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Adaptive Video Streaming over HTTP Through Wireless High Speed Network

M. Tech. Scholar Monu Kumar Saini, Assistant Professor Mr. Rahul Pandey
Department of Electronics and Communication,
SORT, People's University,
Bhopal, India.

Abstract- The adaptive bitrate streaming works on the principle of rapidly changing bitrate based on available resources, which is usually a bandwidth, resource limit and user query. The selection of bitrates is controlled usually by client and not by server. The adaptive HTTP video streaming is similar to progressive download streaming. In adaptive HTTP video streaming, the multimedia segments are split into same length segments. On other hand, the segments are split into video GoP that initiates with key frame. The initiation of key frame in video segments represents the past or future dependencies. Finally, the entire video segments are encoded and it is then hosted over a HTTP streaming server. Recently, the online video streaming traffic is growing massively. It is estimated that traffic in online video streaming services accounts for more than 80% of traffic in internet by 2020. Hence, the design of next generation video streaming services is driven by internet traffic.

Keywords- HTTP, Streaming Server, Video streaming.

I. INTRODUCTION

The developments in real time video processing and streaming technology has facilitated quality streaming of video over heterogeneous networks. Compression techniques have helped in transferring large chunk of data during streaming through limited bandwidth of network. Both wired and wireless networks, act as the platform for video streaming. The unpredictable nature of network bandwidth, the occurrence of errors and the heterogeneity of consumer terminals makes video streaming a highly challenging task.There is a trade-off between the original video quality, the bit rate, and the video quality received by theuser.

The traditional streaming method based on progressive download fails to cope up with dynamic network traffic, thereby degrading media quality. The ultimate objective of all streaming services is to deliver seamless content to the end user in real-time, though it poses a huge challenge due to large fluctuating bandwidth conditions. The media content in any specific video needs to be adaptive to match the available bitrate in the network and also to provide the user the seamless multimedia service with the maximum achievable Quality of Experience (QoE) (Ali El Essaili et al. 2015).

Streaming video is content sent in compressed form over the Internet and displayed by the viewer in real time. With streaming video aweb user does not have to wait to download a file to play it. Instead, the media is sent in a continuous stream of data and is played as it arrives. Video streaming is a very important application in multimedia. "Stream" refers to the process of delivering media in this manner. Streaming video is usually sent from stored video files, but can be streamed as part of a live "feed." In a live stream, the video signal is converted into a compressed digital signal and transmitted.

There exist a very wide variety of video streaming applications, which have very different conditions or attributes. The video channels for communication may also be static or dynamic, circuit switched or packet switched, may support a variable or constant bit rate transmission, and may support some form of Quality of Service (QoS) or may provide only best effort support. These properties of a video transmission network strongly influence the design and implementation of the system (MshariNajeeb et.al 2013).

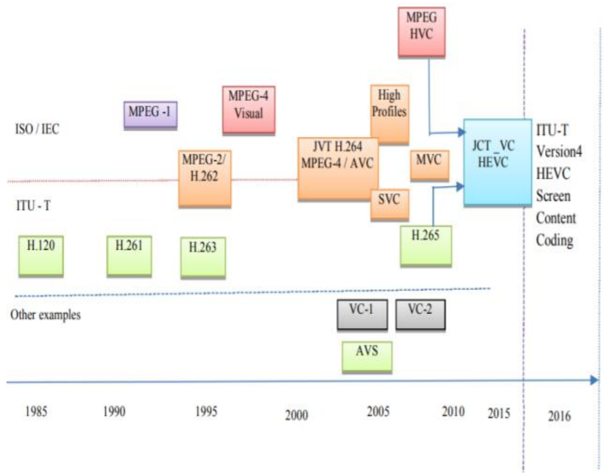

Fig 1. Evolution standard of video codec (Nam Ling, 2011).

The conventional data mining methods to analyze the video for knowledge discovery becomes unsuitable here because the system requirement is not only highly predictable but also strictly defined by the performance parameters. Hence it becomes mandatory to develop dynamic adaptation techniques that consider real time requirements of the video streamingsystems.

Table 1. Video Coding Standards and Purpose (Todd kellison, 2015).

S. No.	Coding Standards	Developed by	Developed for
1.	H.261	VCEG	Transmission over ISDN lines
2.	MPEG-1	MPEG	Digital CD's and tape players, etc.,
3.	H.262 / MPEG-1	MPEG	HDTV broadcasting, DVD players, support for interlaced video
4.	H.263	VCEG / MPEG	Video conferencing and video telephony
5.	MPEG-4 Part 2	MPEG	Surveillance cameras, HDTV broadcasting, Mobile phones
6.	H.264 / MPEG-4 Part 10	VCEG / MPEG	Blue-ray disc players, Streaming Internet, HDTV broadcasting
7.	HEVC / H.265	VCEG / MPEG	Supports up to 8K UHD

II. CHALLENGES IN VIDEO STREAMING

Video streaming is the key components in delivering multimedia content to different users in real time. Streaming can be more complex in a packet based network because they have strong and specific mechanism to regulate data traffic. The video data flows can be formatted, denoting that the latency between consecutive packets can be similar, and the data bit rate has to match the available capacity of link. The constant bit rate of video flow may be desirable to feed the decoder at receiver to play the video without any interruptions.

