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User Centric Configuration using RTP/RTSP Video Streaming Protocol at Server Side

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Abstract— To deliver live streaming and video on demand to the client, server needs to perform some actions. The reference application to that is Real-time transport Protocol(RTP) which gives QoS mechanism for real-time delivery with content labeling (i.e.time synchronization).To communicate with server, client need to send request that comes under Real-time Streaming Protocol(RTSP) with some additional commands of playback,pause etc. At the same time control mechanism is also important for the continuous reports of congestion for that Real-time Transmission Control Process(RTCP).Hence provides quality data with less overhead of jitter delay, bandwidth utilization and can tolerate packet loss with traffic management. This paper presents the on live streaming and video on demand services provided by application with different codec formats to provide quality service to client.

Key words: RTP, RTSP, live streaming, codec formats, file reader

I. INTRODUCTION

In today's advances in computing technology high speed networks high bandwidth storage devices and compressed technology have made it viable to provide real time multimedia services over the internet. for instance, audio and video must be played continuously or else its annoying for clients. the server takes care for continuous stream of data. The play out process will be paused if the data does not arrive on time. The real-time multimedia the transport of live video or stored video is the prime part. There are two mode of transmission of stored video over the internet and the download mode, a user download the entire video file and plays back the video file. Unlike streaming mode, the whole video is not Downloaded, only the parts are being played and the remaining parts are being received and decoded.

In this paper, we are concerned with video streaming, which refers to real time transmission of stored video and live streaming of quality data frames to the client with respective codec formats. To read the file from mp4 file, file reader will be used for that ffmpeg api is used. There are lot of issues regarding real time multimedia transport.These issues are solved under evaluation parameters with less overhead of packet loss, transport managment and jitter delay.

II. RELATED WORK

Raw video and audio data are pre-compressed and then saved in storage devices. Upon the client's request, a streaming server retrieves compressed video/audio data from storage devices and then the application-layer QoS control module adapts the video/audio bit-streams according to the network status and QoS requirements. After the adaptation, the transport protocols packetize the compressed bit-streams and send the video/audio packets to the Internet. Packets may be dropped or experience excessive delay inside the Internet due to congestion

To improve the quality of video/audio transmission, continuous media distribution services (e.g., caching) are deployed in the Internet. For packets that are successfully delivered to the receiver, they first pass through the transport layers and then are processed by the application layer before being decoded at the video/audio decoder. To achieve synchronization between video and audio presentations, media synchronization mechanisms are required. RTP is used in conjunction with the RTP Control Protocol (RTCP). While RTP carries the media streams (e.g., audio and video). RTCP is used to monitor transmission statistics and quality of service (Quos) and aids synchronization of multiple streams.RTP is designed for end-to-end, real-time, transfer of streaming media. The protocol provides facilities for jitter compensation and detection of out of sequence arrival in data, which are common during transmissions on an IP network.

RTP allows data transfer to multiple destinations through IP multicast. RTP is regarded as the primary standard for audio/video transport in IP networks and is used with an associate file and payload format. The RTP server sends data as requested by the client. The server will take data from the repository and delivers it to the client with particular and requested codec format. The server will give response to the RTSP server methods like play, pause teardown, setup etc.

III. LITERATURE SURVEY

Method used : Modified NOVA h.264/AVC Profile Video Decoder System. It supports H.264/AVC baseline decoding of QCIF resolution. The operation of the NOVA H.264 video decoder starts with system reset. After system reset, the BitStream buffer starts to fetch bit stream from Beha BitStream ram via 16bit width data bus (BitStream_buffer_input). Then after 4 clock cycles when half of the 128bit BitStream buffer is filled, it sends the bit stream to the following decoder via 16bit width data bus (BitStream buffer output). An automatically-refill mechanism is employed to refill the buffer if half of its stored bit stream (>=64bit) is consumed. The Synyhesis, map, place and route and trace was doing lattice diamond rule. But to support decoding of higher standards working frequency need to be increased.

