I.J. Wireless and Microwave Technologies, **2022, 2, 51-59**
Published Online April 2022 in MECS (http://www.mecs-press.org/)
DOI: 10.5815/ijwmt.2022.02.05

QOE Improvement for Dynamic Adaptive Streaming of Multimedia in LTE Cellular Network Using Cross-layer Communication

Anand D. Mane
Electronics and Telecommunication Department of Sardar Patel Institute of Technology, Andheri(w), Mumbai, India
Research scholar at Electronics and Telecommunication Department of Terna Engineering College, Nerul, Navi Mumbai, India
E-mail: anandmanearya@gmail.com

Uday Pandit Khot
Electronics and Telecommunication Department of St. Fransis Institute of Technology, Borivali (w), Mumbai, India
E-mail: udaypandit@rediffmail.com

Received: 26 August 2021; Revised: 29 September 2021; Accepted: 28 October 2021; Published: 08 April 2022

Abstract: This Research focuses on cross layer approach to enhance the User Experience and Quality of Service of video streaming in Long Term Evolution mobile network. During run time channel quality index is observed. Application layer requirement is fulfilled by changing modulation techniques as well as dynamic allocation of resources. This paper proposes a algorithm which improves the End-to-End Delay, Peak Signal to Noise Ratio of transmitted video over Mobile Network. Experimental results show the improvement in end to end delay by 89% and Peak Signal to Noise Ratio by 8%. Simulation in Network Simulator-3 provide credible evidence that this proposed cross-layer algorithm outperforms between earlier algorithm in providing better Quality Of Experience for real time, adaptive video streaming over Long Term Evolution.

Index Terms: Cross Layer, Long Term Evolution, Mobile Networks, Mean Opinion Score, Peak Signal to Noise Ratio, Quality Of Experience, Quality Of Service, Reference Signal Received Power, Structure Similarity Index Method.

1. Introduction

The Video streaming in the cellular Network is growing and expanding services due to the emergence of multimedia based applications. The video industry has adopted dynamic streaming over Hyper Text Transfer Protocol(HTTP), to provide smooth viewing experience in varying wireless environment. However there are several challenges to develop robust streaming solutions in mobile network. In streaming application the video content is sent in compressed form over the internet and at the receiver it is decoded and played in real time. The number of parameters need to be considered during live video streaming in wireless environment. Some of the factors that influence the video streaming are delay, packet loss, jitter and compression techniques. Internet traffic is time dependent, show the quality of service (QOS) in heterogeneous networks becomes challenge. However QOS is an end-to-end key issue for communication link quality measure. In wireless channel link conditions are fluctuating, which demand for changing bandwidth that requires adaptive video streaming for the effective and higher quality of experience (QOE). Further, it may create video freeze at display in the user equipment due to more weight for the video frame to buffer. [1]

The LTE cellular services have been deployed to match the growing traffic demand for video streaming, with a peak downloading bit rate of 300 mbps, almost 10 times more than 3G. However, user dependent quality of experience (QOE) remains unsatisfactory. A recent world-wide measurement survey [2] reveals that, even wide LTE coverage in the regions, will boost the video quality by 20% over 3G network on the other hand, the average stalling time remains 7.5 to 11.3 seconds for each minutes mobile video playback. [3]

These two effects are seemingly antithetical: the video streaming applications are sorely under utilize the LTE bandwidth, but stalling happens due to bandwidth overestimation. However, based on Reference Signal Received Power (RSRP) measurements study, we found a single root cause behind is the inability of the streaming application to track the network bandwidth which is affected by down link traffic dynamics at the E-nodes (Base Stations) of LTE.

2. Long Term Evolution Evolved Packet Core Network Architecture

An overview of the LTE-EPC NS3 simulation model is shown in the Fig. 1 Overview of the LTE-EPC simulation model.[4] There are two main components:

1. The LTE Model. This model includes the LTE Radio Protocol stack (RRC, PDCP, RLC, MAC, PHY). These entities reside entirely within the UE and the eNB nodes.
2. The EPC Model. This model includes core network interfaces, protocols and entities. These entities and protocols reside within the SGW, PGW and MME nodes, and partially within the eNB nodes.

