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ASSESSMENT OF QUALITY-OF-EXPERIENCE FOR VIDEO STREAMING OVER LTE NETWORK

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***Abstract*—3G and 4G technologies development has dramatically increasing the mobile internet in the recent years. The devices like laptops, cellular mobiles and tablets using the mobile broadband internet like skyrocketed. The most popular mobile application is the video streaming in the application of multimedia. In a cost effective way, the big challenge is the quality to make available these services to users. The above task is achievable by means of developing the LTE (Long Term Evolution) in the world of mobile. With low latency and high data rates in the applications of multimedia the effective services is provided by the LTE technology features.**

In this paper, we study and analyze the Quality of Experience (QoE) at the end user for Video on Demand (VoD) over the LTE network. To achieve this, we streamed High Definition (HD) videos based on H.264/AVC and these videos are delivered from source to destination using Transport Control Protocol (TCP) and User Datagram Protocol (UDP). Specifically, our study is about QoE evaluation in terms of delay variation, packet loss metrics and provides performance evaluation to characterize the impact of transport layer protocol in video streaming over radio networks like LTE.

Keywords- QoE, Video Streaming, H.264/AVC, LTE, Packet loss.

I. INTRODUCTION

For the last two decades radio access technologies are not just limited to provide voice communications alone, but also used for the video and data applications as well. Due to the rapid development of technology used in telecommunication systems and consumer electronics, network operators are now able to provide better Internet services over radio networks. After the development of 2G and 3G technologies, the Internet based services are available on mobile systems, namely mobile broadband.

According to CISCO report, mobile video has been growing at a Com- pound Annual Growth Rate (CAGR) of 75 percentages between 2012 and 2017, and it is the highest growth rate of any other mobile application [1]. Meeting this demand, maintaining the Quality of Service and user satisfaction has become a big challenge to network operators. To achieve this radio interface is needed, which can provide the best Quality of Services with design parameters like Data rates, Delay and Capacity.

One of the main reasons for LTE evolution is to provide the IP based services to people on mobile devices with better QoS. Some services have been already provided by 3G networks, but providing HD video streaming, interactive video gaming and other multimedia services without degrading the Quality of Services is a major challenge. Among these multimedia services, video streaming over mobile Internet is the most popular one. In the burst growth of data rates and services, being aware of user experience is important to maintain the service quality and the application performance [2].

The Quality of Experience is defined as “The process of understanding the actual performance of services, as delivered to the customer, for the purpose of ensuring those services meet customer expectations and requirements". Understanding user experience is very critical for the network operators in managing the QoS of the network. Quality of Experience measurements are made at the point of delivery directly from the subscriber's smart phone or PC. QoE measurements deals with how well applications (Video streaming, VOIP and Web browsing) work in the hands of subscriber [3]. QoE measurements require an understanding of the Key Performance Indicators (KPI) that impact on the user perception. KPI's vary with the service type and services like VoIP, Video streaming, On-line gaming, Internet browsing has unique performance indicators to measure. QoE considers the individual subscriber experience with a service unlike network conditions in QoS. The knowledge of user actual experience is very important for the operators to know the customers satisfactory levels and then operators can concentrate on issues to prevent the churn.

The mobile communication technologies are tracked back to various generations 1G, 2G, 3G and 4G. 1G which started in 1980 stands for first generation of wireless telecommunications popularly known as Cellular phones, in which analog radio signals are used. In 1991 2G (Second Generation) wireless telephone technology started using digital mobile systems. The second generation mobile technologies provide low bandwidth services, which are suitable for voice traffic. The packet data over cellular systems started with the introduction of GPRS (General Packet Radio Services) in GSM (Global System for Mobile Communications). With the introduction of 3G technology, network operators can provide better and more advanced services like video calls and mobile broadband services [4].