In the real time video transfer over packet switching network the packet is transported over unreliable channel where transmission delay is affected by the variation in throughput and loss of packet in the shared network. One of the main problems in real time transmission of video over public networks like Internet and mobile wireless system is the mismatch between the current status of network and the QoS requirements (throughput, delay, jitter and loss of packet) of the different media applications.

The existing TCP/IP based internet works on best effort service model which is suitable for non-real time applications like web browsing, download etc and has no provision to accommodate the QoS requirements of multimedia services. Likewise, the current wireless networks including wireless local area network (WLAN) and cellular mobile network (3G / 4G) is designed and optimized for telephony applications which works on narrow band channel and lacks support for wide band streaming services.

In internet different web based applications have diverse category of impairments in quality although the network remains same which necessitate the development of mechanism to support heterogeneous QoS (Yunmin Go et.al.2015).

III. ITU-T H.264 CODING / DECODING

The ITU-T H.264 recommendation like other video coding standard defines syntax and semantic of the encoded video and a scheme for decoding a bit stream to display the contents.

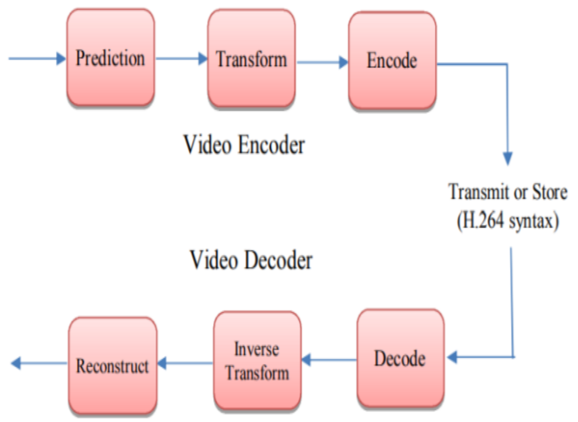

Fig 2. H.264 (vcodex 2013).

The standard does not impose any restriction on methods to encode video and hence there is scope for improvement in the matter of achieving higher compression ratio. Figure 1.2 shows a generalized encoding and decoding procedure and indicates the major components in the ITU-T H.264 standard.

The ITU-T H.264 standard was targeted towards streaming video / audio through the internet. In other words, it supports many encoding / decoding methods for efficient content transfer over the internet. The ITU-T H.264 is one of the best video codec for providing the support for streaming on the top of Hyper Text Transfer Protocol (HTTP).

IV. LITERATURE REVIEW

There is a need to change the rate of content of multimedia codec such as MPEG-2, MPEG-4 / AVC, etc. for minimizing congestion in the network which is difficult to implement (Nuruddeen et al. 2013). One suggested method is to pause the ongoing encoding process and

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change the configuration of the encoder and then restart it. The main problem is that even multiple coders working parallel cannot easily switch between different configurations arbitrarily. So a better approach could be the sender discarding the few select frames; this can help not only to avoid the video stall, but also cause reduction in the traffic rate. These are the major issues involved in developing an efficient system to control congestion. This also opens up avenues for rate adaptive congestion control framework employing a scalable video coding.

Thomas Stock Hammer et al. (2004) has considered streaming of video sequences over both constant and variable bit rate channels. The objective is to enable decoding of each video unit before it exceeds display deadline and also to guarantee successful sequence presentation even if the media rate does not match with the channel rate. The authors have specified the minimum initial delay and the minimum required buffer for a given video stream and deterministic VBR channel.

This has enabled observation of the random behaviour of the channel bit rate. They have found that if the curve of video stream as well as receiver curve is prior to the server, there exists a minimum value for the initial play out delay and decoder size ensures a successful play out.

As a result of the complex nature of input video, the (unconstrained) coded video bit-streams tend to be highly bursty resulting in large variation of bit rate. A constant quantization parameter throughout the session helps to providing uniform quality but still can cause vivid variations in bit rate.

The constant bit rate method of video coding allows easy management of resources but, in many cases degrades the quality of video. Variable bit rate coding could be a better solution (Tea Anselmo& Daniele Alfonso 2010). However, the variable bit rate coding of video cause large variation in output data rate. Very often it creates a mismatch between throughput and bit rate of video which is a major challenge in adaptive video streaming (Tuan Vu et al. 2015).

In order to understand different existing streaming services and thus provide direction on designing resource-efficient and high quality streaming media systems, collection of different streaming media workload from a broadband home user and business user served by internet service provider can help analysis of the commonly utilised switching methods.

The measurement and analysis results by **Lei Guo et al. (2006)** show many of the currently used techniques for streaming tending to over utilize Central Processing Unit (CPU) and network bandwidth to support a better service to the subscriber. It may not be either a desirable or an effective solution for improving the streaming quality. The author further notes that the technique of fast streaming does not suitably work with rate adaptation which results in the worst user experience than the traditional TCP based approach.

