Institute of Advanced Engineering and Science

International Journal of Cloud Computing and Services Science (IJ-CLOSER)
Vol.3, No.2, April 2014, pp. 109~124
ISSN: 2089-3337

Conventional to Cloud: Detailed survey and comparative study of multimedia streaming rate Adaptation

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Article Info

Article history:

Received Jan 29th, 2014
Revised Feb 24th, 2014
Accepted Mar 10th, 2014

Keyword:

Adaptive streaming
Conventional Streaming
Rate adaptation
Quality of Service

ABSTRACT

Infotainment and telecommunication industry is fast evolving towards personalized network connectivity and newer multimedia application services delivered ranging from music playback to ever changing telephony applications. Streaming is most important service, enables the users to view real time multimedia content on-the-go anywhere and everywhere. Quality of service is a major concern in the increasing network traffic and high user demand. Rate adaptation is crucial process which dynamically evaluates, select and control the media rate based on the network deviation, system processing capability and to ensure the best user experience to the consumer. In this paper, the authors conducted comprehensive survey of existing rate adaptation models and algorithms used in conventional, adaptive, cloud assisted streaming methods, lists the important merits, limitations of those algorithms. With an experiment setup, the rate adaptation behavior of each streaming models are evaluated and compared with the other streaming techniques. The analysis shows that adaptive and cloud assisted streaming quickly performs well in adapting to the network variation compare to the conventional streaming models.

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1. INTRODUCTION

Video occupies most of the Internet traffic and going forward more link bandwidth engaged by streaming video players to send high resolution content to the consumers. Media streaming is the most appropriate method to retrieve the video from server. In the complex network environment to ensure best Quality of Service (QoS) is really a challenge and it requires more processing power at client, massive storage and high speed transport. Content download and rendering is a traditional method to watch the video content which is most suitable for smaller size files .Incase of larger size files, user has to wait long time to complete download.

Conventional RTP based streaming [1] uses User Datagram Protocol (UDP) for real time delivery with simplified client and server architecture. RTP provides a flexible framework for delivery of real-time media, such as audio and video, over IP networks. However it suffers with issues like server infrastructure, Network Address Translation (NAT) and port blocked in network firewalls. Alternate to UDP, transmission control protocol (TCP) can be used for streaming delivery and overcomes above issues. Real-Time Messaging Protocol (RTMP) [4] is proprietary Adobe streaming protocol used for streaming over TCP. Hypertext Transfer Protocol (HTTP) Progressive download [8] over TCP allows users can download the media content from web server and render at the same time in client device. As soon as the file downloads starts, client invokes the media player to play after sizable data available in the buffer. The user can store complete data and play the content whenever required without downloading again from the server. However promised packet delivery and packet retransmission in TCP consumes extra bandwidth and time which restricts the real time end user experience.

Adaptive HTTP streaming [8] 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 HTTP adaptive streaming is the replacement to traditional models and overcomes most of the issues faced by customary 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. Numerous adaptive streaming methods developed by different vendors like Microsoft smooth streaming [10], Apple HTTP Live Streaming [11] and Abode HTTP Dynamic streaming [12]. Each method uses different file formats and transportation standards to deliver the content. Dynamic Adaptive Streaming over HTTP (DASH) [13][14] common standard proposed by MPEG which provides unique architecture.It is broadly used, accepted as HTTP adaptive streaming standard by most of the industry leaders.

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. With the advent of cloud brings the multimedia content delivery and rendering data with less effort and cost, without compromising in data quality and security. Cloud based streaming model 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. 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 the current trend, multiple companies have started providing content management systems and streaming delivery over the cloud like Amazon Elastic Compute Cloud (EC2)[15], Google Music[16], Amazon Simple Storage Service, cloud front [17] Apple Cloud service streaming media cloud etc.

Packet switched network allows streaming to push unlimited data into network which introduce network congestion lead to dropping the packets in network and client buffer are filled above some threshold. UDP based streaming server transmits media at the encoding bit rate to the client. In case of TCP conventional streaming, the server transmits the stream in available network bandwidth to fully utilize the available network resources. Under ideal network condition, the media player playback buffer filled and consumed at the same rate. Playback buffer stays at balanced level so there will not be any buffer over or underflow. The play out buffer design plays important role in real time streaming application, where it can accept few milliseconds to few seconds delay. However packet data network is heavily loaded and it does not ensure the promised throughput all the time during the session. This cause buffer drain eventually affects media quality. It is quite common experience of watching internet video with glitches and interruptions, though high speed network connection available. To avoid the problem said above media rate control and adaptation is critical to optimize the streaming performance. Streaming methodologies put intelligent system in place to calculate the network metrics and adjust to the current circumstances which brings tradeoff between network bandwidth and media quality. The client media player initially stores certain amount of data needs to be captured to balance the throughput variation. However this is not enough to compensate the jitter for the entire duration of the session. Rate adaptation can be done using various methods such as variable bit rate methods (SVC H.264), dynamically switching codecs and switching the stream level.