Method used: NAL Units: It focuses on the middle layer of video telephone. We obtain the video data by calling the API of Android system and divide then into NAL units in another thred by detecting NAL unit head. Then send them over RTP. We should divide the H.264 data into NAL units and add RTP head at the front then sent them.
For each NAL unit, its size is different from one to another because there is different data in each NAL unit.

Method used: Transco ding Enabled Streaming Media Caching System. aim is to deliver contents to the client in appropriate format. they used the caching algorithm for the better use of storage. TeC (Transcoding-enabled Caching) system using various experiments, including simulations using the proxy trace derived from a real corporate media server log. The trace reveals some degree of heterogeneous access of media objects.

Method used: Retransmission of Packet Flow. Retransmission can reduce the effect of packet loss on video streaming by using a small number of retransmission packets. Flow control controls the sending rate based on the number of packets lost, so that the sending rate adheres to the available bandwidth. Retransmission and flow-control improves the performance of video streaming, and users can clearly appreciate the difference when viewing images.

Method used: Qos Control Mechanism: on the requirements of the real-time streaming media applications and the problems of insufficient bandwidth of the current Internet, they used the important non-uniform distribution features of streaming media data introduce the caching mechanism and adopt non-uniform QoS control strategy in the flow of scheduling, packet loss handling, error controlling, etc, to ensure the real-time performance of the important data. In order to prevent congestion, early warning is employed to reduce the sending rate before the true formation of congestion. Although this approach does not solve the fundamental problem of video communication..

Overview of various existing techniques and algorithms are described in following table. Also, advantages and disadvantages are included in the table.

Paper title	Mechanism/Algorithm	Advantages	Disadvantages
1.FPGA-Based H.264 Video Decoder in RTP Payload Format.	hH.264/AVC profile video decoder system	A design of the RTP Depacketizer was successfully done and a test bench file for the top module was created. Simulation using Cadence NC-Verilog produced an output video with better quality compared	Need more speed in frequency for better quality of data
2.Image Acquisition and Transmission in the Video telephone Based on Android	NNAL units	Implemented on both pc player and android player.	
3. Performing evaluation of transcoding enabled streaming media caching system.	Tec enabled caching	It provides better byte hit ratio and less startup latency than traditional caching system.	Can improve less the quality degradation.
4. Retransmission and flow control for a video real time transport protocol	Retransmission of packet flow	Retransmission can reduce the effect of packet loss on video streaming by using a small number of retransmission packets.	Can give large no. of retransmission packets and have more good user experience.
5. The system construction and the implementation of Qos control mechanism in intelligent streaming media	Intelligent streaming media.	In order to prevent congestion, early warning is employed to reduce the sending rate before the true formation of congestion.	This approach does not solve the fundamental problem of video communication

Table 1

IV PROPOSED SYSTEM

Raw video and audio data are abridged and then saved in accumulator devices. Audio video data is recouped form the accumulator and process according to client's request or need. In the proposed system the server is being requested by the client with Vod and live streaming services. Once either of the process being processed, the file will be fetched from the file reader and that particular file added into session. The session is created prior the client server connection. Then it is forwarded and checked whether it is audio file or video file and then it will be passed through the encoder and particular file will be converted to requested client's format, and the methods are displayed on client displaying devices like PLAY,PAUSE , TOP,TEARDOWN, FAST FRAME LAST FRAME, STEPUP,STEPDOWN.

