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Network Performance Analysis of Cloud Based Multimedia Streaming Service

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ABSTRACT

Multimedia streaming is the evergreen technology brings quality on-demand and live media content to the end consumer. Streaming enables the users to view real time multimedia content on-the-go at anywhere, any time. The quality of service is a major concern in the increasing network traffic and high demand in media bit rate streaming. An adaptive streaming technique provides innovation compare to the conventional streaming models by dynamically selecting the media bit rate based on the network variation and processing capability to ensure the best quality of service. However traditional environment streaming service suffers with network delay, low quality performance and high cost. With the aid of Cloud, streaming delivers the content at the edges with very minimal delays on pay-for-use model. In this paper, we focus on the comprehensive analysis of various cloud based streaming methods with underlying architecture and compare the performance of cloud based streaming with conventional streaming model. The experiment results shows that cloud based streaming delivery outperforms in terms of throughput, packet delivery and effective use of the network bandwidth under similar network conditions.

KEYWORDS

Adaptive streaming, Cloud streaming, conventional, quality of service, rate control

1 INTRODUCTION

The growth of network communication technologies transformed the world life style. The content providers, network operator's offers enormous number of services for day-to-day enterprise business and people personal needs. With the invent of internet, it builds huge potential market for transferring audio, video contents to consumers instead of relying on storage devices for delivering or downloading large files over the network. The end user is allowed to choose appropriate communication and content delivery methods based on the need and usage. The multimedia applications such as playback, streaming and video telephony services are becomes a part and parcel of everyone's life.

Today the usage of Cloud service is inevitable and it dramatically changes the user's service need and delivery models. It facilitates the clients to access cloud services anywhere, anytime without any physical resource limitation. Multimedia as a service (MaaS) is a key offering in the cloud platform, which hosts and manages many multimedia applications over the cloud. Multimedia streaming requires real time content processing and it has to ensure the susceptible Quality of Service (QoS) in the receiving end. Traditional multimedia services face many technical and infrastructure challenges to provide best QoS under constrained network environment. In the past, there are some difficulties for users to get streaming media services due to jitter, reliability, hardware/software processing capabilities and so on. Cloud assisted streaming solves the traditional streaming problems and gives state of art performance.

Cloud platform provides sophisticated hardware and software, that help capturing, processing, buffering the media data in real time without any delay. Cloud connects many encoding/transcoding engines in a network. The load balancing mechanism in the cloud helps to pre-process/encode the content quickly in multiple formats. The multiple user service support enables users to either access the streaming service on the platform, or distribute streaming service through the platform. Cloud supports two ways to transfer media from server to client. Serial fragments fetching method allows the client to open only single connection and fetch the fragments are requested one after another. Parallel fetching mainly used in cloud, allows client to open multiple connection and fetch the fragments are requested and received in parallel. Cloud also maintains end to end service management, content Digital right management and account management as service.

The main advantages of using streaming as a service are

- Excellent end user experience through edge server delivery
- Time to market and cost optimization
- Different streaming format support and availability
- Global presence and availability using cloud storage
- Control over streaming delivery and impose secure streaming
- No complex hardware and software required in the client.

The main contributions of this paper include the following two aspects.

— The underlying architecture of Cloud based streaming service architecture with example of few currently existing cloud streaming services are discussed.

— Simple experimental setup is formed to evaluate the streaming metrics (i.e., Packet loss, start-up delay, buffer overflow/underflow, throughput, packet arrival delay etc…) and analyzed the end results. The results show that which streaming model provides good performance considering many parameters under the available network bandwidth. The Cloud assisted streaming delivers throughput almost twice compare to the conventional methods with moderate start-up delay.

The rest of the paper is organized as follows. Section 2 presents the related work to describe the various streaming architecture and delivery methods. Section 3 gives a brief review of multimedia streaming and Cloud based streaming delivery architecture and few available streaming services in the market. The experimental setup and results of various the streaming technique performance metrics are provided in Section 4 and 5. Finally Section 6 concludes the paper.