The main contributions of this paper include the following three aspects. First we capture the underlying architecture of rate adaptation models followed in traditional, adaptive and cloud streaming methods with key metrics used to evaluate the adaptation . Second, we completely review the existing rate control methods used in across different streaming methods. Finally, we setup the experimental setup to evaluate the streaming adaptation behavior from conventional to cloud and compare the results. The rest of the paper is organized as follows. Section 2 gives a brief overview of the terminology and architecture of the streaming rate adaptation models. The review and analysis of existing rate control methods are presented in section 3. The experimental setup and evaluation of streaming model rate adaptation metrics are captured in Section 4. Section 5 lists few issues with the existing rate adaptation model and proposes future direction. Finally section 6 concludes the paper.

2. STREAMING RATE ADAPTATION

2.1. Introduction

Network congestion and device capabilities plays important role in end user video experience .It is important to control network congestion and adjust the video playback to get good quality of video delivery at the end client. Even though congestion control procedures are implemented in different levels of communication network stack, the media Quality of Service(QoS) is mainly depends on server and client rate control.

2.2 Rate adaptation system model

The figure 1 exhibits the rate adaptation problem of the system. The rate control research problem can be articulated such a way that the receiving throughput almost closely matches to the sending rate of the server so that there will not be degradation of the video stream quality and efficient use of the bandwidth. Overflow arises when the receiving rate exceeds the upper threshold and underflow occurs when receiving rate goes below lower threshold. Data loss/drop occurs when receiving rate go beyond playback rate.

Table 1.Rate Adaptation model notations

Notation	Definition
A(t)	Actual arrival rate in client at time t >0
p(t)	Playback rate in the client at time t >0
B(t)	Data remaining at buffer at time t>0
α	Pre-buffered duration
I	Lower threshold
u	Upper threshold
E(t)	Error Rate at time t>0
R(t)	target network sending rate at time t>0
d(t)	Decoding rate at time t>0
Si	Packet sending time at server i=0,1,….N
Ri	Packet receiving time at client i=0,1,….N
Di	Packet delay i=0,1,….N
Ji	Packet Jitter i=0,1,…..N

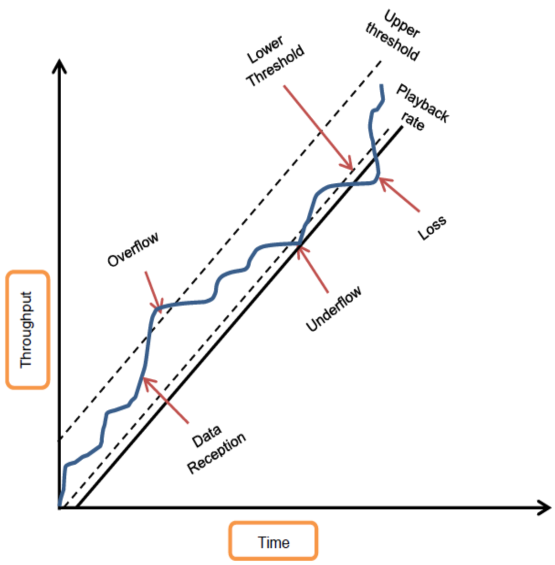

Figure.1.Rate Adaptation system

The key notations listed in Table 1 are used to model the rate adaptation.Our final goal is to calculate target receiving rate R(t) over the interval and adopted to new calculated target bit rate. Switching to new target bit rate depends on the actual bit rate received in the client and client buffer data level.The key parameters can be measured using rate adaptation model described below. R(t) is the function of playback rate , actual received rate at client and data remaining at the client buffer. It can be defined as.

$$R(t) = Fn(p(t), A(t), B(t)) \qquad (1)$$

p(t) is playback rate depends on decoding rate d(t) and pre-buffered data availability α . It is define as

$$p(t) = Fn(m(t), k) \qquad (2)$$

For the streaming media encoded at constant bit rate, the decoding rate should be same as encoding rate for the entire duration. But for the variable rate encoding, the bit rates are keep changing. The rate adaptation efficiency can be measured using the difference between target bit rate and actual arrival bit rate. It is called as error rate and it falls within the lower and upper threshold. It can be expressed as

$$l < E(t) < u \quad (3)$$

$$E(t) = R(t) - A(t) \quad (4)$$

Error rate E(t) depends on many factors in server, network and client such as encoding delay, serialization delay ,network jitter , network buffer queuing ,network congestion and de-packatization delay. When the error rate is within lower and upper threshold there is no need of switching new bit rate. Adjusting to new bit rate is required when E(t) crosses the upper or lower threshold .If E(t) is below lower threshold, R(t) should be reduced. If E(t) is above the upper threshold ,R(t) should be increased. The value of l and u are important because it decides the efficiency of a rate adaptation algorithm.Lower threshold calculated as sufficient amount of data required to continue with current playback rate, data arrival rate and pre-buffered content without underrun till next rate adaptation time or next fragment download. Upper threshold calculated as sufficient amount buffer free space available to hold the date with current playback rate, arrival rate content without overrun till next adaptation time or next segment download.