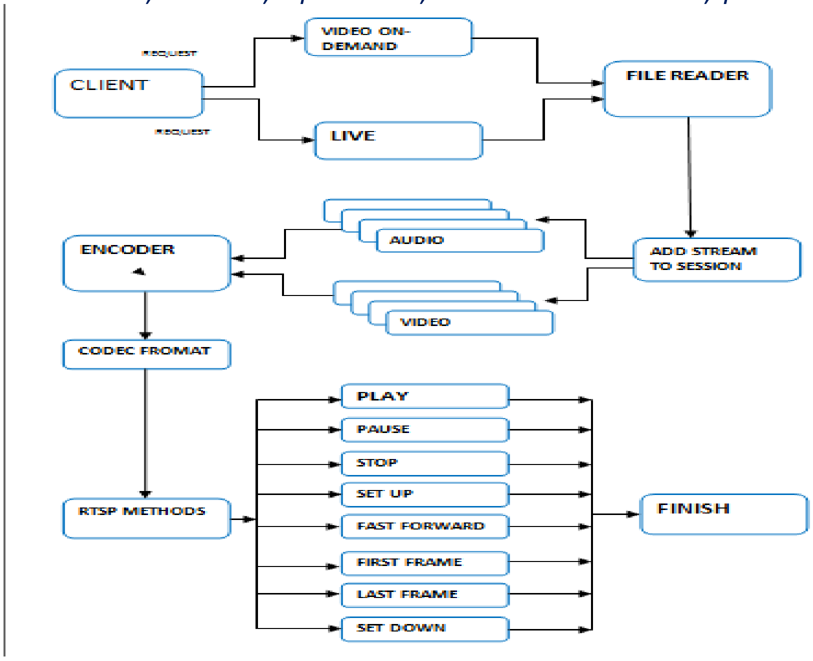

Fig 1: Proposed System

To liaise with server, client send request which falls under Real Time Streaming Protocol (RTSP). The control and reporting mechanism for status of ongoing process comes under Real Time Control Protocol(RTCP), and for transferring of frames form server to client the Real Time Protocol (RTP) is used. In fig 2 is the detail diagram of file reader. The file reader consist of program configuration ,it consist of bit rate, frame rate and resolution. It will do the work of open the file , find the stream and dump the file to the container to extract the file for further processes.

