STREAMING ALGORITHM FOR SEAMLESS VIDEO SHARING OVER CLOUD

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Abstract — **The mobile phone plays an entertainment role in human life with the help of heterogeneous network by accessing multimedia content. But users faces a certain problems while access multimedia like long buffering time and intermittent disruption due to gap between demand traffic and link capacity, along with time varying link condition. The proposed technique focus on providing adaptive video streaming and seamless video sharing .To construct a private agent in cloud with video streaming algorithm to provide adaptive video streaming by reducing buffering duration in streaming and maintain seamless connectivity and acceptable level of quality to end users.**

Index Terms — **Heterogeneous wireless networks, Adaptive video streaming, mobile cloud computing, Video Streaming algorithm and video sharing.**

I. Introduction

In the past few years, increasingly more traffic is accounted by video streaming and downloading over mobile networks. But video streaming is not so challenging in wired networks, mobile networks have been suffering from video traffic transmissions over scarce bandwidth of wireless link bandwidth (like 3G and LTE). Soaring video traffic demands from mobile users are rapidly overwhelming the wireless link capacity. So Mobile users faces trouble while video streaming like long buffering time and intermittent disruption due to limited bandwidth and link condition fluctuation. And also mobile users get problem on video sharing like termination of connection due to handoff while move over heterogeneous wireless networks on cloud.

There are many techniques are placed on how to improve the quality of the video streaming and seamless connectivity to mobile users they are:

Scalability: Mobile video streaming services should support a wide spectrum of mobile devices; they have different video resolutions, different computing powers, different wireless links (like 3G and LTE) and so on. Also, the available link capacity of a mobile device may vary over time and space depending on its signal strength, other users traffic in the same cell, and link condition variation. Storing multiple versions (with different bit rates) of the same video content may incur high overhead in terms of storage and communication.

Adaptability: Traditional video streaming techniques designed by considering relatively stable traffic links between servers and users, perform poorly in mobile environments . Thus the fluctuating wireless link status should be properly dealt with to provide ’tolerable” video streaming services. To address this issue, we have to adjust the video bit rate adapting to the currently time-varying available link bandwidth of each mobile user. Such adaptive streaming techniques can effectively reduce packet losses and bandwidth waste. Video Streaming algorithm [VS] and adaptive streaming techniques can be jointly combined to accomplish effectively the best possible quality of video streaming services.

In this paper, we design an adaptive video streaming and prefetching framework for mobile users with above objectives in mind. We need to constructs a private agent for each mobile user in cloud computing environments to provide an adaptive mobile video streaming which offers the best possible streaming experiences by adaptively controlling the streaming bit rate depending on the fluctuation of the link quality and adjusts the bit rate for each user leveraging the Video Streaming algorithm. The private agent of a user keeps track of the feedback information on the link status. And also provide an efficient mobile video sharing supports distributing video streams efficiently by facilitating a 2-tier structure: the first tier is a content delivery network, and the second tier is a data center. With this structure, video sharing can be optimized within the cloud.

II. RELATED WORK

A. Adaptive Video Streaming Techniques

In the adaptive streaming, the video traffic rate is adjusted on the fly so that a user can experience the maximum possible video quality based on his or her link's time-varying bandwidth capacity [2]. There are mainly two types of adaptive streaming techniques, depending on whether the adaptivity is controlled by the client or the server. The Microsoft's Smooth Streaming [27] is a live adaptive streaming service which can switch among different bit rate segments encoded with configurable bit rates and video resolutions at servers, while clients dynamically request videos based on local monitoring of link quality. Adobe and Apple also developed client-side HTTP adaptive live streaming solutions operating in the similar manner. There are also some similar adaptive streaming services where servers controls the adaptive transmission of video segments, for example, the Quavlive Adaptive Streaming. However, most of these solutions maintain multiple copies of the video content with different bit rates, which brings huge burden of storage on the server. Regarding rate adaptation controlling techniques, TCP friendly rate control methods for streaming services over mobile networks are proposed [28], [29], where TCP throughput of a flow is predicted as a function of packet loss rate, round trip time, and packet size. Considering the estimated throughput, the bit rate of the streaming traffic can be adjusted. A rate adaptation algorithm for conversational 3G video streaming is introduced by [30]. Then, a few cross-layer adaptation techniques are discussed [31], [32], which can acquire more accurate information of link quality so that the rate adaptation can be more accurately made. However, the servers have to always control and thus suffer from large workload.