In this paper, we worked on user quality assessment for video streaming over LTE network. We report on user perception of video quality degradation of selected videos, which are subjected to different delays and packet losses. This paper is specifically aimed to understand the experience of High Definition videos with H.264/AVC. We report the user experience by conducting Subjective Assessment as per the International Telecommunications Union (ITU).

LTE is the latest technology in the telecommunications world using which network operators are able to provide advanced multimedia applications to users maintaining Quality of service. By the introduction of LTE network, users are able to enjoy the broad- band quality Internet services on the mobile devices, but providing the advanced multimedia services over radio network without degrading the Quality of Services is a big challenge. Video streaming is the most popular application in the next generation mobile systems [5]. Due to the rapid increase in data rates using LTE technology, users are able to watch High Definition videos on the mobile devices and at the same time the mobile systems are being manufactured to access advanced services. In this current day of huge resource demand applications, understanding the user experience is very critical for the network operators to manage the QoS of the network and hence our motivation to measure and analyze it.

We choose High Definition videos with 1280 _ 720p and H.264/AVC, since H.264 codec is the widely used codec for video streaming. As compared to the previous codec like MPEG-2 and MPEG-4 Visual, AVC provides good quality of video for wide range of services like broadcast multimedia streaming and Video on Demand [6].

In this paper we analyzed the perception of users towards the videos encoded with H.264 baseline profile in laptops and mobile devices. The videos are streamed through an emulated network with packet losses and packet delay variations. The obtained results from both devices are compared using matched-sample-test. The conclusion infers that the device does not show any impact on user perception for videos of same resolution.

II. BAC KGROUND STUDY

A. *Quality of Experience*

QoE is also defined as “The overall acceptability of an application or service, as perceived subjectively by the end user". It mainly deals with how an individual is satisfied with the provided service in terms of usability, accessibility, retain ability and integrity of the service. Quality of Experience considers the complete end-to-end system effects like, the effects of the client, network and infrastructure of the services [3] [7].

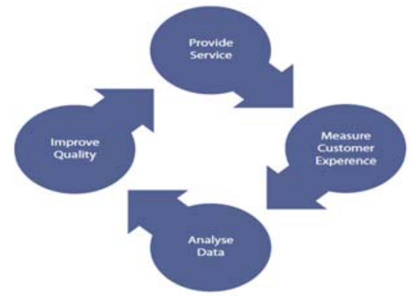

Figure 1: Quality of Experience Measurement

B. *Video Streaming*

Today video sharing is the most effective way of entertainment and gaining knowledge. In order to share a video it must be stored and transmitted over a communication channel, but it is expensive to transmit a raw video over a communication channel because of the huge amount of data size. Even to store a raw video, it requires a lot of space on the data storing devices. Usually the video taken from camera footage contains lot of redundant data [8]. So, there is a need to reduce the size of the redundant data in the raw video also considering the quality of the video. Here video compression comes into the picture, to reduce the redundant data by considering the quality of the video.

C. *Video Compression*

Video Compression is a technique where the raw video is compressed using mathematical models and algorithms to reduce the size of the video to lower bit rates. Video compression is an essential process for video applications like Internet video streaming, mobile TV, video conferencing, digital television and DVD-Video. Video compression is mainly classified into two types lossless compression and lossy compression. In lossless compression no information is lost in the compression process. So it is possible to recover the original raw video from a compressed video. In practical cases, the amount of data reduced is less. Quality of the video is maintained well in lossless compression [9]. This type of video compression is not used for streaming videos because even though video is compressed, it still maintains a large data size. In lossy compression, as the name suggest information is lost in compres- sion process. The information once lost cannot be retrieved.

D. *H.264/AVC Code*

H.264 is a method and format for video compression. It was developed based on the concepts of earlier standards such as

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MPEG-2 and MPEG-4 Visual and provides better compression efficiency [10]. It also provides features like better-quality compressed video and greater flexibility in transmitting and storing videos. Furthermore, it offers robust compression for a wide range of applications, from low bit rate mobile video applications to high definition broadcast services.