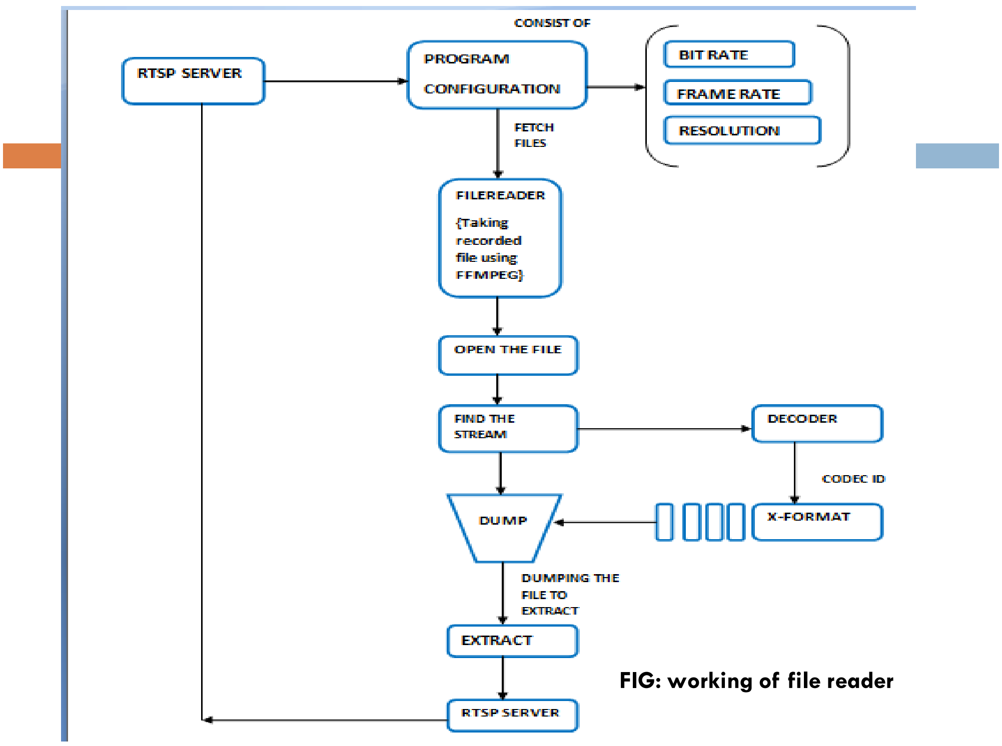

Fig 2 : Working Of File Reader

After that file reader fetch the files using FFMPEG api , the stream will be found as the decoder will understand the codec format with the for starting bits of header then the file is dumped for RTSP server to extract files to move forward. To deliver live streaming and video on demand to the client, server needs to perform some actions. The reference application to that is Real-time transport Protocol(RTP) which gives Qos mechanism for real-time delivery with content labeling(i.e. time synchronization). Hence provides quality data with less overhead of jitter delay, bandwidth utilization and can tolerate packet loss with traffic management. This paper presents the research on live streaming and video on demand services provided by application with different codec formats to provide quality service to client.