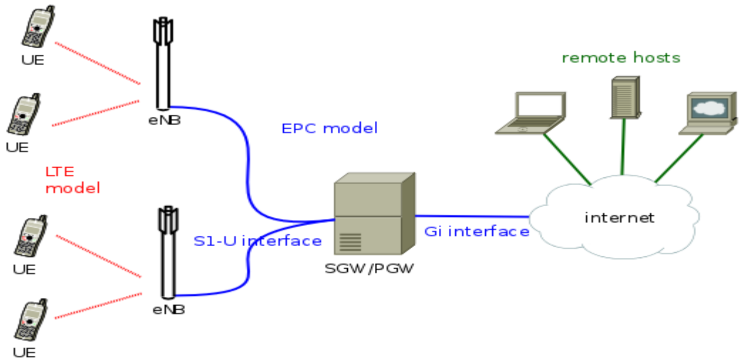

Fig.1. LTE EPC Radio Network [4]

3. Related Work

The Application layer plays very important role in video streaming applications. It is selecting the appropriate encoding techniques for video frames based on type and requirements. The video on demand services require transmitting high quality video, however it can tolerate reasonable amount of delay so less transmission errors. Where as, real time streaming services are very strict and require low delay and jitter. [2] Video compression and encoding is very important to utilize the bandwidth. According to [5], a raw video data of SD TV will need at least 1.09 gigabits/s to be approximately received, while artificial HDTV encoded video application over 6 MHz channel needs only 20 Mbits/s. The idea of compression is to get rid of redundancy in the frames. Redundancies can be spatial, such as when part of picture have the same color like a painted wall. The other type of redundancy is temporal redundancy, such as consecutive frames having the same background. Shannon's lossless coding theorem state that it is impossible to have a lossless compression coding rate less than a source entropy. The year 1984 marks the birth of the first video coding international H120 by the ITU-T formerly known as international telegraph and telephone consultative committee (CCITT). The performance of the H120 was A remarkable on the special resolution however it had a very poor temporal resolution. After that different standards has evolved specially the H26 family and the MPEG family.

The LTE network is widely used for multimedia transmission. The quality of transmission is measured in terms of PSNR, SNR SSIM, VQM and interpacket delay. The various algorithms were proposed on Application, Transport and MAC layers in past decade. The band width allocation scheme was suggested to reduce the transmission delay in [3]. The equi-SNR and equi-PSNR in two dimensional domain was suggested in [6] to improve PSNR for video service over LTE. Further L. Arun Raj et al. [7] used buffer management technique at client side to improve SNR, PSNR and SSIM. Jie Tian et al. [8] proposed game theory and optimization algorithm to improve the PSNR. Play back smoothness was improved using DFSRA algorithm [9] . The mean score opinion was improved by OTT DASH [10] and IP-centric QOS model based on IP policies [11].

The survey of Multimedia Video transmission over LTE was also reported by [2] and [17]. In this survey Application ,MAC and Transport layer modification Techniques were discussed for improvement of video parameters like Delay,PSNR and SSIM.

However the RSRP (CQI) and Modulation technique jointly was not considered for band width allocation. Hence the proposed algorithm allocates the bandwidth by jointly considering RSRP and modulation technique to improve Delay, PSNR of transmitted video.

4. QOE Measurement Parameters

4.1 Peak Signal to Noise Ratio

PSNR is used to calculate the ratio between the maximum possible signal power and the power of the distorting noise which affects the quality of its representation. This ratio between two images is computed in decibel form. The PSNR is usually calculated as the logarithm term of decibel scale because of the signals having a very wide dynamic range. This dynamic range varies between the largest and the smallest possible values which are changeable by their quality.[12] The Peak signal-to-noise ratio is the most commonly used quality assessment technique to measure the quality of reconstruction of lossy image compression codecs. The signal is considered as the original data and the noise is the error yielded by the compression or distortion. The PSNR is the approximate estimation to human perception of reconstruction quality compared to the compression codecs.

