ISSN: 2319-5967
ISO 9001:2008 Certified
International Journal of Engineering Science and Innovative Technology (IJESIT)
Volume 4, Issue 3, May 2015

A study of Video delivery issues in mobile devices over Internet

Prachi Prakash Salvi

Abstract— Now a day's social networking and mobile communication has grown rapidly. Main reasons of this growth are high speed internet connection and various types of multimedia (text, images, videos, animation and sound).Meanwhile change in technology brings your desktop to small, portable device and thus mobile phones. All the work which you have done in your laptop or desktop is now possible to do with your mobile phone. In this paper, we have examined how multimedia content especially videos are stream and broadcast. How they compress and deliver to mobile users. What are challenges in streaming and communication of videos on mobile over internet?

***Key words:-*broadcast, multicast, video streaming, unicast.**

I. INTRODUCTION

Among the most exciting technologies developed in past few years is maturation of mobile communication. The internet on mobile devices offer most powerful and wide reaching new publishing medium since television, and may well surpass all previous media in its importance and scope. Users of mobile devices have different types of interest, preferences and different opinion and they usually like to store multimedia content with them, which is relevant to them. It is somewhat difficult to store and manage large volume of multimedia content in portable devices like mobile phones or tablets, with limited storage space.

A study shows that 91 percent of mobile users are engaged in social networking applications as compared to 79 percent of desktop users. A study also shows that on average, people spend 2.7 hours per day on mobile devices for sharing photos, to post a comment, to connect with friends and many more activities on social networking sites, making it an increasingly favorable platform for socializing.

Mobile devices were firstly used as a communication device, but now days they are used as a social learning platform. These devices are majorly used for capturing multimedia content (text, images, audio, and videos) for various activities. In academia and education, mobile devices are used for mobile social learning. During events such as conferences and workshops, mobile devices can be used as an event guides for the planner, organizers and attendees. In other activities which involve context-aware services such as entertainment, shopping, tourism and location-based activities, mobile devices play different and important roles for individual users.

Most of the applications over internet are multimedia oriented. Applications like live TV, gaming and video conferences distance learning is take place over internet and with use of multimedia. These applications use video, audio, text and images for communication and data traffic.

Video is an important part of communication and entertainment also it has been powerful tool for communication and learning. Multimedia video streaming and broadcasting is one of the most challenging issues in multimedia communication. Multimedia streaming is one of the most challenging issues in multimedia communication.