2.3. Streaming Rate Adaptation measurement models

To implement the rate adaptation schemes sender and receiver should have the mutual cooperation .The sender should know about the receiver reception statistics. In UDP transport, application level protocol RTCP can be used to send the receiver statistics to the sender either constant or flexible interval. In TCP, congestion control messages and receivers acknowledgement messages for correct received fragments are used. Based on the feedback messages, senders can correct the sending bit rate based on the current network state.

In multimedia streaming the rate adaptation can be classified into three states namely overflow, underflow and steady based on the client buffer data level and playback rate. In overflow state, the receiving rate exceeds the player consumption rate causing playback buffer overflow and losing data packets. The overflow state occurred due to drastic change in network throughput or unexpected client device load. During underflow state, the receiving rate drops below the consumption rate thereby causing buffer drain and playback interrupted .underflow occurs due to sudden drop in the network throughput .Steady state the receiving rate falls always within lower and upper threshold of the playback buffer .The network throughput rate is almost near to the consumption rate.The rate adaptation can be driven by either sender or receiver or network or cooperative. In sender driven rate adaptation the receiver collects network statistics and sends to sender where it can be used model the current scenario and to adapt its sending bit rate. An example of sender driven operation is shown in figure 2.

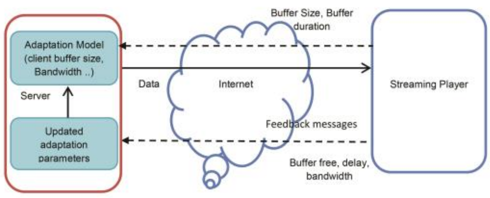

Figure.2.Sender Driven rate Adaptation calculation model

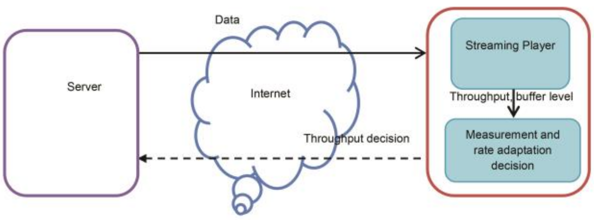

Figure.3.Receiver driven rate adaptation calculation model

In receiver driven rate adaptation, the receiver collects all required data chooses appropriate sending rate and communicate the decision to sender to adapt the bit rate. An example of client-driven rate adaptation used in HTTP progressive streaming and described in figure 3. In network driven adaptation, the network elements such as mixer, router detects the traffic and sends feedback messages to the sender which can be used to calculate the sending bit rate. This kind of rate adaptation mainly used in multi participants streaming session. In Cooperative based adaptation, the sender, receiver and network elements together evaluate the variation and take the decision.

2.3.1. Throughput measurement:

Multimedia applications are bandwidth greedy. Constant network throughput is critical to carry continuous multimedia for real time delivery of steaming multimedia contents. Actual incoming average throughput measurement for the interval T can be done at time t using the formula

$$\overline{A(t)} = 1/T \int_{t=0}^{T} A(t) \qquad (5)$$

2.3.2. Play out Buffer management:

In streaming session, playback buffer is filled to certain amount before starting the actual playback. Playback buffer bring the balance between network jitter and playback. It also removes the effects of jitter from the stream, by buffering each arriving packet for a short interval before playing it out. Client controls the buffer management functionality. High network jitter causes buffer underflow and re-buffering creates playback pause until the re-buffering is complete. The duration of the re-buffering time varies based on the network condition. Playback of prefilled data ensures continue playback even though high network jitter occurs sometimes. To reduce network jitter, media packets are to be captured as early as possible. Play out buffer management is required to removes jitter delay and regulates the receiving data flow. One possible solution could be dynamically adjusting the receiver buffer size so that the streaming performance can be improved. The client buffer data availability b(t) at time t can be calculated using the expression

$$\overline{P(t)} = \begin{cases} 1/(T-\alpha) \int_{t=\alpha}^{T} P(t) & t > \alpha \\ 0 & t < \alpha \end{cases} \qquad (6)$$

$$B(t2) = B(t1) + \int_{t=t1}^{t2} A(t) - \int_{t=t1}^{t2} P(t) \qquad (7)$$

2.3.3. Network Jitter:

Receiving throughput fluctuates mainly due to network jitter. It is the deviation of the difference in packet spacing at the receiver compared to the sender, for two consecutive packets. The jitter may vary from packet to packet due to variations in the network conditions. Low latency network, the delay is acceptable to delivering real time content. Network Jitter can be calculated using the below formula.