PSNR is Expressed as:

$$\text{PSNR} = 10 \log 10\ (\textit{peakvalue}^2)/\text{MSE} \qquad (1)$$

4.2 End-to-End Delay

Delay is the Intrinsic parameter to the communication network, since destinations are far apart and it takes some time to reach information at destination. Delay is also referred as to latency. Delay time can increase if packets experience long queue in network or crosses a less direct route to avoid congestion. The delay can be measured either one-way (the total time from the source that sends a packet to the destination that will receive it), or round-trip (the one-way latency from source to destination plus the one-way latency from the destination back to the source). Round-trip delay is used frequently, because it can be measured from a single point. The round trip delay is a relatively accurate way of measuring delay, because it excludes the amount of time that a destination system spends processing the packet. End-to-End Delay is the time taken by packet to reach the destination, excluding processing of packets at destination.

5. Cross-Layer Design

In this work we strive to develop the algorithm which satisfies the multiple goals. First it considers QOE perceived by the user which is derived from the application requirement, this is main performance metrics. It improves the video playback quality even for random channel fluctuations to give smooth playback user experience. Second task is to maintain fairness among the users even for non-video traffic by maximizing the minimum QOE between them. The resource allocation handled in NS-3 LTE is described in [4], describes how it is modeled in the simulator. The scheduler is in charge of generating specific structures called Data Control Indication (DCI) which are then transmitted by the PHY of the eNB to the connected user equipment's(UEs), in order to inform them of the resource allocation on a per sub frame basis. In doing this, in the downlink direction, the scheduler has to fill some specific fields of the DCI structure with all the information, such as: the Modulation and Coding Scheme (MCS) to be used, the MAC Transport Block (TB) size, and the allocation bitmap which identifies which Resource Blocks will contain the data transmitted by the eNode-B (Base Station) to each user.

5.1 Adaptive Modulation and Coding

The simulator provides two Adaptive Modulation and Coding (AMC) models: one based on the GSoC model and one based on the physical error model. The procedure described in \cite{a4} is used to get the corresponding MCS scheme. The spectral efficiency is quantized based on the channel quality indicator (CQI), rounding to the lowest value, and is mapped to the corresponding MCS scheme. The MCS index goes from 0 to 31, and 0 appears to be a valid MCS scheme (TB size is not 0) but the first useful MCS index is 1. Hence to get the value as intended by the standard we need to subtract 1 from the index reported.

5.2 Channel and Quality Of Service Aware Scheduler

We have used Channel and QoS Aware Scheduler for our simulation scenario. The Channel and QoS Aware (CQA) Scheduler is an LTE MAC down link scheduling algorithm that considers the head of line (HOL) delay, the GBR parameters and channel quality over different sub bands. The CQA scheduler is based on joint time domain(TD and Frequency Domain (FD) scheduling. In the TD (at each TTI) the CQA scheduler groups users by priority. The purpose of grouping is to enforce the FD scheduling to consider first the flows with highest HOL delay. Further details can be found out at [4]. The existing scheduler program is modified to consider the requirements of application program QOE in connection with adaptive modulation. This modification gives us excellent improvements in End-to-End Delay, PSNR values as evident in various plots.

6. Experimental Environment and Simulation Parameters

For all experimentation work of this paper, we installed NS3 version 3.29 on 64-bit Ubuntu 18.04 LTS operating system. We have used NS-3 package Lena-LTE-EPC to simulate 4G-LTE. As an IDE for NS-3, we used Eclipse Neon-3, which is best for Ubuntu based systems. For the conversion of video in different encoding format, we used FFmpeg [13] which is the free tool for video conversion.

