ISSN: 2321-7782 (Online)
Volume 2, Issue 2, February 2014
International Journal of Advance Research in Computer Science and Management Studies
Research Article / Paper / Case Study
Available online at: www.ijarcsms.com

Facilitating Minimized Buffer time and live video streaming from cloud for mobile networks

G.R. Ashok Kumar[1]
M.E.
Computer Science and Engineering
Francis Xavier Engineering College
Tirunelveli, Tamilnadu – India

V. Perathu selvi[2]
Asst. Professor
Computer Science and Engineering
Francis Xavier Engineering College
Tirunelveli, Tamilnadu – India

Abstract: Cloud computing rely on sharing of resources and improve performance, similar to a utility over a network. The video Sharing in the clouds through keeps travelling in faster rate, the locks in the attention through the traffic and security also travels in the same track. The demand in the link capacity according to the traffic has to be delivered. This package comes with the sharing, streaming concepts and security issues. Here we monitor share the high resolution video considering the 720, 1080, which has the large packet size. The virtual channels are laid by RTMPS and they take them and deliver to the appropriate clients. The TCP is a subordinate for the RTMPS since it's the wireless networks. The security is the major concern in the VSP in the cloud environment. They deliver and secure the video to the client with the higher delivery rate and performance.

Keywords: Link Capacity, Live streaming, Scalable Video Coding, Video streaming, uninterrupted streaming, RTMPS.

I. INTRODUCTION

While streaming the video in the mobile devices it is hectic to the user that is accounted as the traffic criteria. They evolve in the cloud environment while the user tries to stream and buffer the video with the traffic problems. The traffic is accounted on the one end and also the loss of packets and also the clarity is also accounted on another end. The user really faces problems with the mobile and mobile devices which he consider to be so compact. Though they are compact they do not deliver the user with the desired requirements. When we consider the wired networks they are much compatible with two factors they are the compatibility and the scalability factors. Though the mobile user are provided with the 3g and LTE connection in the fast access and transmission rate they are not in a mode to provide the user with the sufficient quality in the video in which the user expects. The main reason is the traffic and the traffic is concerned in this paper with the new way for the uplink and downlink which means with the upload and download. In this area the video is presented in the cloud environment which is a part advantage to the user still the traffic is a prone**.** While receiving video via the fast transmission channels that is by 3g networks the user takes long time buffering in the video. At times the grey scaled video that is the very poor quality of video is also been delivered which makes the user very unlucky. Here we have an algorithm namely Adaptive Mobile video streaming which automatically adjusts the video and another algorithm namely Efficient social video sharing which prefetches the interacts among the connected mobile users. The combined theory is used here to have to provide the user with the high quality video and deliver with the proper resolution capabilities. Further to strengthen the concept of congestion control we also provide with the shortest path tracker and provide them to travel and choose alternative path in terms of traffic. The security behaviors are been invoked to reasonably provide and deliver the user with the quality and to the server with the authenticated user.

II. PROBLEM STATEMENT

In MS2, a real time multisource mobile streaming systems [3] we come across the problem of handoff management mechanism which is to predict the handoff occurrence time with high accuracy level. This mainly aims to provide the seamless and smooth playback of video for roaming users. In Dynamic QoS negotiation for real time systems in video streaming [6] users dynamically negotiate Qos profiles with different networks. They propose a bandwidth aggregation which helps users to negotiate their desired service levels and reach them by more interfaces; also enhanced EDPF scheduling algorithms were used in these Qos negotiation systems. To provide the user with the desired video we also have to scale the video so adopt an techniques know a SVC, a standard namely H264/ AVC video encoding standard has been demonstrated [4] the SVC enables transmission and decoding of partial bit stream to provide video services with lower temporal or spatial resolutions. In which the SVC provides a graceful degradation in lossy transmission and power adoption. In H.264/MPEG-4 advanced video coding standard we enable encoding of high quality video bit stream that continues 1 or more subset bit streams that can themselves be decoded with complexity and reconstruction quality similar to that achieved using the H.264/AVC [7].

III. METHODOLOGIES IN LIVE STREAMING

In the traditional system we come across two types of algorithm based on the streaming facilities and also on the bandwidth capabilities. The cloud computing makes us comfortable with the wired networks while considering factors such as scalability and capability. Since we are going to the wireless devices there prevails the link capacity problem as per the traffic demand. When would like to switch towards the mobile networks from that of wired networks we have to face certain additional parameters such as Mobility, Peak usage, lack of proactive monitoring. This is the traditional to that of the wired network devices and the new capabilities have to be adopted and here the new technique makes an impact in the wireless mobile cloud environment. The proposed system makes the design and overlays the bandwidth and prefetch the video exactly and deliver to the user. The prefetching video is been segmented by temporal segmentation technique. To maintain the video quality according to the user's mobile devices some techniques have to be maintained to deliver the proper video with reliable quality to the video to the user. The Social Video Coding is a technique that is been embedded in the process to maintain and scale the video accordingly. The prefetching delay is been observed when streaming a video by the user. The watching delay is also been monitored, that is the user has to wait once the video is been clicked. Since a single storage of video location is maintained there is a delay in watching the video. The packet loss is been still our concern while sharing the video to various mobile users.