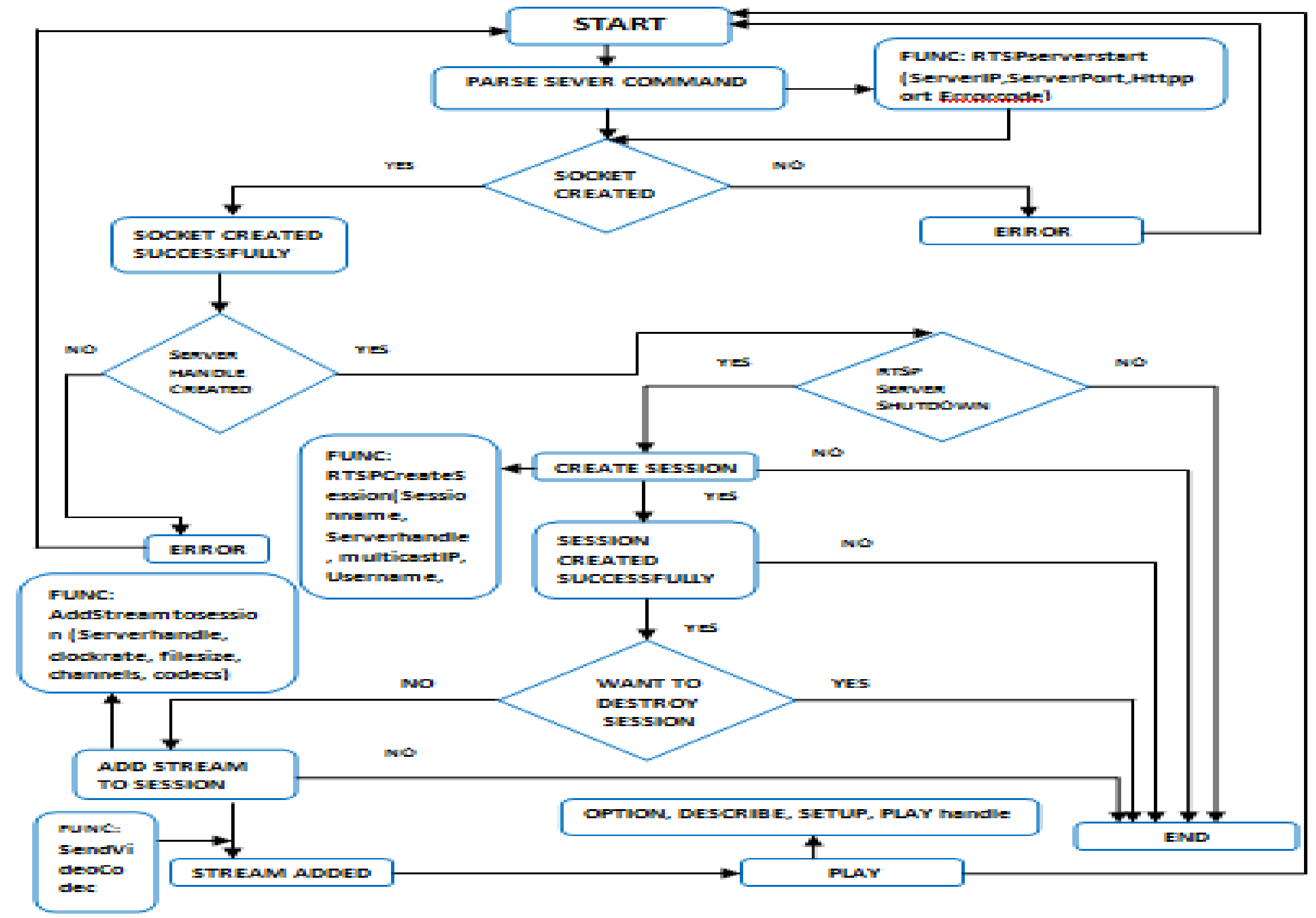

Fig 3: Server API Diagram

Creates a Server Instance with specified IP address and port number and create Socket of type TCP for RTSP communication with client returns Server Handle on Successful creation of Server"s Instance and Socket Returns Error Code in argument, in case of any Error.

In fig 3 RTSPServerShutdown it destroys all the sessions and Frees memory used by them In RTSPSessionCreate it creates a specified Session, Each Session should have at least one stream in it. So it is mandatory to call RTSPAddStreamToSession API after calling this API Returns Session Handle on Successful creation of Session"s Instance. In RTSPSessionDestroy it destroys a particular Session with the specified Session Handle and returns a proper error code, in case of any Error .In RTSPAddStreamToSession and RTSPDestroyClient it adds a stream to a particular session with specified Session Handle and destroys the client with specified client handle .This will be used only during VOD sessions when explicitly client is to be destroyed from application. This is optional API. And frees memory used by it ,Closes the sockets associated with that particular client and returns a proper error code, in case of any Error .

V SIMULATION RESULTS

a. We concluded that the with the help of optimized data collection we generated the following output for client server connection session is being created with clients IP address and Port number .After adding Stream to session the particular stream is added .