E. Supported Protocols for Video Streaming

There are several protocols that support media streaming. Some of the major protocols are Hypertext Transfer Protocol (HTTP), Session Description Protocol (SDP), Real-time Streaming Protocol (RTSP), Real-time Trans- port Protocol (RTP), Real Data Transport (RDT) and Real-time Transport Control Protocol (RTCP). Each protocol is used depending on the type of application in that particular point and also depends upon the requirement of service. For most of the video streaming protocols, TCP and UDP serves as the underlying protocol. UDP is a preferable to its contrary protocol TCP in video streaming application because UDP send packets at a constant rate and it doesn't care about the lost packets. But the packet loss due to network congestion is avoided in TCP protocol. On the same hand, TCP introduces delay due to the retransmission of data whenever data is lost. But in case of UDP, there is no point of retransmission [10].

F. LTE (Long Term Evolution)

The term LTE includes the development of the Universal Mobile Telecommunications System (UMTS) radio access through the Evolved UTRAN (E- UTRAN). It is mainly accomplished by the evolution of core network known as System Architecture Evolution (SAE). LTE is able to support different types of services including HD video streaming, VoIP, Multi user online gaming, Video on demand, Push-to talk and Push-to-view [5].

G. Architecture of LTE

The enhancement features like higher packet data rates and significantly lower-latency of LTE cannot be possible without the evolution of System Architecture Evolution (SAE). This includes the Evolved Packet Core (EPC) network. The Evolved Packet System (EPS) consists of LTE and SAE. It was decided to have a "Flat Architecture". EPS is defined to support only packet-switched traffic. It uses the concept of EPS bearers to direct the IP traffic from a gateway in the Packet Data Network (PDN) to the User Equipment (UE). EPS bearer is a virtual connection provides transport service with specific QoS attributes between the gateway and the UE [11]. Likewise the HSPA architecture the LTE architecture also divided into two networks as radio access network and a core network. However the ultimate goal of the LTE is to minimize the number of nodes. As a result of this, the Radio Access Network (RAN) contains only one node.

III. RESEARCH METHODOLOGY

This section describes the hardware utilities and software tools that we used for conducting the experiment. The section consists of explanation of experimental setup for the video quality assessment over LTE network using Subjective analysis.

A. Video Quality Assessment Using Subjective Analysis

In video quality assessment section, two types of videos are taken and they are streamed over a live LTE network with introducing different delays and packet losses. The streamed videos are saved for video quality assessment.

B. Experimental Setup and Procedure

The experiment is carried out using the experimental setup shown in Figure. It contains the server, which is operating on Ubuntu 12.04, with AMD Athlon processor, 2 GB RAM and it is connected to the BTH network. The gateway is operating on Ubuntu 12.04, with AMD Athlon processor, 2 GB RAM and it is connected to LTE network via USB modem. The client runs on Ubuntu 12.04, with AMD Athlon processor and 2 GB RAM. It is connected to gateway via Ethernet full duplex link, bandwidth of 100 Mbps. The client system receives packets from server through LTE network via gateway. The Measurement point is connected in between the sender and the client. The MP consisting of two DAG cards, one of them is connected to capture the traffic at the server side and the other DAG card is used to capture the traffic at the client side. The traces are collected and saved in the consumer system. The Wowza Media Server software is installed in the server system and the videos are encoded to the required format and placed in the server. In this paper, the video streaming used is Video on Demand. Before streaming the video, required delay and packet loss settings are made at traffic shaper. Videos are streamed from server to client using TCP and UDP protocols as per the request of client. These streamed videos are saved in the client system. The saved videos are further used to get the Mean Opinion Score from the users.