2 RELATED WORK

RTP is Real-Time Transport Protocol [1] is an application layer protocol, uses lower level protocols such as UDP and TCP for transport across the network. RTP provides a flexible framework for delivery of real-time media, such as audio and video, over IP networks. Server controls the flow and rate of data transfer. Typically the server sends the data at the bitrate of the media stream. RTCP is the Real Time Control Protocol [2] designed to work alongside the real-time transport protocol (RTP) to provide feedback for flow control, manages several aspects of the delivery of real-time content. Real Time Streaming Protocol (RTSP) [3] is stateless protocol and responsible for session initiation and control the entire streaming session. Progressive download (PD) [4] uses HTTP protocol for content streaming where media playback is typically started before receiving the entire stream. HTTP protocol runs on top of TCP/IP, data rate adaptation is controlled at the client side. TCP has its own congestion and flow control mechanism. HTTP server sends data to the client at the available bandwidth. Real network [5] and Microsoft [6] supports HTTP progressive download streaming. Real-Time Messaging Protocol (RTMP) [7] is proprietary Adobe streaming protocol used for streaming over TCP. It manages Audio, video and data for live and on-demand streaming between a Flash player and a Flash Media Server

Adaptive streaming is a hybrid streaming delivery method using HTTP. Since HTTP is widely used proven web protocol, media delivery over HTTP can be easily optimized to deliver real time media and adapted to the network variations. The input media divided into number of small chunks for appropriate duration, encoded in different bit rates, stored and delivered to the client sequentially using HTTP download. The client can opt for different bit rate media chunks from server. Numerous adaptive streaming methods developed by different vendors like Apple HTTP Live Streaming (HLS) [8],Abode HTTP Dynamic streaming (HDS)[9] and Microsoft smooth streaming [10],. Each method uses different file formats and transportation standards defined by the vendors. Dynamic Adaptive Streaming over HTTP (DASH) [11][12], novel common standard in adaptive streaming proposes unique architecture and it is widely used and accepted as adaptive streaming standard by most of the industry leaders. Cloud assisted streaming offers content hosting, processing, storage and delivery on single sign on in the cloud servers. Cloud based streaming eliminates the use of dedicated servers and disk space limitation. The server in the cloud will take care of the media content processing ranging from mixing, trans-coding and unifying streams from various clouds and so on. In the current trend, multiple companies have started providing content management systems and streaming delivery over the cloud like Google Music[13], Amazon Elastic Compute Cloud (EC2)[14], Amazon Simple Storage Service [15], cloud front[16], Apple Cloud service etc.

In Cloud backed streaming, cloud streaming client follows the rate measurement same as like conventional and adaptive streaming depending on the method it streams. The cloud streaming service delivery model and infrastructure is quite different than conventional model and more effective. To best of our knowledge only limited information available for the service model, streaming delivery and performance improvement over cloud. In Paper [17] proposes the cloud-assisted procedure to improve the user's quality of experience with cloud based downloading strategy. It does the buffer management based on the cloud side calculations which increase the significant load at the cloud and it does not dynamically control the rate based on the network and end client parameters.The cloud assisted model for live media streaming is discussed in paper [18]. It provides the optimal solutions to the real time streaming service in cloud servers with diverse capacities.

3 CLOUD BASED STREAMING SERVICE

Streaming engages the significant space with the growth of the communication medium due to downloading large multimedia files is time consuming and occupies more system memory in client. Media streaming is the most capable method to pull the video from server to the media player directly. In heterogeneous network architecture, achieving best Quality of Service (QoS) is really a challenge. Growing demand in streaming requires more processing power, massive storage and high speed transport. Most of the video streaming uses User Datagram protocol (UDP) for real time delivery with simplified client and server architecture. However it suffers with server infrastructure, Network Address Translation (NAT) and network firewalls issues. Alternate to UDP, Transmission Control Protocol (TCP) can be used for streaming delivery and overcomes above said issues. TCP uses Hypertext Transfer Protocol (HTTP) as streaming application level delivery protocol in progressive download. But it has limitation in real time stream distribution due to transmission latency and packet retransmission delay.