IV. ADAPTIVE TECHNIQUE IN THE STREAMING OF LIVE VIDEOS

The techniques that come under the adaptive technique of the live video streaming are

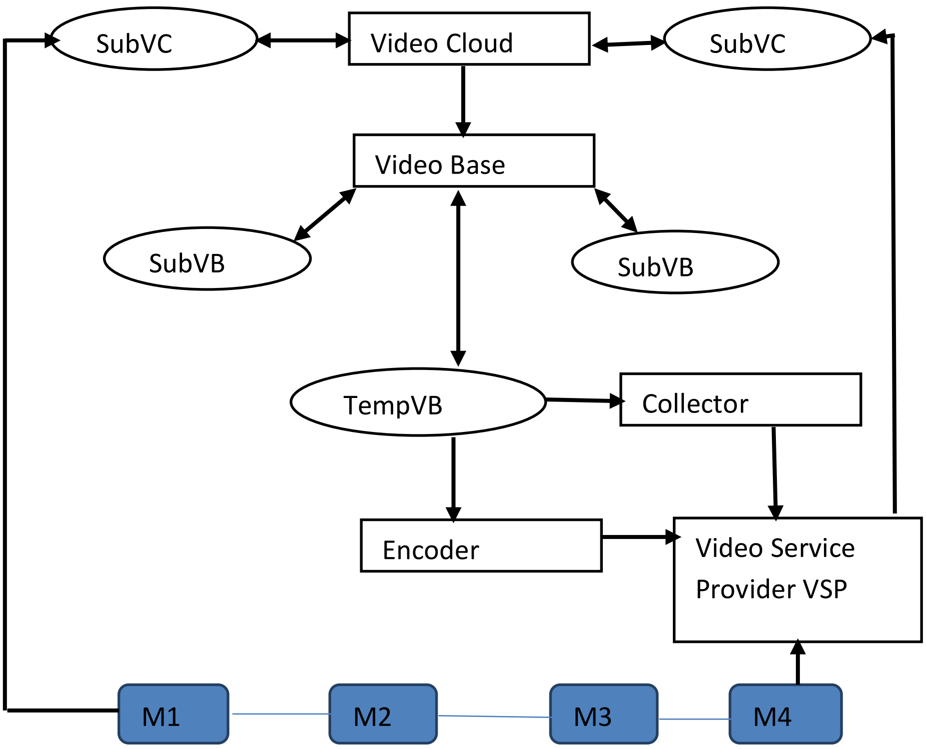
A. Adaptive Streaming Technique

In Adaptive, we use SVC which is bandwidth improved at server side with adapting rate when SVC video streaming is controlled. The Connection quality watch the mobile client which monitors the strength of the signal and packet loss with regular period of time. The video is then been split by temporal segmentation by interval (Twin) also sub VC performs calculation and predict potential bandwidth in next time window. The video cloud calculates and predicts bandwidth is another Twin. Once the bandwidth is predicted subVC decide how many BL and EL are able to transmit. The final process is matching between bandwidth prediction and SVC segments. After the above two process subVC will match and decide how many video segments of BL and EL can be transmitted. When the Twin is predicted to be small then there will be more Enhanced Layers (EL). Since Twin is small and there are more EL, SVC based video source is with high resolution

B. Improved Social Sharing

This algorithm come across the technique known as social content sharing in which a particular video shared can be streamed and watched by another user at the another window. The VC is the main storage location like the hard disk and they have the total videos that are been available. After video sharing, there will be delay that the receipt gets to know the sharing. Once the video has to be played they are prefetched in the VB. Also the rate of prefetching segments has to be calculated as per the size of the video that is requested by the user.

V. Architecture Design



VI. Security Mechanisms

The Encryption scheme is been implemented in this cloud framework. The encryption scheme is carried out in the streamed content of the video and the transmission is been done later to the various clients. Here we use a protocol known as the basic protocol initially known as the TCP which is derived from the RTMPE which is abbreviated as Real Time messaging protocol. This particular algorithm slits the packets of video into 128 bytes. The encryption is been carried out in the VSP to deliver only the video to the particular video client that is registered. Once the video are encrypted they are stored initially in VSP and then they are transmitted. As the RTM has several virtual channels they makes the packet transmission easier with no delay in the packets. During the higher transmission rate the RTMP combines the MP3 or AAC audio multimedia streams and make RPC using the Action Message format.

A. Handshake

After the connection is established between the client and the server, we perform a handshake with a small number of video packets to verify the successful transmission of the packets. There is a proper response such as acknowledgment from the source and destination with SO-1 and RO-1.This conforms the proper delivery of packets in the client side.