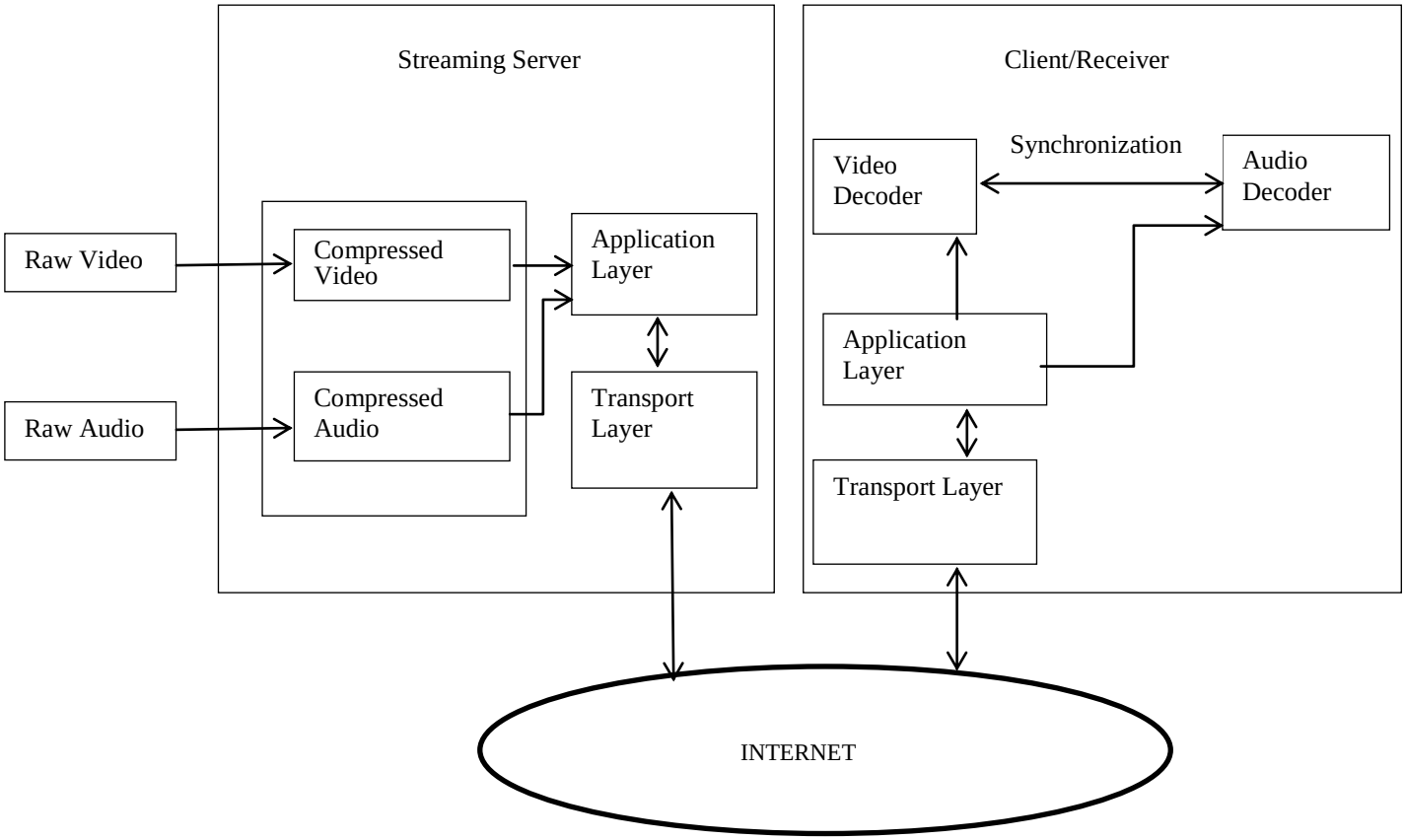
II. ARCHITECTURE OF VIDEO

Initially videos were transmitted in analog form. The digital video enables a revolution in the compression and communication of video. The growth and popularity of the Internet motivated video communication over best-effort packet networks. Video is complicated by various factors including varying bandwidth, delay, and losses, also many additional issues like how efficiently perform one-to-many communication for popular content

Below figure [1] shows video streaming architecture. Raw audio and video are compressed by compression algorithms and stored in storage device. On request of client video is retrieve from streaming server in the form of packets and send it to Internet through application and transport layer. The packets may be loss or delayed due to congestion inside the Internet. The packets which are delivered to client's server firstly pass through transport layer and then proceeds to application layer. After this audio and video decoded at decoder and audio and video is synchronized with use of media synchronization.

ISSN: 2319-5967
ISO 9001:2008 Certified
International Journal of Engineering Science and Innovative Technology (IJESIT)
Volume 4, Issue 3, May 2015

There exist various ways of video communication and streaming applications, which have different operating conditions. The communication applications are may be unicast, multicast, broadcast. Let us briefly discuss these properties.



III. UNICAST, MULTICAST AND BROADCAST AND ANYCAST COMMUNICATION

A common form of communication is point-to-point or one-to-one communication. It is used when two nodes need to talk with each other. These communications have special property that is there may be a back channel exists between receiver and sender. If a back channel exits, the feedback can be provided by receiver to sender. Sender uses this feedback to adapt its processing. This type of communication wastes bandwidth by sending multiple copies of data.

The most popular video communication is one-to-all communication or broadcast communication and best well known example is television broadcast. Broadcast becomes a viable solution if all of the nodes are on same subnet. All nodes on the subnet will see all traffic. Broadcast is a layer 2 feature in the Ethernet protocol, and also layer 3 feature in IPV4. [2]

In broadcast bandwidth is wasted by sending data to whole network. Also it slows the performance of client's machine. The main challenge for broadcasting is scalability issue.

Another form of communication which lies between point-to-point and broadcast communication is multicast communication. Multicast is one-to-many communication it is not like broadcast that is one-to-all communication. Multicast is like broadcast but not touches to all nodes like broadcast. Nodes have to subscribe to multicast group to receive the information. Nodes transmitting data to a multicast group do not know what nodes are receiving. Multicast protocols are usually UDP protocols.

IV. CHALLENGES IN VIDEO STREAMING

In this section we will discuss main challenges in video streaming. Mainly three problems in video streaming are discuss in this section. Video download is similar to a file download, but video is not a generic file. So video

ISSN: 2319-5967
ISO 9001:2008 Certified
International Journal of Engineering Science and Innovative Technology (IJESIT)
Volume 4, Issue 3, May 2015

download is like a file download, but a large file. However it has number of disadvantages. Generally video download requires long time and large storage space.

