove the constraint. This leads us to make the component of hardware which will have a big memory buffer to overcome the packet loss. And nowadays memory is available in terabytes, so it is not an issue at all.

Overall, the results are highly related with the congestion concept that is happening if the sender delivers more packets than the receiver can keep. DCCP offers a method to achieve access to congestion control methods without implementing them at the application layer of the OSI model. It has license the flow-based semantics such as TCP, but does not offer reliable in-order transmission. DCCP is helpful for applications with timing restrictions on the data transmission. Such applications consist of multiplayer online games, streaming media, and Internet telephony. At present, such application has streaming media. Such applications consist of timing restrictions on the data transmission. Such applications consist of multiplayer online games, streaming media, and Internet telephony. At present, such application has streaming media.

Results show that the TCP protocol has a good throughput when the number of nodes becomes 10-50. But the UDP and TCP protocol have less throughput if compare with DCCP. The difference throughput between UDP and TCP is small difference even with this small difference the UDP is better than TCP protocol. Also the result shows the TCP and UDP protocol have highest delay than DCCP. While in cases of packet loss and PDR, TCP has better performance than UDP and DCCP.

VII CONCLUSION

Video streaming demands more bandwidth and high quality of communication. The main contribution in terms of the quality of communication is the development of LTE technology, which helped in increasing the data throughput and decreasing the latency. Through LTE environment the transportation layer protocols were the dominant players of the latest advancements in multimedia applications. These advancements motivate the need for evaluating the performance of the prominent protocols, i.e. TCP, UDP and DCCP, in MPEG-4 video data transferring in LTE. There is a shortage in the literature review in analysing these protocols for transporting this kind of data in this important environmental, this paper came to produce new experimental insights, conclusions, clarifications and results to fill this research gap. The TCP, UDP, and DCCP protocols are analyzed on various performance metrics such as delay, throughput, packet loss and packet delivery ratio. As per the simulation results on network simulator, DCCP protocol outperforms the other conventional connection-oriented and connection-less protocols in delay and throughput. The TCP performance gave a maximum packet delivery ratio and minimum packet loss count due to its connection oriented architecture. At the end, for multimedia applications where the packet loss difficult to be handled, the developers should go to TCP otherwise DCCP is the best suited with good throughput in MPEG-4 video streaming over LTE environment.
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