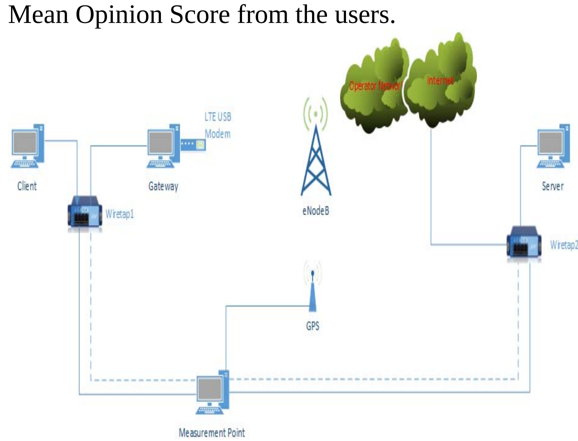

Figure 2: Experimental Set up for Video Streaming

C. WOWZA Media Server

Wowza Media Server is a software used to stream video and audio files over public and private networks. It can stream Live video streaming, Video on Demand and Video recording over Adobe Flash player, Microsoft Silverlight player, Apple iPhone, iPad, iPod touch, Quick Time player, Smart phones devices, tablets and IPTV set-top boxes. The supporting protocols of Wowza Media Server are Real Time Messaging Protocol (RTMP), Microsoft Smooth Streaming, Apple HTTP Live Streaming (HLS), Real Time Streaming Protocol (RTSP), Adobe HTTP Dynamic Streaming (HDS), Real-time Transport Protocol (RTP) and MPEG-2 Transport Streams (MPEG-TS). It is an alternative to Adobe Media Server, Darwin Streaming Server, Microsoft IIS Media Services, and other media servers. The encoded videos are saved in the server machine with a specific name. These videos are accessed from client system using VLC player and also web browser having flash player with provided names in the client machine as per the requirement [12].

D. Test Video Parameters

We selected two types of videos for our experimentation. One is a Rugby game video, which comes under the fast moving video and the other is Big Buck Bunny animated video, which comes under slow moving video. We covered the two types of videos suggested by ITU-T. With the availability of high speed data networks, users are more attracted to view high definition videos. There is a scope in this direction, to test the high definition videos over high speed data networks. So, we used 720p High Definition videos with resolution 1280 _ 720 pixels and is coded with a Main Profile encoder at Level 3.1. These two raw videos used for video quality testing are taken from test media website. These two videos are encoded to the required format using FFmpeg encoder. The duration of selected video sequence is about 4 minutes. In this, performance of 3G data services was observed by taking packet data streams for 5 minutes. To observe the user QoE for video streaming over radio network, a minimum 4 minutes length of video is considered as appropriate. We chose the .mp4 as a container, since it is supported by most of the latest smart phones based on Android and iPhone. We selected H.264/AVC as video codec [6], because it has more advantages than MPEG-2 and MPEG-4 Visual in better image quality at same compressed bit rate. H.264 offers greater flexibility in terms of transmission support and compression options

Video sequences	Rugby , Big Bucks Bunny
Codec	Perceptible, H.264 Main Pro_le, Level 3.1
Resolution	720p, (1280 _ 720)
Frame Rate	25fps
Container	MP4

Table 1: Test Video Parameters

E. NetEn

NetEm is a network emulator on Ubuntu that provides network conditions by adding delay, packet dropping and duplicating packets. The main motivation to use NetEm is, it provides long distance network scenarios in the lab environment [13]. In this paper we showed that performance of NetEm is more reliable as compared to NIST Net and KauNet.. The NetEm delivers each packet that flows through it with certain delay that should be in the delay range given to it. In the case of packet loss the amount of required packet loss must be in percentage form. It drops some packets randomly as per the given loss percentage before they are queued.

F. Delay

To choose the packet delay variation values range, some set of laboratory tests were performed for final assessment. Constant delay values were introduced from 100 ms to 500 ms and we found significant changes at 150 ms of constant delay. In case of delay variations, we introduced D from 0 to 50 ms with an increment of 5 ms and we could not find significant changes. We repeated the test with values from 0, 10, 25, 50 and 100. We observed significant changes in the range of constant delay 150 _ f0, 10, 25, 50, 100g and similar settings are used.