We have used client/server architecture application designed by GERCOM group [14] known as Evalvid. This software provides platform for researchers to analyze video quality for their designed scenario. This tool set provides the facility to transmit real-time compressed mp4 video towards communicating node. We modified the original code of this tool set for our requirements to simulate 4G-LTE network. we have used constant mobility, constant acceleration and random way point model for our scenario creation. We have done another modification in original Evalvid code to send video file from server to client and from client to client via EPC node through UDP. We have also added fading model to test the performance of algorithm.

We have used four types of real video highway_cif.264, flower_cif.264, akio_cif.264 and bus_cif.264 which is available online in various resolutions. The various length and formats of these videos tested our algorithm rigorously for best results. We have picked up yuv raw format of these videos and converted them to mp4 format by using FFmpeg tool set.

To calculate the received signal quality RSRP values are used which are ranging from -45 dBm to -140 dBm. The sample RSRP value plot is shown in fig 2. These values will be treated as channel quality index to decide the modulation technique to be used. It is switching between lower modulations to higher modulation technique for current RSRP value. We have followed standard QCI reference table 1 for LTE to switch between modulation techniques. Our experimentation proved that for varing channel condition, RSRP values are degrading which increases the end-to-end time delay but PSNR values are not changing much. This indicated that QOE of received video is of good quality.

Table 1. LTE Standardized QCI Values

QCI	Resource Type	Packet Error Loss Rate	Services
1	GBR	10^{-2}	Voice
2	GBR	10^{-3}	Video Streaming
3	GBR	10^{-3}	Real Time Gaming
4	GBR	10^{-5}	Non-Conversation Video
5	Non-GBR	10^{-3}	IMS Signaling
6	Non-GBR	10^{-5}	Video(buffered streaming), TCP-based (ex.email,chat,FTP,P2P,file sharing,progressive video etc.)
7	Non-GBR	10^{-5}	Voice, Video,Interactive Gaming
8	Non-GBR	10^{-3}	Video(buffered streaming), TCP-based (ex.email,chat,FTP,P2P,file sharing,progressive video etc.)
9	Non-GBR	10^{-5}	Video(buffered streaming), TCP-based (ex.email,chat,FTP,P2P,file sharing,progressive video etc.)

To perform the experimentation we have applied first constant mobility module and tested the performance. Then we have increased the number of nodes in the EPC with dummy traffic. The calculation of PSNR, end-to-end delay of received video for all scenarios was done for comparison.

7. Result Analysis

The various simulation setup parameters are shown in table 2.

Table 2. Experimental Setup

LTE Parameters	
Carrier Frequency	2GHz
System Bandwidth	5MHz
Number of PRBs	25
SNR averaging cycle	2 sec
Link Layer Model	50
Channel Model	Urban macro cell

For experimentation simulation parameters are: carrier frequency 2 GHz, the system bandwidth 5MHz, Number of resource blocks allotted 25, signal to noise ratio averaging cycle is for 2 seconds and channel model used was urban macro cell.

The table 3 gives the comparison of proposed algorithm results with earlier work done. Results clearly shows that proposed cross-layer algorithm outperforms compared with previous algorithm. End-to-end delay is 0.02 seconds which is 80% less than previous results. The PSNR value is 40 dB showing 8% improvements in results, this indicates the quality of received video is much better even in distorted channel conditions.

Table 3.Comparative Results

Method Name	**Delay (seconds)**	**PSNR (dB)**
BW Allocation [3]	0.28	--
Equi PSNR and SNR [6]	1	36
Buffer Analysis [7]	1.28	30
DPBS and DFSRA [8]	--	37
Interference Aware [9]	0.2	37
Proposed Cross-Layer	**0.02**	**40**