B. Video Streaming algorithm:

Recently the H.264 Video Streaming algorithm (VS) technique has gained a momentum [10]. An adaptive video streaming system based on VS is deployed in [9], which studies the real-time VS decoding and encoding at PC servers. The work in [12] proposes a quality-oriented scalable video delivery using VS, but it is only tested in a simulated LTE Network. Regarding the encoding performance of VS, Cloud Stream mainly proposes to deliver high-quality streaming videos through a cloud-based VS proxy [20], which discovered that the cloud computing can significantly improve the performance of VS coding. The above studies motivate us to use VS for video streaming on top of cloud computing.

C. Mobile Cloud Computing Techniques:

The cloud computing has been well positioned to provide video streaming services, especially in the wired Internet because of its scalability and capability [13]. For example, the quality-assured bandwidth auto-scaling for VoD streaming based on the cloud computing is proposed [14], and the CALMS framework [33] is a cloud-assisted live media streaming service for globally distributed users. However, extending the cloud computing-based services to mobile environments requires more factors to consider: wireless link dynamics, user mobility, the limited capability of mobile devices [34], [35]. More recently, new designs for users on top of mobile cloud computing environments are proposed, which virtualizes private agents that are in charge of satisfying the requirements (e.g., QoS) of individual users such as Cloudlets [21] and Stratus [22]. Thus, we are motivated to design the AMES-Cloud framework by using virtual a gents in the cloud to provide adaptive video streaming services.

D. Heterogeneous wireless networks:

Wi-Fi: Wi-Fi is abbreviation of Wireless Fidelity is a set standard in the wireless domain based on IEEE 802.11 specifications. In simple terms it allows a user to connect to the local area network and also have access to the internet without any wires connected to the computer. It is faster than a typical cable modem connection hence giving all the flexibility that the user needs in terms of connectivity and bandwidth [15].

WiMAX: WIMAX is Worldwide Interoperability for Microwave Access. WIMAX is an emerging technology fulfilling QoS requirements of the customers. WiMAX signals have the property to adopt the atmospheric conditions everywhere. WIMAX electromagnetic waves also offer the support of adoptive coding and different operation modes, so voice and data services can easily be transported by WIMAX network platform [16].

HSPDA: High Speed Downlink Packet Access is introduced in (3G) wireless network obtain high speed data rates. HSDPA is a modified interface version of UMTS in 3GPP. It provides not only down link packet access but also it can be used for uplink data up to 14 Mbps per user.

UMTS: Universal Mobile Telecommunication System is a third generation (3G) mobile communication system that provides a range of broadband wireless and mobile communication services. UMTS's target is to build an all-IP network by extending the 2G GSM/GPRS system and using complex technologies including Code DivisionMultiple Access(CDMA), Asynchronous Transfer Mode (ATM), and Internet Protocol (IP) [17].

SAN: SAN is Satellite Area Network. Satellite is a communication device used for high scale broadcast and monitoring purpose that may be stationary or revolving in an orbit. Modern satellite systems use advanced technology to provide broadband data service to areas unserved or underserved by other telecommunications systems. Satellite systems provide rapid setup/teardown of end-systems in field-deployable systems, and have an easily-satisfied requirement for high-elevation line-of-sight communication.

III. VIDEO STREAMING

In streaming procedure, it clip data file is sent to the end individual in a (more or less) continuous flow. It is simply a strategy for shifting information such that it can be prepared as a stable and ongoing flow and it is known as Streaming or encoded movie that is sent across information system is known as Streaming. Streaming movie is a series of "moving images" that are sent in compacted form over the Internet and shown by the audience as they appear. A end user never hang on to obtain a large data file before viewing it clip or enjoying the sound.

A. Streaming Principle

Real-time video applications require media packets to arrive in a timely manner; excessively delayed packets are useless and are treated as lost . In streaming programs it is necessary for the information packets to reach their location in regular basis because the wait can cause the network blockage, and can result in the decrease in all those packets suffering from extreme wait. This causes decrease in quality of information, the synchronization between customer and hosting server to be damaged and mistakes to distribute in the provided movie. Two kinds of steaming are, real-time and pre-recorded streaming. User Datagram Protocol (UDP) is used for streaming which delivers the multi-media flow as a sequence of small packets [4]. The majority of transport protocols perform over an RTP stack, which is implemented on top of UDP/IP to provide an end-to-end network transport for video streaming.

B. Video Streaming Architecture

A cloud based mobile movie streaming scheme is represented in Figure 2. A cloud based source implements a streaming hosting server which is responsible for retrieving, sending and adapting it clip flow. Depending on the application, it clip may be protected on-line for a realtime broadcasting or pre -encoded and stored for broadcasting an on demand.

Programs such as interactive movie, live broadcast, mobile movie streaming or interactive online games require real -time encoding. However, applications such as movie on-demand require pre-encoded movie. When the multicast session is initialized, the streaming hosting server retrieves the compressed movie and begins the loading with the adequate bitrate stream.