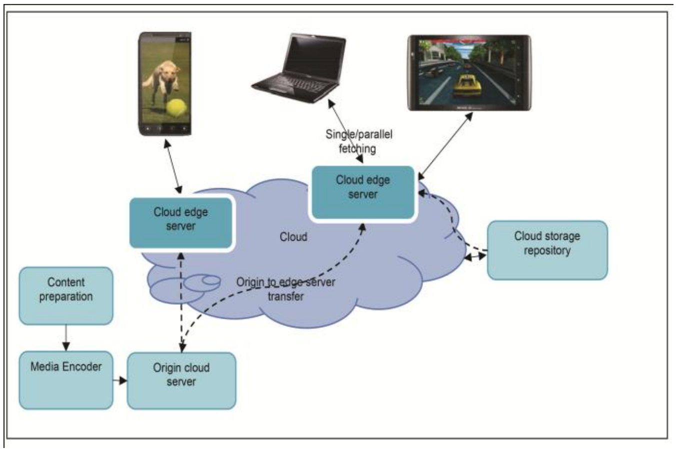

Fig.1. Cloud based media streaming architecture

The adaptive streaming over HTTP is the successor and overcomes all the issues faced by traditional streaming techniques. Adaptive HTTP streaming has multiple bit rate download approach and client driven rate control technique to overcome the network bandwidth variation and to ensure the smooth playback. The cloud is seen the future of the mobile ,connected home and enterprise IT, which replaces the complex processing units and offloads lot of energy from clients. Advent of cloud brings the multimedia content delivery and rendering data with less effort and cost, without compromising in data quality. Cloud assisted streaming in cloud infrastructure provides numerous advantages such as scalable download, anywhere video presence, fail-over alternative, concurrent download session, etc. Cloud based Adaptive streaming service provides content presence on the global edge and state-of-art Quality of service.

In cloud assisted streaming, multimedia files are hosted on the HTTP Web server or streaming server depends on the distribution. When an end user requests a piece of streaming content, the request is directed towards the nearest available edge server. The edge server makes a determination as to whether or not it has the content item. If the edge server does have the content, the server will immediately begin to stream that content to the end user. If not, the edge server will initiate an HTTP byte range request to retrieve the content from the server or Cloud Storage repository. The multimedia file is requested via TCP to ensure a lossless copy of the stream to the edge server from origin and the content is streamed to the end user over UDP or TCP depending on what protocol the end user supports. Most of the cloud streaming services using HTTP or HTTPS protocols, or RTMP protocol for download and streaming. The logical flow of the cloud streaming presented in figure 1

3.1 Cloud Transcoding/Encoding

Cloud encoding/transcoding services are provided by cloud-based companies to which upload source video files and set appropriate parameters to get the output encoded/transcoded media. Amazon Web Services provides cloud encoding service on pay for use model .It offers a variety of services, including scalable computing capacity and online storage capability

3.2 Cloud Assisted Live Streaming:

Cloud efficiently uses the resources to create dynamic pipeline and respond with fragmented data to the dynamics of demands from live streaming users. The live streaming process in server involves real time media Capture, Content processing, buffering and Efficient Transmission. The resources are freed after the streaming session.

With cloud, only one bit representation streams, probably high quality stream to be stored in the server .whenever the request comes from client, cloud encodes/transcode in to required bit representation using dedicated hardware and software engine quickly and transmits .Depends upon the capability of the client and bandwidth, the streams can be transferred with single fetching or multiple connection fetching to the client. The client always connect with the nearby edge server, the edge server to the origin server/storage is under the supervision of the cloud with very fast transfer speed. Failure with one path or server can be easily overcome with the cloud provisions. In case of incapable client devices, the cloud can do necessary processing and send only raw data to the client for rendering. For example Amazon EC2 creates the dynamic pipeline, jobs to transcode the media, Amazon S3 is used to store the content in HTTP cache and Amazon cloud front brings the media to the edge. The cloud client such as Amazon OSMF, Flow player is used to stream the content.

3.3 Cloud Assisted ON-Demand Streaming:

Cloud supports efficient delivery of on demand audio, video streaming content. With cloud infrastructure, it can store files with huge size. The media files moved to edge server to improve the transmission efficiency. Sometimes the stored files required be editing, concatenating, trimming or pre, post-processing required before transmitting to the client. This can be done seamless way by using hardware, software infrastructure of the cloud. It uses third party components if required. The complete operation is independent and does not involve any client handling. For example Amazon S3 stores and Cloud front brings the media to edge for RTMP based streaming to the users.

3.4 Amazon cloud streaming

Amazon Web provides various cloud based services to the client. Streaming is part of AWS provides media content delivery from Amazon simple storage (Amazon S3) or HTTP server to the end client using Amazon Cloud front. It is possible to stream the content directly from Amazon S3. Amazon web service supports live and on demand streaming using various methods such as Apple HLS(HTTP Live streaming), HTTP Live Smooth Streaming ,Adobe HTTP Dynamic Streaming, MPEG DASH, HTTP progressive download and Adobe RTMP streaming. Amazon Elastic Transcoder can be used to create dynamic bit rate media packets, index files and change from one to another media file format. The Amazon cloud streaming architecture presented in figure 2.

- **Amazon Cloud Front:**

The cloud front is the front end in cloud based Amazon streaming service enables the media files to be delivered to the client from the edge locations. Cloud front is the decision maker to decide the nearest edge location available and check the cache .If files are not available in edge cache, it initiates the connection with origin server, make edge copy and serve to the end user. It creates web and RTMP Distributions for HTTP/HTTPS and RTMP contents respectively.