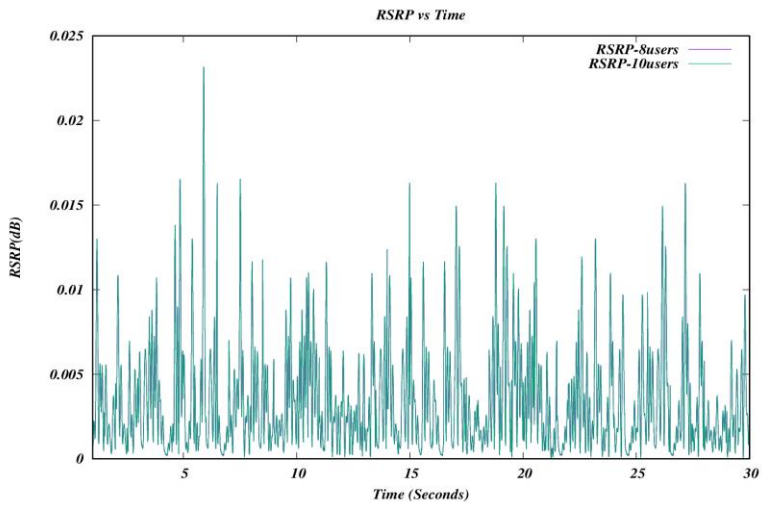

Fig.2. RSRP-SINR for 8 and 10 Users.

Figure 2 shows RSRP-SINR values which was used for deciding the modulation technique is to be used and resource block allotment for that event video transmission. These values are also called as channel quality index (CQI) as reference values are indicated in table 1. According to this the respective modulation technique and resource bandwidth allocation is done to improve the quality of video transmission.

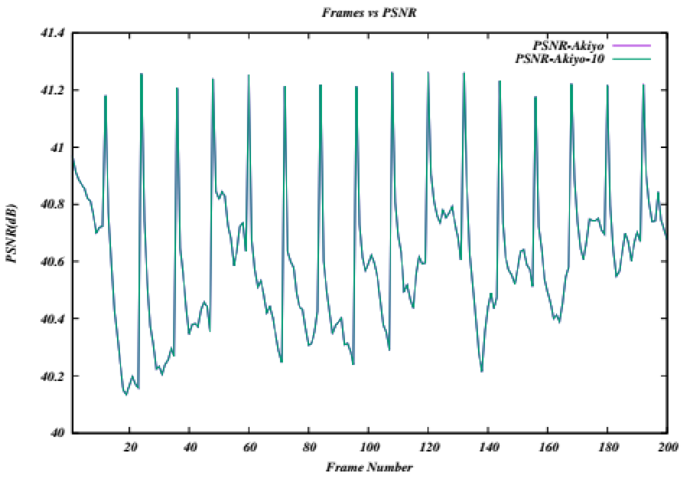

Fig.3.PSNR Akio Video Comparative

Figure 3 is a plot of PSNR values of Akio video transmission over LTE when only one user was transmitting and 8 users are transmitting. This comparison shows that even more number of users are active in a cell, PSNR values are not going below 40 dB. This is 8% improvement in PSNR values compared with [6], [7],[8] and [9].

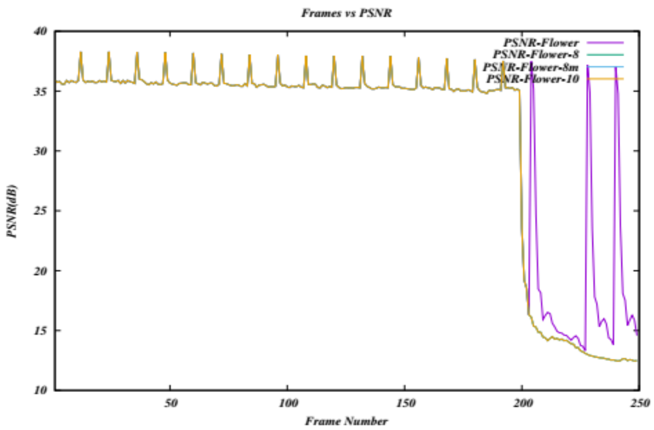

Fig.4.PSNR Flower Video Comparative

This figure 4 is the comparison of Flower video transmission results when 8 and 10 users are active in a cell. It indicates that PSNR value is upto 37 dB even more users are active and fading channels are activated. During last frames transmission some drop in PSNR values is observed. This result is very good compared with [6], [7], [8] and [9]