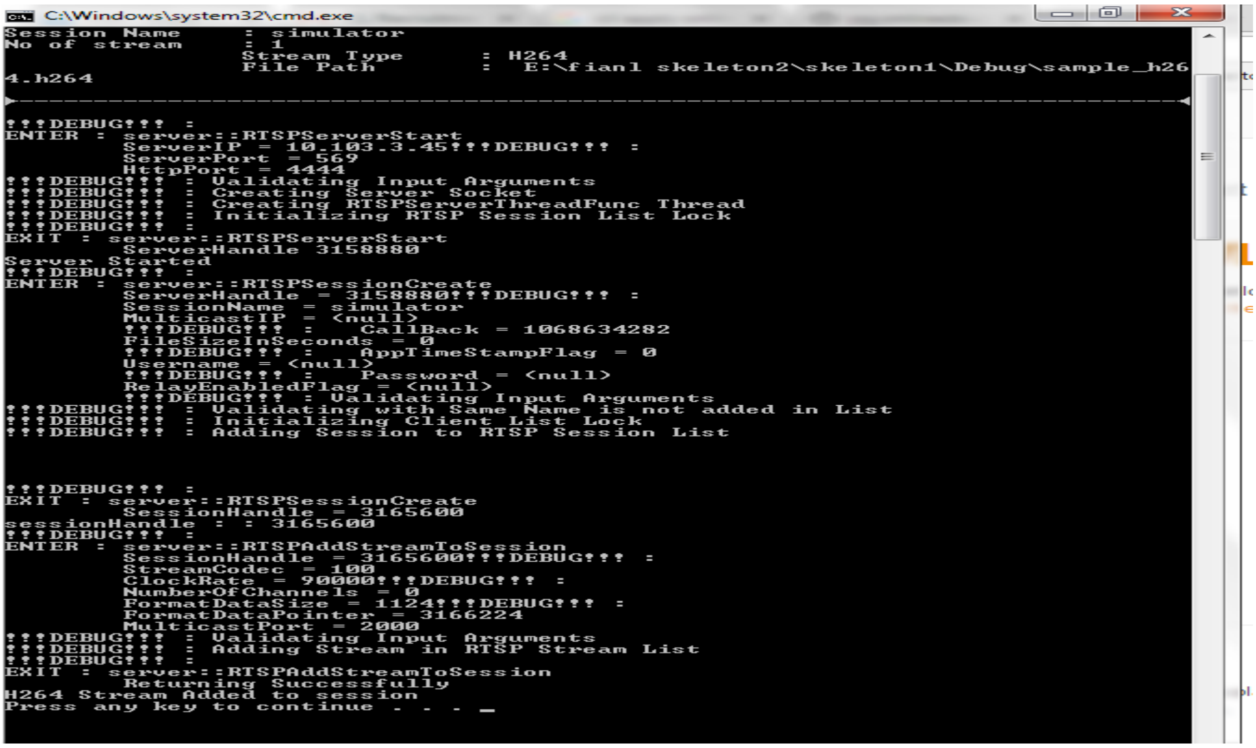

Fig 4. Scrrenshort Of Output

b. The two images shows below indicates the smooth playing without buffering of video and satiety of buffer while frames are consign to the client displaying device.

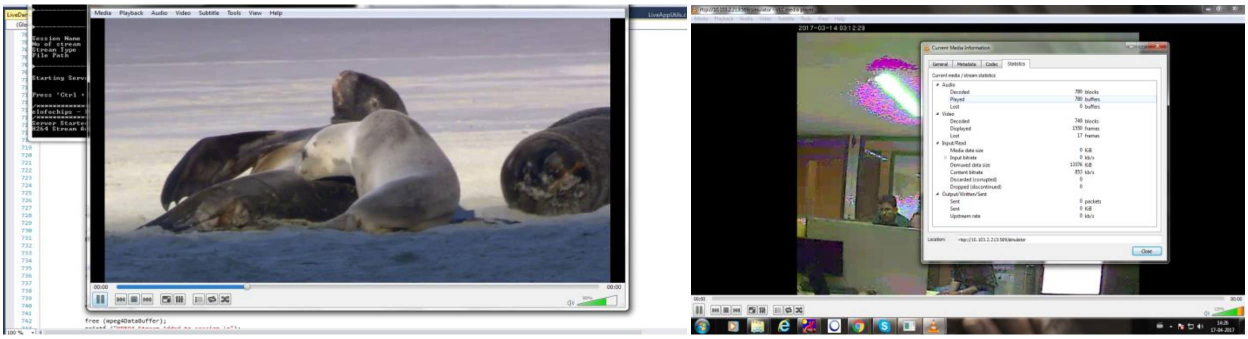

Fig 4: The video is being displayed and the buffer sis filling with frames which is being send by the server

c. The two images shown below shows the difference between delay of existing system which is 1s while displaying and shows blurred display or video till some more number of seconds. The proposed system show the delay of 500ms where frames are immediately displayed with clearity.