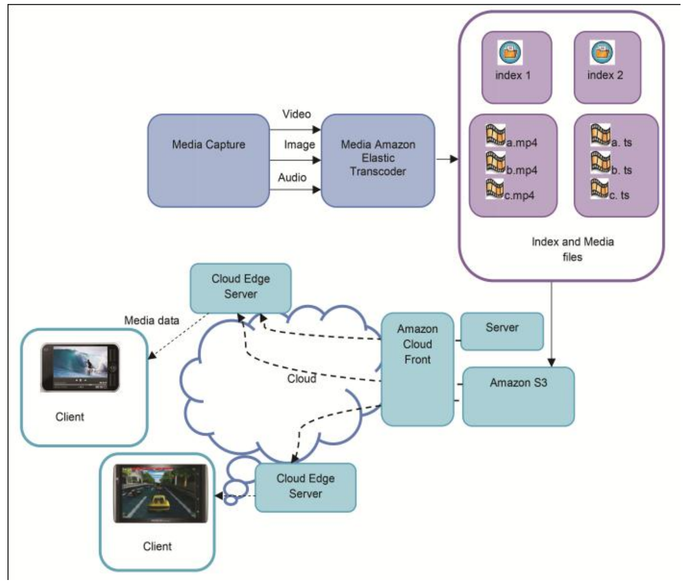

Fig.2. An illustration of amazon cloud computing platform architecture

- **Amazon simple storage service (Amazon S3):**

Amazon S3 is the storage service provided by the Amazon web services. The various interfaces of Amazon S3 is exposed to the content up loader and user for store, manage and retrieve the data from any device. It provides flexible, cost effective environment for the streaming compare to the HTTP or dedicated web server

- **Amazon Elastic Cloud Computing (Amazon EC2):**

Amazon EC2 does the cloud hosting service to the users. The EC2 interfaces can be used to create, launch, load and configure the custom environment image for the requested platform. The resources can be created dynamically and freed whenever not in use.

3.5 Google Cloud streaming service:

Google plays diverse of roles in offering cloud services. Streaming over the cloud is one of the most wanted service. Google cloud platform offers integrated audio, video streaming platform and application service. It uses various cloud platform components dynamically to build the service based on the user request. The architecture of the Google cloud services illustrated in below figure 3. The content developers and users can upload the media files in Google cloud storage either directly using the cloud storage interfaces or via Google App engine.

For streaming session ,the client Application make the HTTP request to the Application engine through cloud streaming interfaces to get the media files URL hosted and select the specific media file URL to stream the data. The Application Engine validates the request and instantiates to handle the media streaming. Application engine can access the Google Cloud Storage to get the meta data of the file. The App engine storage stores the meta data and user information of the file. Application engine interface allows users to get the file information and routes the media data to network edge location HTTP servers via Google global network and delivered to the users from cache to improve the speed of the service. Google Music is a cloud storage service is an example for audio streaming .It allows upload and stream using any capable device with Google Music app.

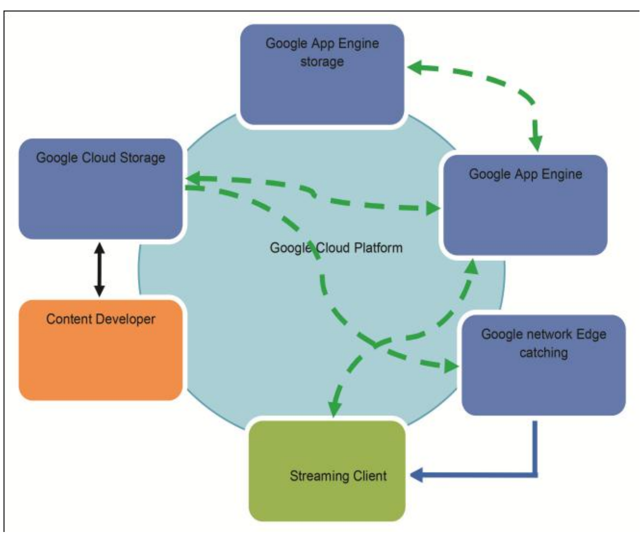

Fig.3. Google Cloud streaming platform architecture

3.6 Microsoft Azure Cloud streaming services

Microsoft has Azure cloud platform which provides various services to the cloud users. It takes care of service provisioning, monitoring, management, processing and storage. Azure media Streaming enable the media content to be delivered in a seamless way to clients across windows, android, iOS and other platforms. The figure 4 captures the end-to-end Azure streaming flow.