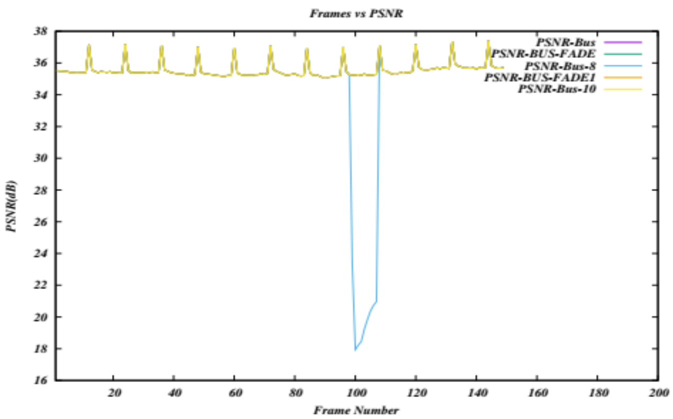

Fig.5. PSNR Bus Video Comparative

Figure 5 shows the PSNR values of Bus video transmission over LTE. It shows that PSNR values are 35.8 dB for 8 and10 users active in cell as well as fading channel is activated. This result is improved compared with [6]and [7].

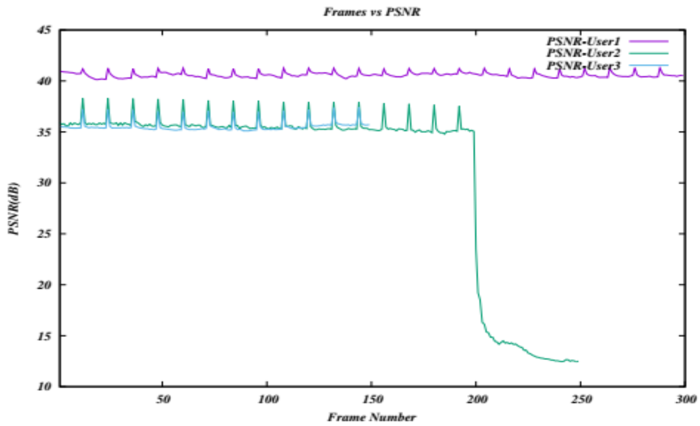

Fig.6. PSNR of Three Video Users

Figure 6 shows the comparison of PSNR values in a cell when three video users are sending video over LTE. It indicates that PSNR values are not reducing below acceptable value (35 dB). These values are even better compared with [6],[7],[8] and [9]

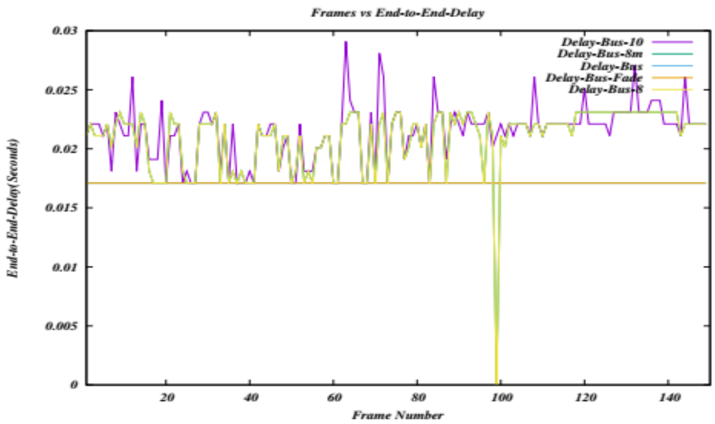

Fig.7. End-to-End Delay Bus Video Comparative

End-to-End Delay is another quality measure for video transmission over LTE. This indicates how long it will take to transmit video from one user to another. It also treated as quality of user experience parameter in terms of time taken to upload and download video. Figure 7 is a plot of End-to-End delay of Bus video tranmission over LTE for 1,8,10 users active in a cell. It shows that delay is not going above 0.02 seconds for all configurations. Improvement in delay by 89% compared with [3], [6] and [7].