Video delivery via Streaming

The basic idea of delivering a video streaming includes split video into parts, these parts are then transmitted and receiver when receive these part enable him to decode and playback, without having delay for entire video to delivered. Video streaming can consist of following steps:

1. Packets containing compressed video.
2. Start delivery of packets.
3. Start decoding and playback while video is still being delivered at the receiver.

In video streaming delivering a video and playback it are simultaneous tasks. In streaming a short delay is there also called pre-roll delay and usually it is in the order of 5-15 seconds. This delay is between the start of delivery and beginning of playback at receiver's end. The length of pre-roll delay is based on time duration of pre-roll buffer and storage is given by amount of data in pre-roll buffer.

1. Basic Problems in Video Streaming:

There are number of problems that affect streaming of video. Internet offers an only best effort service that's why video streaming is difficult over internet. This provides no guarantees on bandwidth, delay, and loss rate also these characteristics are unknown and dynamic. When dealing with these characteristics the designing of system to reliably deliver high-quality video over the internet is the main aim of video streaming.

a) Bandwidth

The bandwidth available in between is usually unknown and time-varying. If available bandwidth is slow and sender transmits faster than available bandwidth then congestion can occur, packets are lost and video quality may be differ. If sender transmits slower than available bandwidth then receiver produces good video quality but not best. To overcome this problem first available bandwidth should be determine and then math the transmitted video bandwidth to available bandwidth.

b) Delay

The delay that experiences by packet may varies from packet to packet. This delay is referred as delay jitter. Jerk in video is a good example of delay jitter. This happens because receiver must be receive/decode/display frames at constant rate, so any late frame can produce problem in video.

c) Loss Rate

The third problem is losses. Depending on network condition number of different types of losses may occur. For example, in wired networks like Internet are affected by packet loss, while in wireless channels are affected by bit errors or burst errors. Losses may cause destructive effect on video quality. A video streaming is designed with error control mechanism, these error control techniques are roughly grouped into following classes:

A. Forward Error Correction (FEC)
B. Retransmission
C. Error Concealment
D. Error Resilient Video Coding

These three fundamental problems of bandwidth, delay jitter and loss rate are discuss detailed in following sections.

A. Overcoming Techniques

1. Rate control for Bandwidth:

A common problem that occurs in communication channel is congestion. Other symptoms of congestion are packet losses and delay jitter may represent challenges to streaming multimedia. Control techniques may also employ to avoid symptoms of congestion and to limit the network load. This technique is also called as congestion control or rate control.

i. Streaming media rate control:

To determine appropriate rate of transmission is a difficult task. But rate control mechanism is implemented in Transmission Control Protocol. TCP is used to deliver web-pages, email and some streaming multimedia content. A simple rule is used in TCP fro rate control and that is "Additive Increase Multiplicative Decrease" rule. This AIMD rule is used for TCP congestion avoidance. AIMD combines linear growth of the congestion window with an exponential reduction when congestion takes place. [3]

ISSN: 2319-5967
ISO 9001:2008 Certified
International Journal of Engineering Science and Innovative Technology (IJESIT)
Volume 4, Issue 3, May 2015

Mathematical formula:

Let w(t) be sending rate during time slot t, a(a>0) be the additive increase parameter and b(0<b<1) be multiplicative decrease factor. [4]

$$w(t+1) = \begin{cases} w(t)+a & \text{if congestion is not detected} \\ w(t)*b & \text{if congestion is detected} \end{cases}$$

From above formula we can say that when there is no congestion, packet transmission is increased at constant rate (additive rule) and when there is congestion is noticed, packet transmission rate is halved.

ii. Streaming media over TCP:

There are number of advantages of using TCP for streaming media objects. First, TCP rate controls have optimal scalability and stability. Second, provides guarantee of packet delivery, with elimination of losses effectively. The difficulties for streaming media with use of TCP include following. First, it gives rise to long delivery time due to packet delivery guarantee. Second, the "Additive Increase Multiplicative Decrease" rule generates saw-tooth pattern which is not suitable for media transport.

iii. Streaming over UDP:

Retransmission and rate control mechanism of TCP are not useful for streaming media. User Datagram Protocol (UDP) allows more flexibility in terms of error control and rate control. It is more suitable to implement error control technique instead of only relay on retransmission. For rate control AIMD generates varying throughput but also gives scalability and stability. It has been observed that average throughput of TCP is differ from end-to-end measurements, this give rise to TCP-friendly rate control technique. It has more scalability and stability than ordinary rate control mechanism.