Packet delay between ith and i-1th packet

$$\mathrm{D(i, i-1) = (Ri - Ri - 1) - (Si - Si - 1)} \qquad (8)$$

Packet Jitter at ith packet

$$\mathrm{J(i) = J(i-1) + (D(i, i-1)/16)} \qquad (9)$$

2.3.4. Client Device Capabilities:

Even though proper network resources available, selection of capable device plays important role in quality of streaming service. The device capability metrics such as size, display, hardware acceleration, processing power and memory needs to be considered while choosing the device for multimedia applications.

2.4 Adaptation Control system model

Figure 4 shows the simple adaptation feedback control system to model the rate adaptation calculation. The adaptation controller goal is to minimize the error rate almost to zero so that sender rate matches to the receiving rate and client buffer will be at steady state. The controller takes input E(t) and minimizes the error by varying propositional parameters. Stream adaptation calculator takes the input error from controller and evaluates the algorithm with actual throughput, client buffer level to fix threshold value and outputs the signal. The Bit rate quantizer receives the appropriate bit rate, moves up and down to the nearest feasible bit rate and get the target output rate R(t).

3. REVIEW OF RATE ADAPTODATION SCHEMES:

Adaptation is one of the key mechanism to solve congestion problem in the network .There are many researches in congestion and rate adaptation to avoid packet loss, buffer overflow/underflow and smooth playback. Figure shows the different methods used in conventional and adaptive streaming delivery.

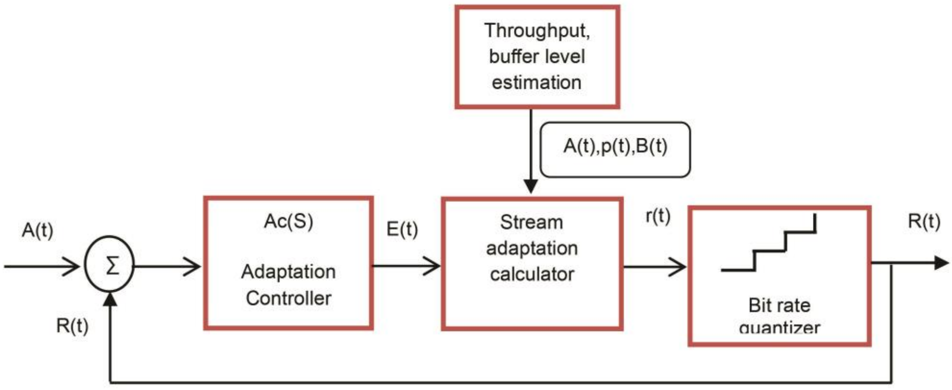

Figure 4.Rate adaptation control system

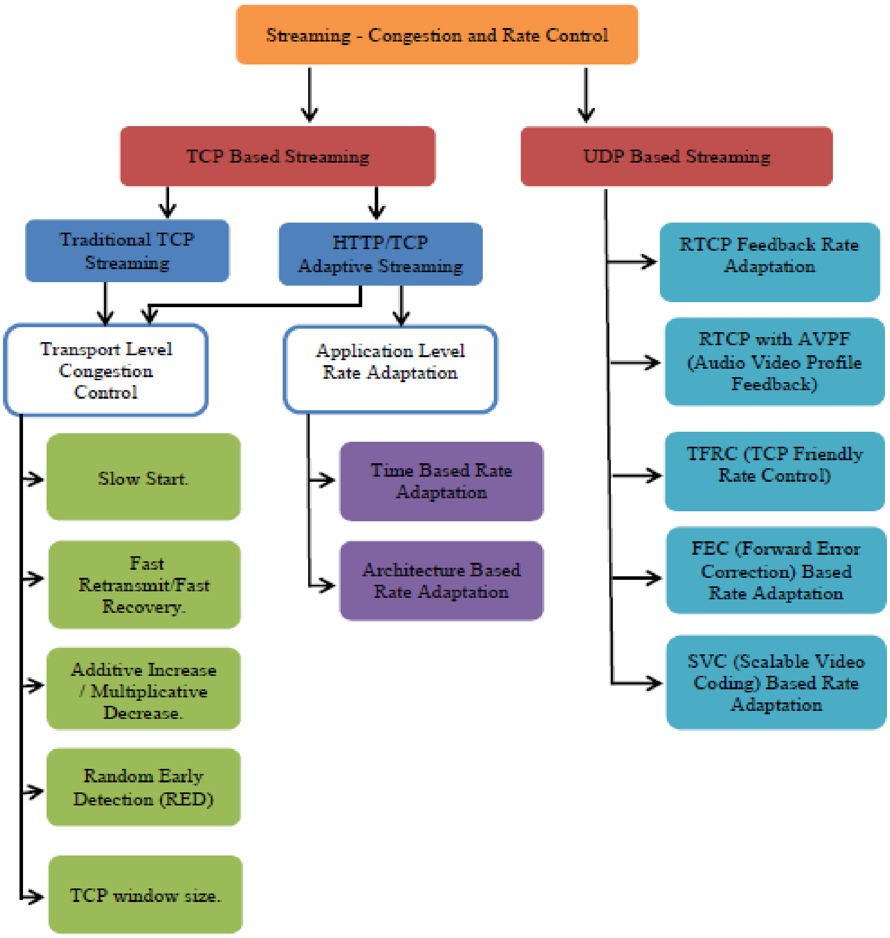

Figure 5. Congestion and rate Control techniques.