Azure media streaming provides scalable streaming solution using the core components within it. Content uploader to upload on-demand media into azure media platform (stored into Azure Blobs). In case of Live streaming, it directly feed data into encoder and encoder/transcoder takes care of compress/transcode the input media data into supported audio, video formats and generates index files if required. Content protection provides right management and access control of the media contents. Streamer interacts with the distribution network and sends the data to the user. Applications can request media available in the server using RESTful interfaces provided by the platform.

The platform supports HTTP progressive download, adaptive smooth streaming and HTTP live streaming methods. Dynamic content transcoding is one of the key features supported by Azure media streaming where the media files are stored in one format and transcoded to application requested format on the fly.

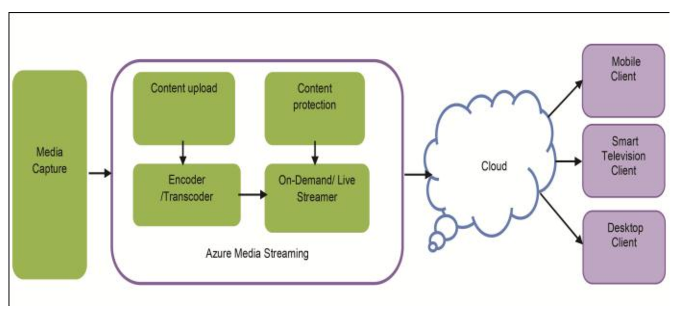

Fig.4. Windows Azure cloud streaming platform architecture.

4 EXPERIMENTAL SETUP

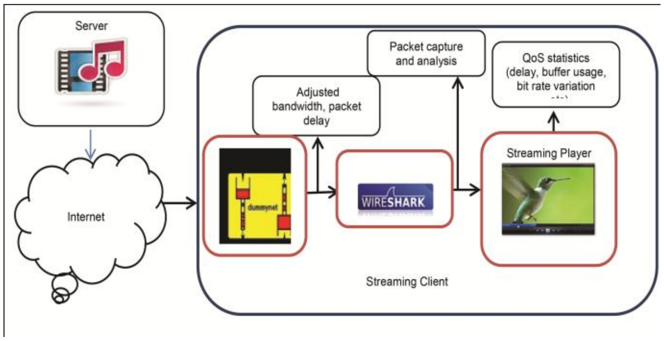

Fig.5. Experimental setup

The internet experimental test setup for conventional RTP, HTTP-PD , Adaptive streaming HLS, DASH is shown in Figure 5. The representation of dataset used for testing is is encoded with bit rates from 100Kbps to 6000 Kbps and quality level ranging from 240p up to 1080P resolution with 16 different levels .To maintain the testing consistency, each time the video streamed from the server in internet environment and streamed over the duration of 300 seconds. There are enough debug messages added in receiving streaming player components to capture the session logs along with wire shark packet analysis.

The streaming evaluation has been carried out network simulation with different internet bandwidth configuration using dummy net tool. Dummynet is a network emulation tool, which serves to access and limit link bottleneck. Internet experiment use the same segmented video files for testing. The bandwidth is configured from 4 Mbps in order to study the system behavior under the same network availability. Segment duration 10 seconds for HLS and 4 seconds for all other streaming methods are taken for experiment. HLS allows only fixed 10 seconds duration segments. The experiment is aimed to measure and compare the following metrics useful to analyze the performance of the rate adaptation.

1. Average throughput: The average received output of the streaming media is calculated for the duration of 300 seconds. Average throughput is one of the metric to measure the effectiveness of the stream switching algorithms for the available bandwidth.
2. Average PSNR: Average value of peak signal to noise ratio of the received video. PSNR gives the efficiency of the received throughput versus the target bit rate and it defines the quality of the stream.
3. Average Delta delay: It is the average delay between each received consecutive packet.
4. Average Bandwidth utilization: The bandwidth utilization measures the actual use of available link capacity. It is important performance metric of the client rate adaptation.
5. Packet loss: The number of packets arrives later than play out latency due to network/retransmission delay or retransmitted due to loss or completely lost in transport.
6. Packet loss/retransmitted count: The number of packets lost during transmission or packets discarded due to late arrival.