G. Packet Loss

One or more packets that originate from the source being transmitted across the network, fail to reach the destination. This shows major impact on the performance of the network and causes significant problems in applications like Video conference, VoIP and video streaming. The packet loss is measured in percentage of packets lost from the overall transmitted packets. To choose the packet loss variation values range, some set of laboratory tests were performed similarly for delay. Significant variations in video were observed for values 2%, 4%, 6% and 8% and similar settings are used.

IV. MEASUREMENTS

The aim of our experimentation is to calculate the packet loss over LTE network.

A. Packet Loss (PL)

The percentage of packet loss is calculated as the ratio of the number of packet lost (L) to the total number of packets sent by the sender (N). The number of lost packets is the difference in the number of packets received by receiver and the total number of packets sent by sender with unmatched sequence numbers.

$PL = L/N$

V. RESULT AND ANALYSIS

As increasing number of users watching HD videos over mobile networks, the facts provoked us to find the QoE of video streaming over mobile network like LTE. The main idea behind this paper, to present QoE of video streaming over the

LTE network. The mean is defined as the average and it is computed as the sum of all the observed outcomes from the collected samples divided by the total number of events.

The mean is defined as:

$$U_{jk} = \frac{1}{N}\sum_{i=1}^{N} U_{ijk}$$

The SD is defined as:

$$S_{jk} = \sqrt{\sum_{i=1}^{N}\frac{(Ujk - Uijk)2}{N-1}}$$

Where: N is the number of observers, Uijk is the score of ith observer for test condition j, video sequence k.