VII. Performance Analysis

In the Performance the comparison is made between the live streaming and the without the live streaming and the graph is been simulated for the channel usage. As the Channel rate decreases there is an effective transmission in the video packets.

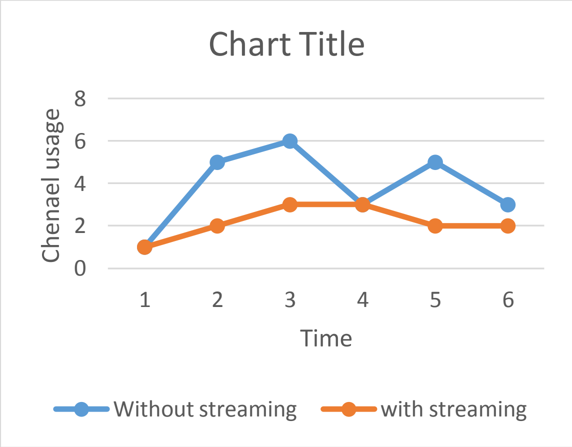


Fig: 1 Comparison in streaming videos

This Performance shows the channel usage as for that of the live and non-live streaming. Since the live streaming comes with the less number of usages of channels they travel in a better and produce optimum output.

The live and non-live video streaming is been compared based on the channel utilisation. The live streaming deliver with the high quality and with the less number of node are been chosen to reach the destination.

VIII. Conclusion

In this paper link capacity according to the traffic demand has to be delivered. The techniques used in this paper do the exact prefetching of the video. Once the prefetching job is done the delay in the video is been minimized. The watching delay in the cloud environment is also been minimized. The routing technique and the path tracing concept is viewed and maximized in this paper. The SVC technique does the good delivery of the video to the user. Further new techniques have to be invoked and the replacement of the SVC is been concentrated. While the higher networks are taken in to concern the delay is been completely minimized. The future work comes with the SWF verification in which allows only the authorized entry in the clod environment. Which eliminates the anomaly entering the cloud environment? The encoding and the decoding of the video packets after the segmentation is been done. This eliminates the unauthorized access of the video and the entering in to the cloud environment too.

Acknowledgement

I thank the Lord Almighty who has been with me through every walk in my life, for guiding me and for the blessings showered on me to complete the project in successful manner. I convey my sincere thanks to Dr .R. Ravi M.E., Ph.D., Professor and Head, Department of Computer Science and Engineering, Francis Xavier Engineering College who inspired me and supported me throughout. I also thank my guide Ms.V.Perathu selvi M.E., Assistant Professor, Computer Science and Engineering for his valuable guidance throughout and it is great privilege to express my gratitude to her.

References

1. Xiaofei Wang, Min Chen, Ted Taekyoung Kwon, LaurenceT. Yang. and Victor C. and M. Leung ‘ AMES-Cloud : A Framework of Adaptive Mobile Video Streaming and Efficient Social Video Sharing in the Clouds’, IEEE Transaction on Multimedia, vol.15, No.4.
2. Ashish Khisti., Zhenzhong Chen., Yoggang Wen. and Weiwen zhang. (2013) ‘QoE Driven Cache management for HTTP adaptive bit rate streaming over wireless networks’, IEEE Trans. Multimedia, vol. 15, No. 6.
3. Tarik Taleb. and Hashimoto K. (2011) ‘MS2: A new real time Multisource Mobile Streaming Architecture’, IEEE Trans. Broadcasting, vol. 57, No. 3, pp. 662-673.
4. Chin- Feng Lai., Hanggang Wang., Han-Chieh Chao. and Guofang Nan. (2013) ‘A Network and device aware QoS approach for cloud-based Mobile streaming’, IEEE Trans. Multimedia, vol. 15, no. 4, pp. 747-757.
5. Esa Piri., Mikko Vitto., Janne Vehkapera. and Tiia Sutinen. (2010) ‘Dynamic cross-layer adoption of Scalable video in Wireless Networking’, IEEE GLOBECOM, pp. 1-5.

6. Heiko Schwarz., Detlev Marpe. and Thomas Weigand. (2007) 'Overview of scalable video coding extension of the H.264/AVC standard ', IEEE Trans. Circuits and systems for Video technology, vol. 17, No. 9, pp. 1103-1120.
7. Heiko Schwarz. and Mathias Wien. (2008) 'Scalable video coding extension of the H.264/AVC standard', IEEE Signal Process Mag, vol. 25, No. 2, pp. 135-141.

AUTHOR(S) PROFILE

Ashok Kumar, received the B.E. Degree in Computer Science and engineering from Scad College of engineering and technology, Anna University, Tirunelveli in 2011.

Currently pursuing M.E. Computer Science and Engineering in Francis Xavier Engineering College, Anna University Chennai.

Perathu selvi, received her M.E Degree from Francis Xavier engineering college, Anna University, 2012.

Currently she is working as an Asst. Professor in Francis Xavier Engineering College.