5 RESULTS AND DISCUSSION

Table.1. streaming performance metrics comparison

	Avg. good put	Avg. PSNR	Avg. Delta Delay	Avg.BW utilization	Lost/ Retransmitted packets
	(KB/Sec)	(dB)	(ms)	(%)	(Count)
RTP	147.17	16.13	3.886	31	235(0.3%)
HTTP-PD	82.746	15.92	4.274	20.992	78(0.37%)
HLS	225.86	18.87	5.133	46.08	101(0.17%)
DASH	164.209	19.76	4.67	33.63	640 (0.99%)
Cloud Streaming	342.861	22.19	4.37	70.03	337(0.58%)

The key streaming performance metrics are measured in the experiment for the duration of 300 Seconds and the same are listed in table 1. It shows that the cloud based adaptive streaming outperforms in terms of PSNR, average good put and bandwidth utilization. Since the cloud streaming evaluation uses amazon streaming environment where the amazon cloud front is located near to the end user location which enable fast data transfer and efficiently uses the bandwidth by transferring good quality data. Adaptive streaming performs average and efficient than traditional streaming .Though adaptive streaming uses same bandwidth and transfer various quality data based on the user need so that the end user experience is exceptional compare to the conventional streaming.

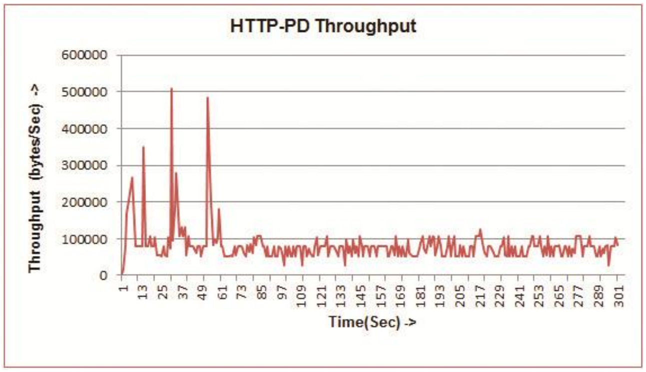

Fig.6. HTTP-PD streaming throughput

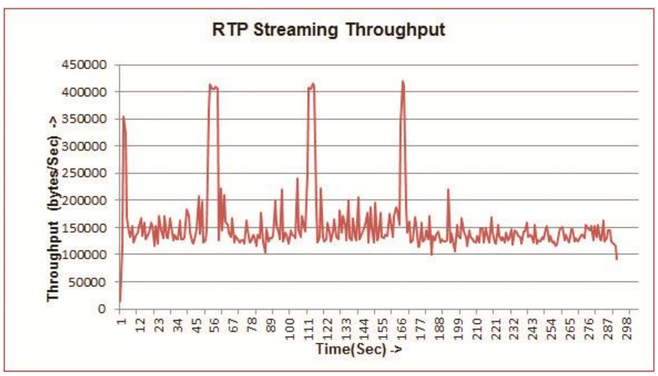

Fig.7.RTP streaming throughput

The number of packets arrives later than play out latency in due to network/retransmission delay or retransmitted due to loss or completely lost in transport (UDP transmission method) impacts the application quality. Considerable amount of packets are retransmitted/lost in MPEG DASH and cloud adaptive streaming compare to conventional streaming. This occurs due to more network congestion and incompetent rate selection for some duration.

The actual data (throughput) received is measured in the client machine where the streaming player is running. The figures 6 to 10 represents the captured throughput for RTP/RTSP, progressive downloads, HLS, DASH and Cloud Assisted-Adaptive streaming respectively. We could understand that the cloud streaming maintains high throughput throughout the session and does not go below 300K where other methods throughput oscillates between 150K to 400K most of the times. The graph shows that the cloud streaming uses bandwidth very efficiently and adaptation logic works very well than other methods. It is evitable that HLS and DASH throughput oscillates many times whenever the network condition changes in turn client changes rate. It also forces client to handle the varying size incoming data.

The packet arrival delay in streaming methods varies from 1 ms to 900 ms and the same captured in figures 11 to 15. RTP streaming maintains consistent low packet arrival delay due to connectionless nature of the underlying transport method. Sometimes Adaptive streaming packet arrival delay goes up whenever network bandwidth goes down and frequent rate change decision. The delay can be minimized further by optimizing the rate change algorithm. Even though cloud assisted method uses adaptive streaming the delay considerably low due to edge location streaming and efficient resource handling at server.