TCP is most reliable most widely used transport protocol in the internet world and it plays important role in overall network functionality .TCP provides adequate algorithms to control congestion and data flow. The following are some of the key techniques used in TCP transport to detect and control congestion. Table 2 lists the literature review of different congestion control algorithms are used in TCP transport and its characteristics.

Slow Start: It is used in conjunction with other algorithms to avoid sending more data than the network is capable of transmitting to avoid causing network congestion .It helps to start fast and prevent full window at once.

Fast Retransmit/Fast Recovery: Fast retransmit technique detects the lost packet using ACK and TCP Sender retransmits the lost packet. Fast recovery used in conjunction with fast retransmit which reduces the congestion window and start the recovery process using linear addictive increase.

Additive Increase / Multiplicative Decrease: It allows host holds the congestion window and detects the congestion using packet delivery or drop based on the Acknowledgement and increase/decrease the sending rate.

Random Early Detection (RED): Random Early detection algorithm controls the congestion by dropping the packets with certain probability whenever network traffic crosses the threshold.

Increase TCP window size: Increase TCP window size method avoids congestion up to some level and offers better performance.

Streaming methods relying on UDP transport does not provide any rate control mechanism by the transport protocol level and the rate control is completely depends on client and server application stack. In general the client application sends QoS metrics in feedback messages at constant interval to server and those numbers can be used in server to model and decide the rate switch. In RTP based streaming, congestion indicators provided by standard RTCP feedback insufficient for rate adaptation of conversational applications. Various other methods rate control methods such as Extended RTCP, TCP Friendly Rate Adaptation (TFRC), and Forward Error Correction (FEC) based rate adaptation are widely used in RTP streaming. The literature analysis of RTP streaming rate control and adaption models and its classifications are given Table 4.

RTCP Feedback messages: RTCP feedback reports contain the QoS metrics such as packet loss, delay, jitter, and received throughput, play back buffer usage .The server to evaluate and choose the appropriate rate.

RTCP with AVPF (Audio Video Profile with Feedback): Extension to RTP audio video profile, allows more relaxation in feedback report timing to send transport, payload and application events/ statistics whenever required without exceeding the RTCP bandwidth.

TCP Friendly Rate Control (TFRC): Rate control mechanism of the UDP should be fair and evaluated based on the TCP traffic flow which allows improving bandwidth utilization towards TCP.

Forward Error Control (FEC) based Rate Adaptation: Forward Error Control mechanism not only for error control and recovery but can be used to adapt the rate control mechanism.

SVC (Scalable Video Coding) Based Rate Adaptation: Scalable Video coding allows sender to generate different scalable streams based on the network variations and supports for rate adaptation.

In addition to the traditional TCP congestion control methods, application level rate adaptation needs to control the end –to-end network variation and streaming client effectively. Constant monitoring of network variation required to transmit best possible bit rate to the client. Network metrics such as packet loss, network jitter, delay, round trip time needs to be calculated time to time along with client buffer usage to make the proper stream/bit rate selection decision either in client, server or both. The server sends at maximum possible rate and playback buffer used in client to smooth the playback control mechanism. Rate adaptation in adaptive media streaming over HTTP and TCP protocols presents unique challenges and obstacles.

Table 2. TCP congestion control methods

			TCP based Transport congestion Control	
No	Congestion Control Mechanism	Algorithms	Advantages	Limitations
1	Slow start	Ref:[65],[66]	1. slow start provides slow start and exponential increase and decrease the rate based on the network load. 2. It helps to reach balanced sending rate quickly.	1. Slow-start blindly treats unacknowledged data fragments are due to network congestion as lost fragments. 2. It performs badly in constrained network conditions. 3. Few extra resources are needs to be maintained for computation. 4. Network components may not cope with the transmission rate change.
2	Fast Retransmit/ Fast Recovery	Ref:[65],[66]	1. Fast Retransmit improves the throughput improvement approximately by 20%. 2. Fast Retransmit will retransmit the packet immediately based on acknowledgment instead of waiting for timeout to occur. 3. Fast Retransmit eliminates spurious retransmissions and nearly half timeouts. 4.When congestion occurs , the recovery starts from the point using additive increase 5. Fast Recovery Algorithm adapts to resource availability .	1. Fast recovery triggers retransmission of a packet faster than permissible by the regular time-out 2. Fast retransmit does not eliminate all the timeouts.
3	Additive Increase / Multiplicative Decrease	Ref:[65],[66]	1. It provides stability over TCP congestion control. 2. Gives fair, accurate timeout and efficient mechanism. 3.It periodically probes for available bandwidth by increasing the rate	1.It takes too long to ramp up a new TCP connection from initial start
4	Random Early Detection (RED)	Ref : [67]	1. Only few packets are dropped much earlier than other models. 2. Preserve the buffer resources from complete exhausted level. 3. It avoids a bias against burst traffic.	1. RED does not provide quality of service (QoS) differentiation. 2. RED does not solve unfairness among many connections where all of them get congestion indications.
5	Increase TCP window size	Ref:[68]	1. Increase window to send faster; decrease to send slower. 2.Cheap to implement, good failure properties	1.It requires bigger size buffers and creates more traffic bursts