1) Packet Delay Variation

a) Packet Delay Variation for TCP

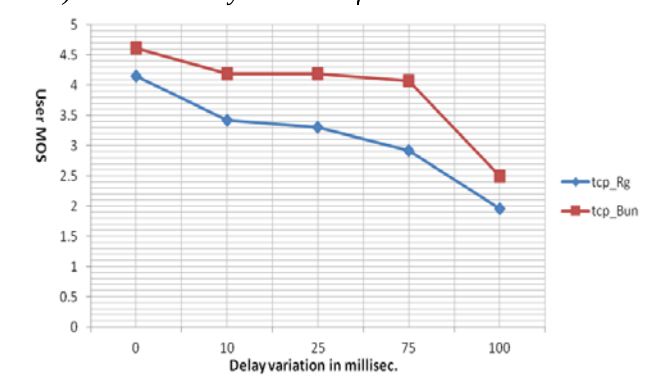

Figure 3: MOS for TCP videos subjected to Delay Variation

b) Packet Delay Variation for UDP

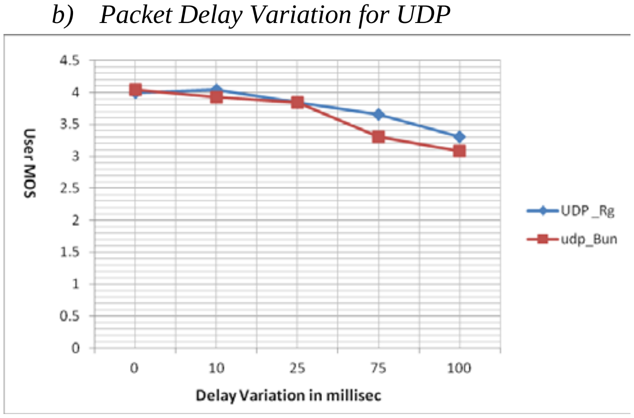

Figure 4: MOS for UDP videos subjected to Delay Variation

c) Standard Deviation for Delay Variation

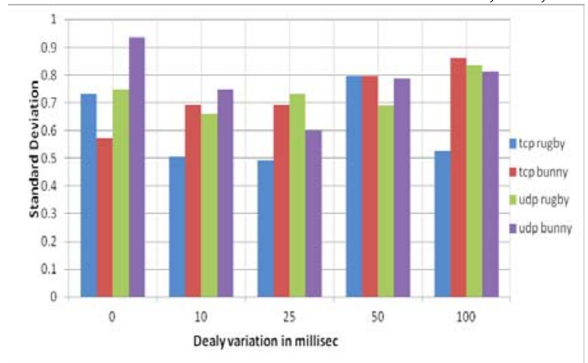

Figure 5:Standard Deviation for Delay Variation

d) Confidence Level for Delay Variation

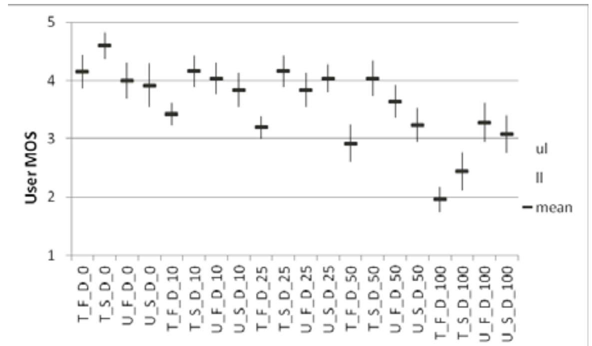

Figure 6:Confidence Interval for Delay Variation

The 95% confidence interval for the packet delay variations are calculated and the graph is plotted. As shown above, is the 95% confidence interval graph plotted for both types of videos and for both the protocols too. The graph was plotted by taking MOS on Y-axis and packet delay variation of the video sequences on X-axis. From the graph it is observed that 95% confidence interval is high for videos of TCP in all packet delay variations except the packet delay variation at 100 ms.

2) Packet Loss

The MOS rating of packet loss is taken for five successively increasing values. The MOS rating are for fast and slow moving videos in both the TCP and UDP protocol.