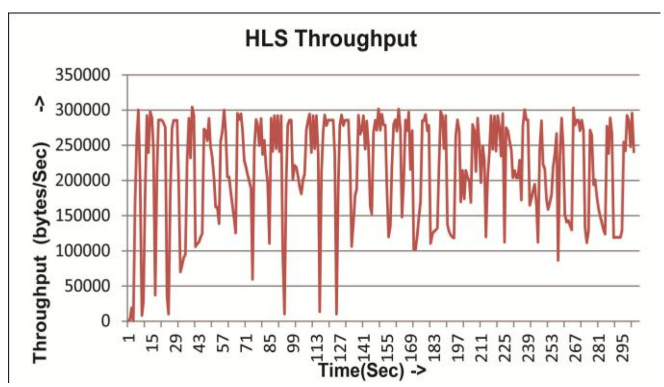

Fig.8. HLS throughput

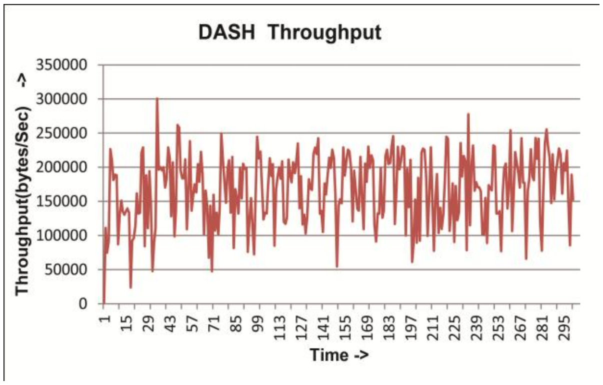

Fig.9. DASH throughput

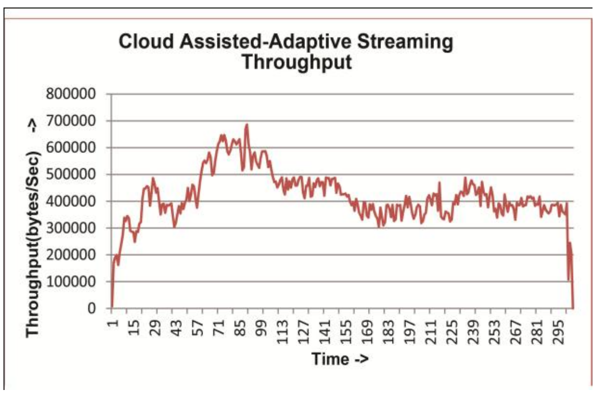

Fig.10. Cloud Assisted-Adaptive streaming throughput

We have evaluated delay of the first fragment reach to the client from the point user starts the session which we call as Initial fragment delay. As shown in figure 16, when the video is streamed using cloud distribution, the delay is less than 0.5 seconds and video can be displayed almost immediately. If the video is streamed in progressive download, the delay is more than a second. Delay can be significantly reduced if the user accesses the server with higher bandwidth network.