Table 3. Adaptive HTTP streaming Rate Adaptation methods

			Adaptive HTTP/TCP streaming	
No	Rate Control Mechanism	Algorithms	Advantages	Limitations
1	Time Based Adaptation	Ref:[71],[72],[73], [74]	1. It gives optimal user experience. 2. It helps to take appropriate action on time to keep the bit rate in control. 3.Server and client reacts in time and get enough time to performing rate Adaptation.	1. It will not suit for random varying network conditions. 2. It will not include include the transport layer informations to calculate rate change so the rate change will not match end to end network capacity always.
2	Architecture based Adaptation	Ref:[69],[70]	1. Evaluation starts only after the event triggered. 2. Rate Adaptation calculation uses the QoS metrics of client, server and network effectively.	1.Incase of more packet loss ,delay and session discontinuity, client and server rate adaptation reacts after the event 2. Estimation based on the past informations of network and client .

Table 4. UDP based streaming Rate Adaptation methods

		UDP based Streaming		
No	Congestion Control Mechanism	Algorithms	Advantages	Limitations
1	Normal RTCP Feedback messages	Ref:[18],[19],[20]	1. Provides QoS metrics in feedback to avoid spending many bytes which are required to inform loss situation. 2.Feedback messages in sender helps to avoid excessive network congestion control	1. RTCP packets long reporting interval and the fixed timing transmission slowdown the adaptation decision. 2. Waits till regular RTCP interval to carry the reports. 3. Congestion due to RTCP packets traffic. 4. Increases the average delay.
2	RTCP with AVPF(Audio Video Profile with Feedback)	Ref: [21],[22], [23], [24], [25], [26], [27], [28]	1. Flexible Timing and Early RTCP packet scheduling allows to report the event immediately. 2. Flexibility to report RTP Transport, payload and Application layer events/ information. 3. Allows network components to request for adaptation.	1. RTCP AVPF Feedback reports are insufficient for prompt congestion and rate control as it operates slower than other congestion control methods. 2. Feedback messages cause additional overhead especially on low-speed network. 3.Increases average packet transmission delay
3	TCP Friendly Rate Control (TFRC)	Ref: [34],[35],[36],[37],[38],[39] 1. Equation Based algorithm. Ref:[40],[41] 2. Rate Adaptation Protocol (RAP). Ref: [42] 3.Addictive increase multiplicative decrease (AIMD). Ref [43] ,[44] 4.Linear Increase Multiplicative Decrease with history(LIMD/H) :Ref :[45] 5.Distance weighted Addictive Increase and Loss dependent Multiplicative Decrease (DWAI/LDMD). Ref: [46],[47],[48] 6.loss-delay based adaptation algorithm (LDA/LDA+):[49],[50] 7. TCP Emulation At Receiver(TEAR). Ref:[51] 8. Direct Adjustment Algorithm (DAA). Ref:[52] 9 .Multicast TCP(MTCP). Ref:[53] 10.Pragmatic General Multicast Congestion Control(PGMCC) .Ref:[54] 12.Hop Based control method: Ref:[55] 13.Rate-adjustment congestion control protocol(TFRCP). Ref :[56].	1. TFRC performs lower variation of throughput and very well suited for delay sensitive and smooth transmission rate applications. 2. It provides the exact loss rate from receiver to sender via feedback mechanism.	1. TFRC requires feedback on a per-packet basis. 2. Receiver can send false report and avail more network bandwith share. 3.Delay in restoring bit rate back which give under utilization of the network resources
4	Forward Error Control(FEC) based Rate Adaptation	Ref:[33]	1. FEC based rate adaptation the sender is aggressively probe for available network capacity. 2. FEC redundant same packets used for error correction and rate adaptation purposes.	1.Rate changes adaptation is quite slow 2.It suffers with excessive coding redundancy

5	SVC Based Rate Adaptation	Ref: [29],[30],31],[32]	1. It does the rate adaptation by dropping packets to fit various network capacities without re-encoding the data. 2. It will not degrade the video quality immediately and gives acceptable video quality when video reception is bad. 3. It allows different end client device with different capability by serving different quality layers . 4. With slow catchup , SVC does not cause buffer under/overflow	1. SVC adds extra overhead which leads to degrade the performance especially in constrained network capacity links. 2. When network capacity improves ,It consumes more time to reach the optimal quality.