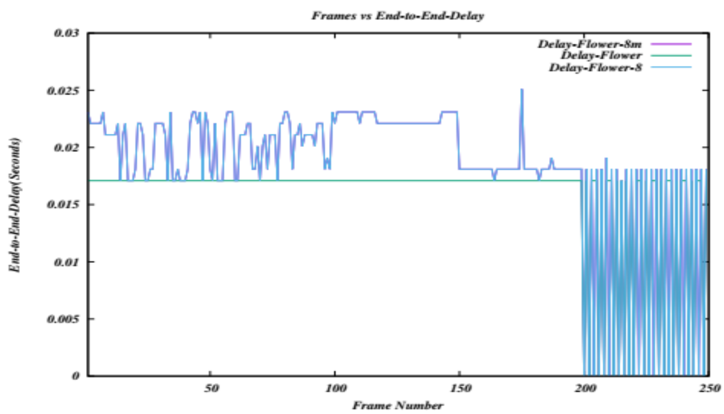

Fig.8. End-to-End Delay Flower Video Comparative

Figure 8 is delay comparison of Flower video transmission.In this plot the delay variations are evident between 0.015 seconds to 0.022 seconds. These variations are due to 1,8 and 10 users are active in a cell. Results are better compared with [3], [6] and [7]

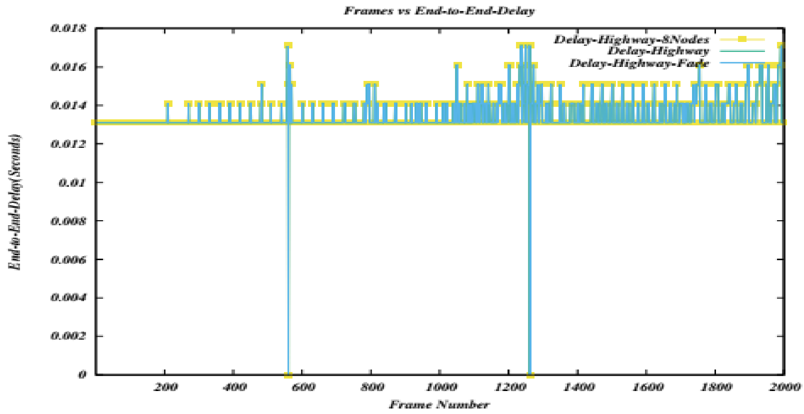

Fig.9. End-to-End Delay Highway Video Comparative

This figure 9 is a delay analysis of long duration video (2000 frames) Highway. It shows that average delay is 0.013 seconds and varying between 0.012 to 0.014 seconds. These variation are for 1 user, 8 user and 10 users active in a cell. Compared with [3],[6] and [7] results are still better for long duration video transmission.

8. Conclusion

We proposed improved cross-layer approach to enhance the user experience and QOE of multiple adaptive video streams in mobile cell. The proposed solution is having two modifications over existing work that is, adaptive modulation and dynamic resource allocation for better QOE. We separately analyzed up link and down link transmission. Further for investigation, added multiple nodes to the system having video and non-video traffic. This algorithm provides better results compared with existing methods. PSNR values are improved by 8% as well as excellent decrease in end-to-end delay by 89%. This method is giving good results for all types of videos. Simulation results prove better improvement in QOE for non-moving as well as moving users in the LTE cell for various cases.

References

[1] Hieu Le, Konstantin Miller, Arash Behboodi, and Adam Wolisz " Cross-Layer Approach for HTTP-Based Low- Delay Adaptive Streaming in Mobile Networks", IEEE 18th International Symposium on World of Wireless, Mobile and Multimedia Networks (WoWMoM) , June 2017.

[2] Ahmed Ahmedin, Amitabha Ghosh, and Dipak Ghosal " A survey of Multimedia Streaming in LTE Cellular Networks", arXiv , 1803.05080v1 , cs.NI, 14 March 2018.