The Multiprotocol Label Switching (MPLS) technique is used by many service providers in which packets are switched based on level. Many believed that integration of MPLS in IP ensures improving the switching speed in the network. However, implementing it in the core network remains an issue.

Mihai Chiroiu et al. (2016) have analyzed the different factors that influence the quality of video traffic in Internet Protocol Television (IPTV) through MPLS system. Although the MPLS system is designed to avoid congestion, the failure of links and components may result in redirection of traffic on some links. Any proposed solution has to consider the main QoS parameters namely, throughput, delay, delay variation and packet loss.

The internet offers radio, TV, home cinema, and share of video for commercial and personal use. YouTube, Daily motion, Meta cafe, etc., are among many video sharing services used as the social networks and content sharing in Web 2.0.

Mohit Saxena et al. (2008) has measured the different characteristics of video traffic in two phases. In the first phase, they have measured the latency in serving video streams, and in the second phase, they have gathered the metadata of video streams along with the location information of the content servers which store the file. Further, they have explored the happenings in the background, i.e., the characteristics of content delivery networks along with their locations, and also the action relative to the content which is downloaded.

The work by **Delia Ciullo et al. (2010)** shows the large scale experiments on comparing the most successful P2P-TV systems: PPLive, SopCast and TVAnts. The main objective is to assess the network awareness in the application, and so, the proposed methodology highlights the metric which is needed for exploiting in P2P-TV applications to optimize video delivery. A general framework was defined to quantify the specific network parameters that have control over application choices i.e., the parameters that mainly drive the peer selection and data exchange.

The three different applications were experimented for an hour by making the peers watching the same channel simultaneously. Packet-level traces were collected and analysed. The nominal video rate was 384 kbps, the video quality perceived by users were similar across systems. The framework used here was mainly to assess whether peers fairly respect each other. The methodology applied on a large dataset resulted TVAnts and PPLive showing a

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mild preference to exchange data among peers in same autonomous system and this clustering effect seemed less intense in SopCast.

V. METHODOLOGY

In streaming media applications the user has no need wait to download the entire content but can rather start playing the audio / video content. Since the internet works on best effort service model and streaming application requires delivery of packets in real time, it poses many challenges in system design. If the network bandwidth fluctuates and does not guarantee minimum throughput as per the requirement of quality of service, it may degrade the quality of experience, thereby necessitating the mechanism to support adaptive streaming.

1. Traditional Approach:
Traditionally the User Datagram Protocol (UDP) is used as transport protocol to stream media content because of its simplicity and efficiency of data transfer. However, the UDP protocol lacks reliability i.e., the datagram may be delivered out of order with no trace of lost datagram as it does not support mechanism for acknowledgement. It is left to the application program at the receiver to handle the problem of the out of order and the lost packets (Kcchao 2010). The Real Time Streaming Protocol (RTSP) is used for controlling streaming media at server through a client request. The client request consists of actions like play, pause, fast forward, fast reverse, etc., to facilitate control on streaming contents as per viewing requirements.

The RTSP helps in establishing media session while controlling the stream between the client and the server. The transfer of streaming media content is not directly carried out by the RTSP. Rather it uses the Real-time Transport Protocol (RTP) along with the Real-time Control Protocol (RTCP) for content delivery. The RTP and RTCP are designed and developed to work on the top of the UDP (Figure 3)

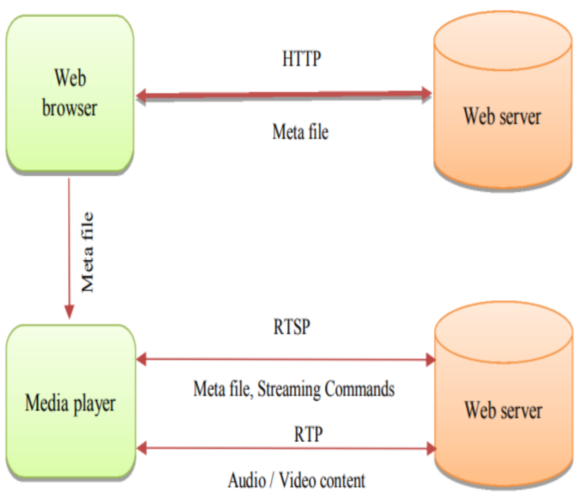

Fig 3. Traditional Approaches: Multimedia over IP.

VI. CONCLUSION

In recent years, the DASH approach is accepted as an important model for video streaming, since it provides simplicity, improved scalability, reduction in the wastage of bandwidth and ability access through the firewalls and NATs. However, the main problem associated with DASH system is its poor adaptation capability over client. It provides poor streaming of videos and that tends to be unmanaged and heterogeneous in radio access networks.

Hence the utilization of adaptation algorithm in proposed work is to increase the maximum utilization of network while maintaining high quality video streaming. The algorithm intends to maintain the trade-off between the network utilization and video quality, which are not easy to achieve. Hence, the design of effective adaptation algorithm in video streaming networks using DASH system is considered as a bottleneck, which needs to be addressed.

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