In Adaptive HTTP streaming rate adaptation calculated before start request every segment download. It require dynamic, aggressive rate adaptation algorithm to make decisions at the client. Rate adaptation in adaptive HTTP streaming is quite new and many research is going on in this area. Based on our literature review, the adaptive streaming rate control algorithms are designed to use one or more parameters such as network condition, media duration, playback speed, actual throughput, and segment fetch time and segment duration. Rate control in adaptive streaming is evolving. The few rate adaptation models used and its characteristics are illustrated in table 3.

Time Based Adaptation: Algorithm performs the adaptation and decides the rate control based on real time which critical for the end user experience.
Architecture based Adaptation: Algorithm performs the adaptation and decides the rate adaptation decision based on different network, sender and receiver parameters.

In Client backed streaming, cloud client follows the rate adaptation same as like conventional and adaptive streaming depending on the method it streams. However there is lot of room to do the research in this area. In Paper [75] proposes the cloud-assisted procedure to improve the user's quality of experience with cloud based downloading strategy. The cloud assisted model for live media streaming is discussed in paper [76].

4. EXPERIMENTAL SETUP

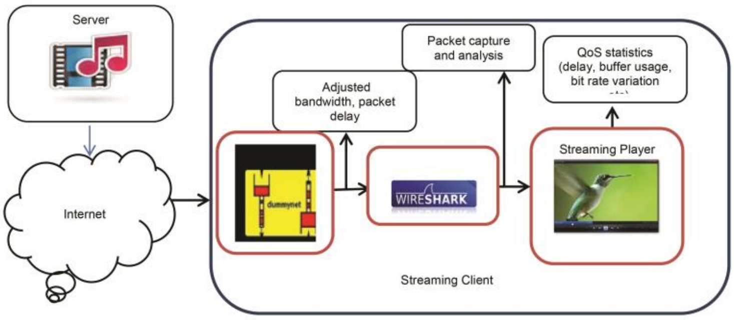

Figure 6.Experimental Setup

An experimental setup which evaluates conventional, adaptive and cloud based streaming models with transferring on-demand media content in the internet. Specifically, we compare the rate adaptation efficiency of different streaming methods using various performance metrics such as packet loss, startup delay and buffer overflow/underflow. The experimental evaluation of streaming techniques has been carried out by the test setup as shown in Figure 5.The host is running with Ubuntu Linux 12.04 desktop machine having 3.8.8 kernel with 4GB RAM and intel core i5 CPU .It has wireshark, dummynet and streaming media framework installed.

Two conventional streaming methods(RTP/RTSP, progressive download) ,two adaptive streaming methods (dynamic adaptive streaming over HTTP ,HTTP Live Streaming) and one Cloud assisted adaptive streaming (Adaptive HTTP Live streaming) to evaluate the performance in internet environment . The same network settings configured for all the streaming evaluation process using dummynet. It is a network emulation tool, which serves as our access link bottleneck .In this experiment we set the input and output bandwidth without any restriction in a 4Mbps internet broadband connection.

To test the conventional streaming method, the video encoded in 720P, 30fps is used. Adaptive and cloud based streaming uses video with different bit rates from 100Kbps to 4Mbps and quality level ranging from QVGA up to 720P resolution. To maintain the uniformity, each streaming method stream the video has play-out duration of 300 seconds .The media data flow analytics are captured in client and analyze the behavior of the streaming methods. We have also added enough traces in receiving media player components (RTSP source, RTP manager, HLSdemux, DASHdemux) to capture statistics along with wireshark capture analysis.

5. RESULTS AND DISCUSSION

The key streaming performance metrics are measured in the experiment and the same are listed in table5. It shows cloud based adaptive streaming outperforms in terms of PSNR 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

Table 5. Network performance metrics of streaming methods

	Avg. throughput	Avg. PSNR	Avg. Delta Delay	Avg.BW utilization
	(KB/Sec)	(dB)	(ms)	(%)
RTP	147.17	16.13	3.886	31
HTTP-PD	82.746	15.92	4.274	20.992
HLS	225.86	18.87	5.133	46.08
DASH	164.209	19.76	4.67	33.63
Cloud Streaming	342.861	22.19	4.37	70.03

Though adaptive streaming uses same bandwidth and transfer varying quality data based on the user need so that the end user experience is excellent compare to the conventional streaming.

The evaluation result in the figure 7 shows that the first fragment rendering delay from the point user starts the session which we call as initial fragment delay. The delay is significantly less in cloud and adaptive streaming distribution compare to the traditional methods. If the video is streamed in progressive download, the delay is closer to a second. In adaptive cases the server and client have the mutual understanding of bandwidth variation and rate change for every single fragment download. The delay can be reduced by adjust the rate change based on the network variation and transfer the data to the client .It also depends on the client streaming player buffering and parsing capability. Delay can be significantly reduced if the user accesses the server with higher bandwidth network.