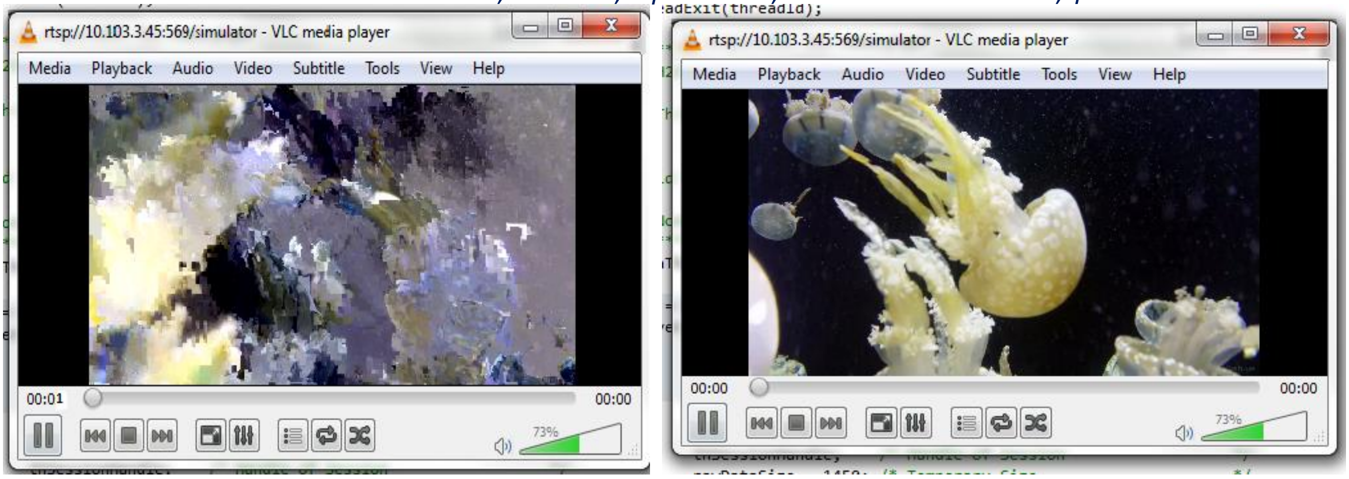

Fig 5: The vlc player showes the difference between delays of both existing appication and proposed application

d. Below shown graph is the delay difference between existing and proposed system which shows the existing delay is much higher than the proposed system delay.