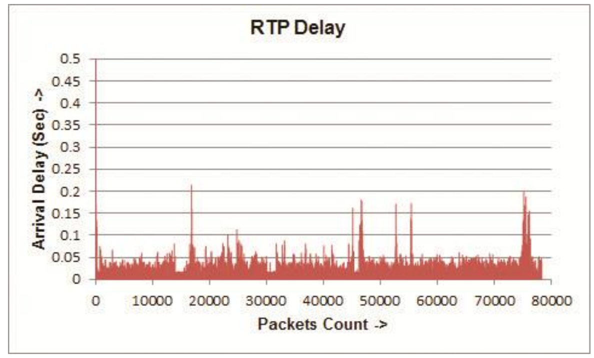

Fig.11. RTP Delay

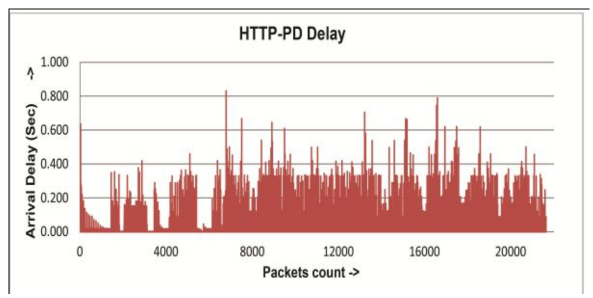

Fig.12. HTTP –PD Delay

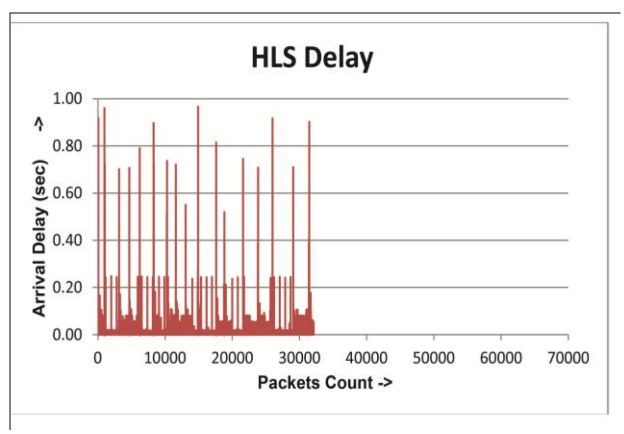

Fig.13. HLS Delay

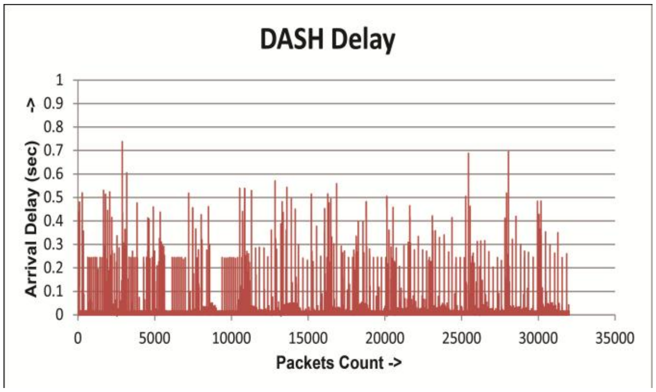

Fig.14. DASH Delay

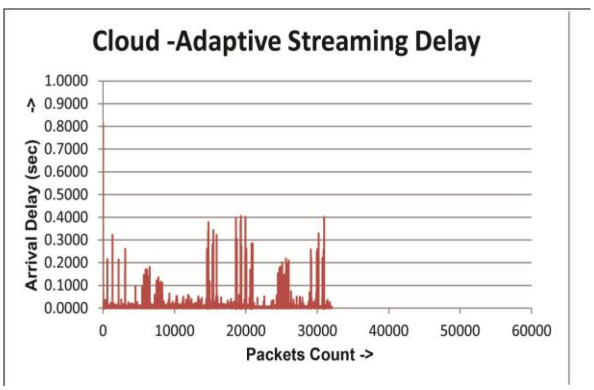

Fig.15. CA-Adaptive streaming Delay

From figure 17, it is evident that many times buffer full and empty condition occurs in cloud based streaming delivery and progressive download. This is because of the end client data consumption from buffer by the streaming player is not properly matching with received data rate due to more device load and varying network transmission speed and also client player is lack of effective rate change logic.

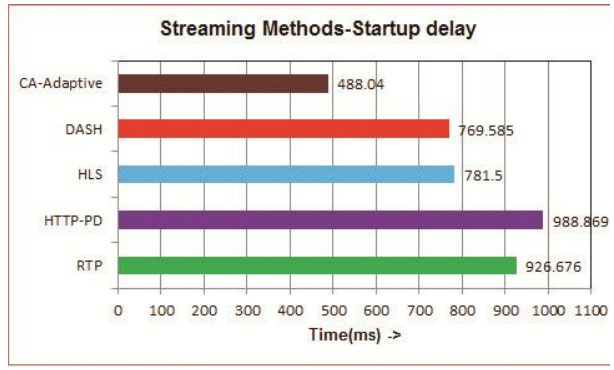

Fig. 16. Startup delay comparison

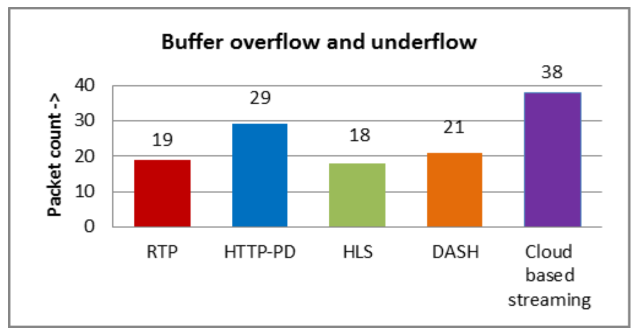

Fig.17. Buffer overflow and underflow occurrences

6 FUTURE WORK AND CONCLUSION

In this paper, we have discussed evolution of cloud streaming service delivery with underlying technology and architecture with example of certain familiar cloud streaming models. The performance metrics of streaming methods and efficiency are measured, which gives the behavior understanding of the streaming methods in the bottleneck environment and provides more research possibilities in the area of improvement required thereby paving way for further research and enhancement. The outcome of the performance result shows that cloud based delivery excels in efficient use of resources and overall quality of experience. It also helps to understand issues and challenges of the current implementations of the streaming models. Extension to this work will involve analyze the cloud multimedia service under different network conditions and propose solution to address all the performance issues.

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