a) Packet Loss for TCP

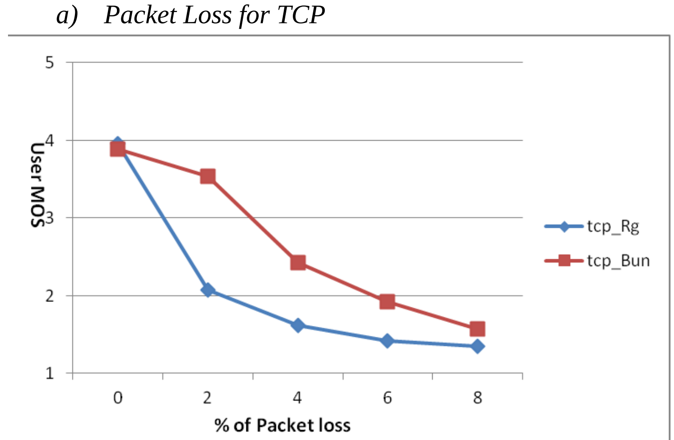

Figure 7: MOS for TCP videos subjected to Packet loss

b) Packet Loss for UDP

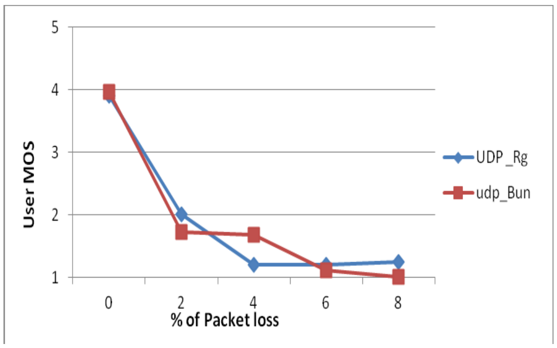

Figure 8: MOS for UDP videos subjected to Packet Loss

c) Standard Deviation for Packet Loss

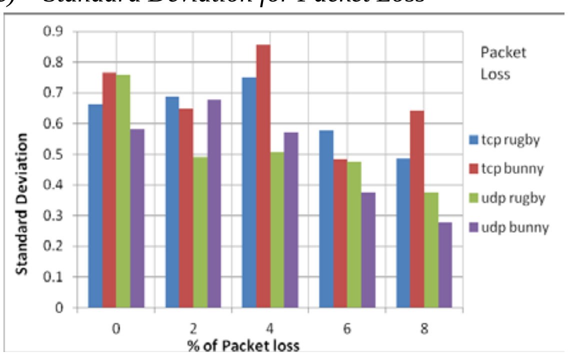

Figure 9:Standard Deviation for Packet Loss

d) Confidence Interval for Packet Loss

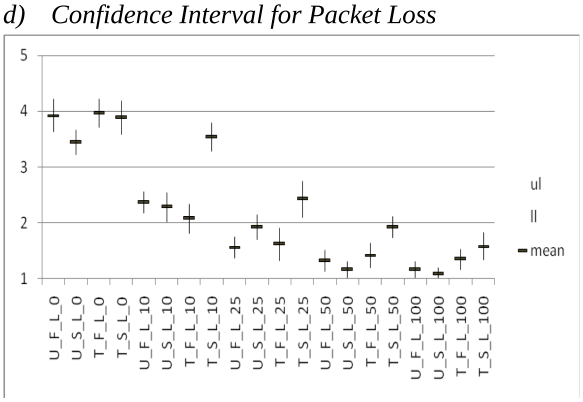

Figure 10:Confidence Interval for Packet Loss

The 95% confidence interval for the packet loss percentage was calculated and the graph plotted. As shown above, this is the 95% confidence interval graph plotted for both types of videos and for both the protocols too. The graph was plotted by taking MOS on Y-axis and packet loss percentage of the video sequences on X-axis. From the graph, it is observed that as the packet loss percentage increases, the 95% confidence interval drops close to the bad video quality level.

VI. CONCLUSION

This paper work is based on the performance of video streaming over live LTE network.. The work deals with streaming video from server to client with emulated packet loss and packet delay in the setup. The raw videos are encoded into the H.264 Mainline profile with the help of FFmpeg encoder. The encoded videos are streamed through emulated packet delay variation in NetEm traffic shaper from server to client, on client request. Like packet delay variations same process is repeated for packet loss too and the videos are saved for further subjective analysis from the users. Statistics like Mean, Standard Deviation (SD) and 95% Confidence Interval (CI) is calculated for both fast and slow moving videos of TCP and UDP protocols. From the results of packet loss percentage variations it was observed that a drastic fall in the quality of fast moving video of TCP protocol even at 2% packet loss and user considered this video as annoying. But the slow moving video of TCP protocols is better sustained compared to the fast moving video. Both videos are considered as very annoying from 6% packet loss. The video quality is even more degraded for both the videos in UDP protocol. The quality of video is degraded drastically from slightly annoying to very annoying at 2% packet loss. The quality went even worse and it was hard to play as further increase in packet loss percentage.

The delay variation in the videos of TCP was observed and a slight fall in both fast moving and slow moving videos observed a change at delay variation 10 ms when compared to the no delay variation case. The video quality is maintained well even as the delay variation is increases up to 75 ms. At 100 ms a drastic fall in the video quality is observed and the user considered slow videos as annoying and fast moving videos as very annoying. But in the case of videos of UDP almost same video quality is maintained for both fast and slow moving videos and slight degradation of video quality is observed up to 25 ms. As further increase of packet delay variation there is no significant degradation of video quality observed for both the videos up to 100 ms. The user considered both videos as slightly annoying at 100 ms delay variation.

In this paper, the videos used for streaming is H.264/AVC and there is lot of scope to use video that is encoded to H.264/SVC scalable videos (SVC). Scalable video coding allows video conferencing devices to send and receive multi layered video streams. In those multi layered streams, a small base layer is present along with some optional layers and that help to improve resolution frame rate and in turn quality.

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