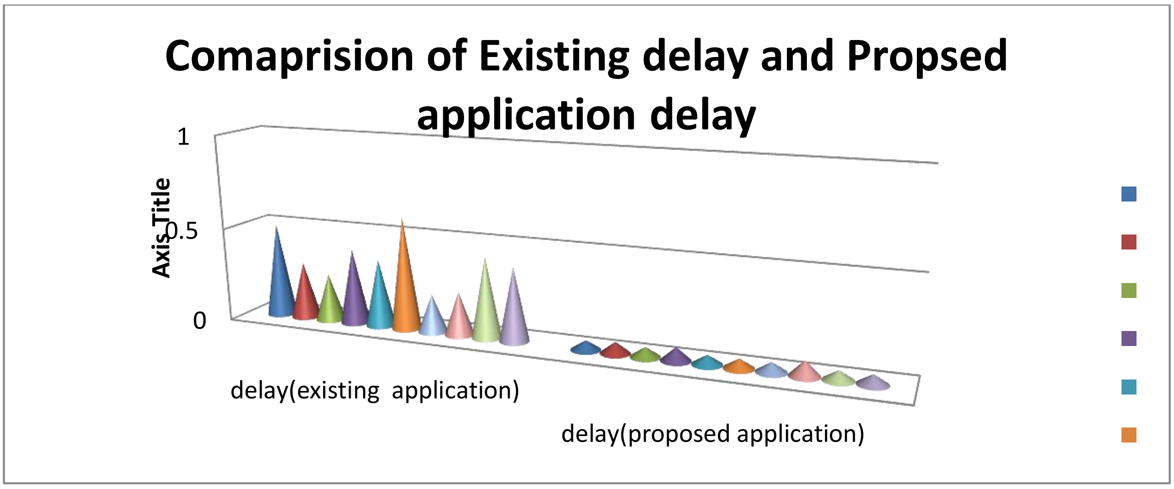

Fig 6: comparison graph of system

e. Shows the application speedy work vs existing work speed

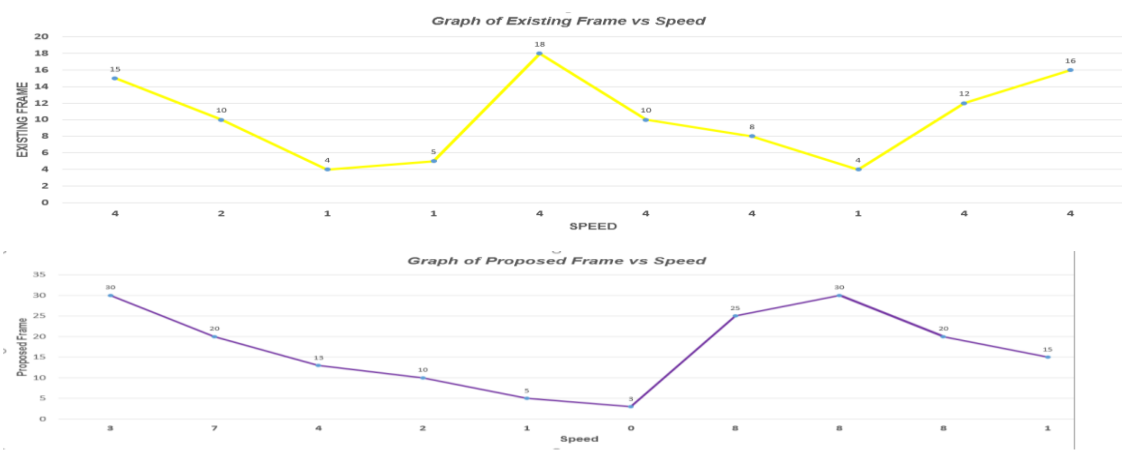

Fig 7: comparison of proposed system vs existing system

VI. CONCLUSION AND FUTURE WORK

In our vms system we are using RTP protocol in video on demand and live streaming of video. We considered the RTP/RTSP and RTCP environment where RTP is used to transmit and realtime data and RTCP is used in synchronicity with RTP to monitor the network. The server side will be taking care of the all the raw data taken form repository should be directly delivered to client in appropriate format as this provide various codec supports like h.264, mpeg, mpeg4 etc. with the additional methods of PLAY,PAUSE, STEPUP, STEPDOWN ,FIRSTFRAME,LASTFRAME. After the codec processing the RTP packet are forged to client site. Server side will be working according to client's command like PLAY. Possible extensions of the present work include the analysis of similar delay and jitter loss.Less delay while transferring of data and more improvement in the quality of video being streamed with more number of codecs used.

REFERENCES

[1] Jennibeth F. Gatal1, Edzel Raffiñan2, John Imperial2 and Jefferson A. Hora,"FPGA-based H.264 video decoder in RTP palyload format" microelectronics lab, 8th IEEE international conference Humanoid 9-12 December 2015 .

[2] Tzu-Hung Lin, Cheng-yeh, Shaw-Hwa Hwang," Retransmission and flow control for a video Real-time Transport protocol" AASRI Conference of sports engineering and computer science 2014.

[3] Binwen Fan, Guoyue Sun, Ming Liu, ChunyangLi"aquistion and transmission in the video telephone based on android" 12th IEEE international conference on fuzzy system and knowledge discovery 2015 Key Laboratory of Key Technology of the Internet of Things Terminal Harbin Institute of Technology Shenzhen Graduate School Shenzhen, China.

[4] Bo Shen, Sung-Ju Lee, Sujoy Basu, "performance evaluation of transcoding-enabled streaming media caching system".

[5] Iffat Ahmed, Lenoardo Badia, Khalid Hussain "Evaluation of deflicit round robin queue discip;ine for reeal time traffic management in RTP/RTCP environment"2012.

[6] Du Xiao-dan1, Hu Qing1 and Liu Yong-hong1, Yu Hong2 "The system construction and the implementation of Qos control mechanism in intelligent streaming media." IEEE 2012.