2. Overcoming on delay jitter:

It is very common to have 5 to 15 second buffering for streaming of media before actual playback begin. Buffering provides number of advantages over best-effort networks. These advantages are as follows:

1. Reduction of jitter: Jitter causes jerk in playback of video. It causes due to variation in network and time taken to travel between end hosts. To avoid delay in streaming buffering is usually used. Use of buffers effectively avoids delay and eliminates jerkiness in streaming media.
2. Error recovery: A retransmission is take place when packets are lost in the network. Media stream objects are sensitive to errors. Recovery of losses improves quality of streaming.
3. Error resilience: If losses are isolated instead of concentrated then some loose in streaming media are better to be concealed. Interleaving is allowed by buffering to transform burst loss into isolated losses and enhancing concealment of concentrated losses.

3. Error Control:

The third fundamental problem related with video communication is losses. Losses can lead to very destructive effect on video quality. There are different types of losses depending on network condition. It is assumes that video decoder expected to receive packets without any error or it may receiver packets with error. The losses can vary depending on network condition and also amount of cross traffic. There are four types of error correction.

(1)Forward Error Correction (2) Retransmission
(3)Error Concealment (4) Error Resilient Video Coding

1. Forward Error Correction:

Forward error correction is mainly used for recovery from losses. The core idea of FEC is sender encodes the media in redundant way by using an error-correction code (ECC).The redundancy allows the receiver to detect a limited number of errors, and often correct these errors without retransmission. FEC gives receiver to correct error without need of a reverse channel. [5] FEC provides number of advantages and disadvantages. FEC does not require back channel, which require in retransmission. It also may provide little delay. Disadvantages of FEC are overhead even when there is no loss. If the amount of loss is less than threshold then loss can be recover from lossy data, if amount of loss is greater than threshold then it became difficult to recover loss. Only some portion of lost data can be recover.

ISSN: 2319-5967
ISO 9001:2008 Certified
International Journal of Engineering Science and Innovative Technology (IJESIT)
Volume 4, Issue 3, May 2015

2. Retransmission:

In retransmission approach receiver uses back-channel to intimate sender that he has receive packet correctly. This approach wisely uses available bandwidth, and only resends the packets which are lost. It is adaptive to changing channel conditions. Retransmission has some disadvantages. In retransmission an additional delay is present. For proper execution of retransmission a back-channel is required, which may not be present in many applications. If back-channel or delay is not there then retransmission is not suitable.

3. Error Concealment and Error Resilient Video Coding:

Error concealment is used to estimate lost information in order to conceal error wherever it has occurred. In other hand aim of error resilient video coding is resilient specific error by designing video compression algorithm and compressed bit stream.

V. ADDITIONAL CHALLENGES DUE TO MOBILE ENVIRONMENT

Mobile devices often change their location; change their base station with which they are communicating. Changing location while streaming may cause break in transmission or may cause packet loss or delay.

VI. CONCLUSIONS AND FUTURE ENHANCEMENT

In this paper we have highlighted the problems related with video streaming and broadcasting in mobile devices. We discuss the technologies related to overcoming streaming issues. As the use of mobile devices with multimedia objects are increasing day by day, therefore considering the impact for next generation requirement is important. The AIMD rule for TCP streaming avoids congestion so in future we are looking forward to study on AIMD rule for congestion avoidance and making streaming faster. Also we are concentrating on error control techniques.

REFERENCES

[1] Google search (Video Streaming architecture): www.cs.ubc.ca Dapeng Wu, Yiwei Hou et.al. "Streaming Video over the Internet: Approaches and Directions" 24 January 2004.

[2] Google search (Difference between unicast, multicast, and broadcast):http://serverfault.com/questions/279482/what-is-the-difference-between-unicast-anycast-broadcast-and-multicast-traffic.

[3] Wikipedia- Additive Increase Multiplicative Decrease (AIMD).

[4] Wikipedia- Forward Error Correction (FEC).

[5] Andrea Broughton, Tom Higgins, Ben Hicks and Annette Cox (The Institute for Employment Studies) "Workplaces and Social Networking The Implications for Employment Relations"2010.

[6] MICHAEL WYNBLATT ,DAN BENSON , ARDING HSU Siemens Corporate Research Princeton, New Jersey, FELIX BRETSHNEIDER,LARRYSCHESSEL, GRAHAM HOWARD "Multimedia Meets the Internet: Present and Future".

[7] Hatem BETTAHAR Heudiasyc, Universities de Technologie de Compiègne, France: "Tutorial on Multicast Video Streaming Techniques".

[8] Editor: Harrick VIN University of Texas at Austin: Multimedia Broadcasting over the Internet: Part I.

AUTHOR BIOGRAPHY

Prachi Prakash Salvi currently pursuing Master of Computer Application from Finolex Academy of Management and Technology, Ratnagiri.