From the figure 8, 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.The packet loss scenario is illustrated with the help of figure 9 .It indicates that 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 impacts the application quality. Considerable amount of packets are retransmitted in MPEG DASH and cloud adaptive streaming compare to conventional streaming. This happens because more network congestion in the environment and incompetent rate selection for some duration.

5.1 Summary

Based on the literature survey of existing rate control and adaptation algorithms and experimental evaluation, we found that the following issues are existing in the current system even though different strategy used.

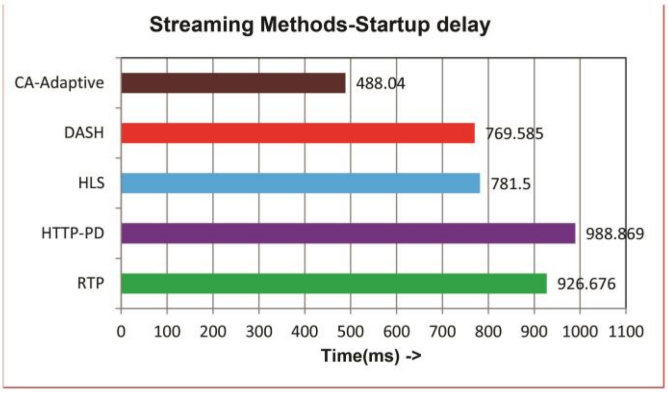

Figure 7. Startup delay comparison

1. Client fails to choose best likely bitrates with current available Bandwidth.
2. Interrupted playback and fluctuates between bit rate changes frequently.
3. More delay in session start up.
4. Pause -Playback delay over due to buffer overflow and underflow.
5. Client Players are not efficiently utilizing the available Bandwidth.
6. Improper Playback buffer size.
7. Client has buffer storage, processing and network bandwidth constraints.
8. The effective rate adaptation algorithm design in any streaming should consider the following factors
9. Estimate the available network bandwidth.
10. Measure server side processing constrains.
11. Calculate playback rate and client buffer data consumption.
12. Predict the network variance and decide time to download fetch the next packet of data.
13. Deliver based on end device processing capability.

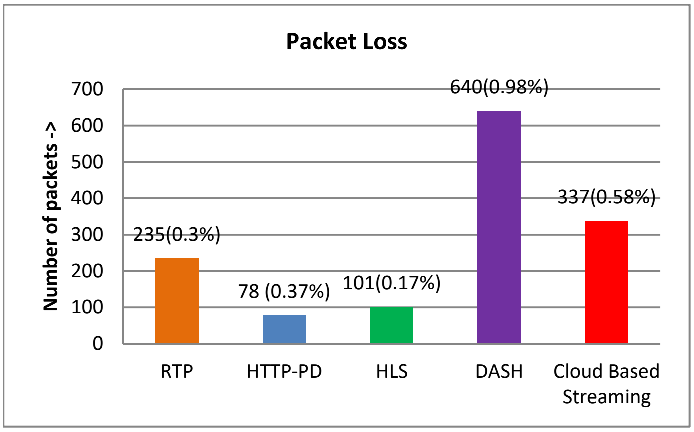

Figure 8. Streaming packet loss evaluation

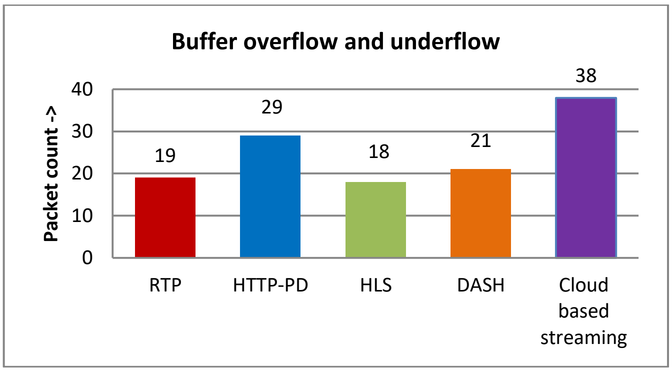

Figure 9. Buffer overflow and underflow occurrences.

6. CONCLUSION

In this paper, the rate control and adaptation strategies used different streaming models are reviewed. The different rate adaptation and evaluation models used in end to end multimedia streaming are discussed in brief. The rate adaptation performance metrics of different streaming methods and efficiency are measured, which gives the behavior understanding of the rate adaptation in the bottleneck environment and provides more research possibilities in the area of improvement required. The outcome of the performance result shows that HTTP adaptive and cloud assisted streaming 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 design and implement novel adaptation logic to address all the issues faced in the current models.

ACKNOWLEDGEMENTS

The authors would like to thank Karunya University for providing extensive guidance and support to carry out this research.

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