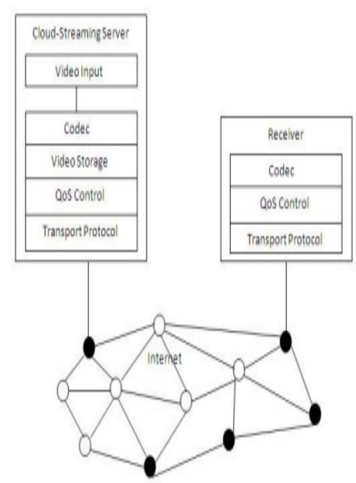

Fig.2: Video Streaming Architecture

C. Matching Between Bandwidth Prediction and VS Segments:

After obtaining the predicted bandwidth, or say good put, of next time window, subVC will match and decide how many video segments of BL and ELs can be transmitted approximately. We hereby define the term "resolution" to indicate the level of temporal segmentation and the number of ELs. If *T*win is small and there are more ELs, we say the VS-based video
source is with a higher resolution. We illustrate two cases of coarse-grained (low resolution) and a relatively fine-grained (high resolution) for matching between the VS segments and the predicted goodput . The resolution with two ELs and a larger *T*win can

bandwidth. However a higher resolution also induces more encoding workload to the servers. hardly fit to the signal fluctuation, and thus there are some bandwidth wasted or packets lost. In contrast a higher resolution with more ELs and a smaller *T*win can always fit the fluctuation of the

Suppose there are totally ELs, and the bit rate of the *j*th EL is denoted as *RELj* while the bit rate of the BL is).We let *ELi* indicate the VS segment of the BL with temporal sequence , and let indicate the VS segment of the *j*th EL with temporal sequence . So the algorithm of matching between predicted bandwidth and VS segments is shown in Algorithm as following:

Algorithm 1 Matching Algorithm between BW and Segments

$i = 0$
$BW_0 = R_{BL}$
Transmit BL_0
Monitor $BW_0^{practical}$
repeat
  Sleep for T_{win}
  Obtain p_i, RTT_i, $SINR_i$ etc., from client's report
  Predict $BW_{i+1}^{estimate}$ (or $BW_{i+1}^{estimate} = BW_i^{practical}$)
  $k = 0$
  $BW_{EL} = 0$
  repeat
    $k++$
    if $k >= j$ break
    $BW_{EL} = BW_{EL} + R_{EL^k}$
  until $BW_{EL} >= BW_{i+1}^{estimate} - R_{BL}$
  Transmit BL_{i+1} and $EL_{i+1}^1, EL_{i+1}^2, \ldots, EL_{i+1}^{k-1}$
  Monitor $BW_{i+1}^{practical}$
  $i++$
until All video segments are transmitted

IV. IMPLEMENTATION AND RESULTS

There are two major operations in Real-time video streaming client video request and network selection . The process starts with the video request to the cloud database and then the search is being made for the request, if the requested video is found then it streamed backed to the user this streaming is done by using adaptive video streaming algorithm as the result it uses the H.264 encoding technique to stream the video to the user end mobile device.

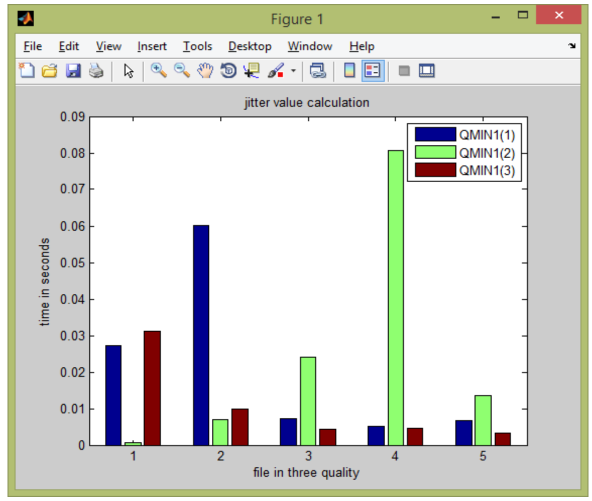

Fig 6. Jitter and loss calculation graph

V. CONCLUSION

In this paper, we discussed our proposal of an adaptive mobile video streaming and sharing framework, which efficiently stores videos in the clouds (VCS), and utilizes cloud computing to construct private agent (subVC) for each mobile user to try to offer "non-terminating" video streaming adapting to the fluctuation of link quality based on the Video Streaming algorithm technique. And also calculating the bandwidth estimation and the jitter value calculation for real-time video streaming over cloud was implemented successfully. From this when moving over one network to another network it would be easy for the end user to check the quality of the video and delay during the video playing can be easily observed and future work is to all more parameters to improve the video quality.

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