[3] Stefania Colonnese,Francesca Cuomo,Tommaso Melodia,and Izhak Rubin," A Cross-Layer Bandwidth Allocation Scheme for HTTP-Based Video Streaming in LTE Cellular Networks", IEEE Communication Letters, VOL. 21, NO. 2, February 2017.

[4] Website "NS3-discrete-event network simulator ",official reference for NS3 Project Available[online]: https://www.nsnam.org/ (accessed on 20 January 2021)

[5] Nabeel Khan and Maria G. Martini," QoE-driven multiuser scheduling and rate adaptation with reduced cross-layer signaling for scalable video streaming over LTE wireless systems", EURASIP Journal on Wireless Communications and Networking, Springer Open Acess Journal,2016.

[6] Yongil Kwon, Doug Young Suh, Sung Chun Kim and EenKee Hong, "Application driven, AMC-based cross-layer optimization for video service over LTE", EURASIP Journal on Wireless Communications and Networking , Springer Open Acess Journal, 2011.

[7] L. Arun Raj,Dhananjay Kumar. H. Iswarya, S. Aparna and A. Srinivasan "Adaptive video streaming over HTTP through 4G wireless networks based on buffer analysis",EURASIP Journal on Image and Video Processing,10.1186/s13640-017-0191-4,2017.

[8] Zhang, Dalei Wu,and Dongfeng Yuan,"Interference-Aware Cross-Layer Design for Distributed Video Transmission in Wireless Networks", IEEE TRANSACTIONS ON CIRCUITS AND SYSTEMS FOR VIDEO TECHNOLOGY, VOL 26,NO.5,MAY 2016.

[9] M Zhao, X Gong, J Liang, W Wang,X Que, S Cheng, "QOE-Driven Cross-layer optimization for wireless dynamic adaptive streaming of scalable videos over HTTP", IEEE Transactions on Circuits and Syatems for Video Technology,25(3),451-465, 2014.

[10] Ali El Essaili, Damien Schroeder, Eckehard Steinbach,Dirk Staehle,and Mohammed Shehada, "QOE-based Traffic and Resource Management for Adaptive HTTP Video Delivery in LTE", IEEE Transactions on Circuits and Syatems for Video Technology,1051-8215, 2013.

[11] William Diego and Isabelle Hamchaoui, Xavier Lagrange,“ Cross-layer Design and Performance Evaluation for IP-Centric QOS Model in LTE-EPC Networks”, IFIP 8th Wireless and Networking Conference,2015.
[12] Li, C.F. and Bovik, A.C., “ Three-Component Weighted Structural Similarity Index. Image Quality and System Performance VI”, SPIE Proc. 7242, San Jose, CA, 19 pp 1-9, January 2009.
[13] Website “Ffmpeg”,Avaliable online:https://www.ffmpeg.org/ (accessed on 20 January 2021)
[14] Website “ GERCOM group ”, Avaliable online: https://www.gercom.ufpa.br/ (accessed on 20 January 2021)
[15] Hasan sinky,Bechir Hamdaoui, Mohsen Guizani,“ Handoff-Aware Cross-Layer Assisted Multi-Path TCP for Proactive Congestion Control in Mobile Heterogeneous Wireless Networks”, IEEE ,2015.
[16] Bo Fu, Yang Xiao,Hongmei Deng ,Hui Zeng,“ A Servey of Cross-Layer Designs in Wireless Networks”, IEEE Communication Surveys and Tutorials, Vol 16,No. 1,2014.

Authors’ Profiles

Anand D. Mane received his B.E.degree in Electronics Engineering from the Dr.B.A.M.U. Marathwada University, Aurangabad-MH and M.E. in Electronics from V.J.T.I, Mumbai University, Maharashtra, India.

He is Research Scholar in Electronics and Telecommunication Engineering Department at Terna Engineering College,Nerul (W), Navi Mumbai, Mumbai University Research Center. He is Faculty of Electronics and Telecommunication Department at Sardar Patel Institute of Technology, Andheri (w), Mumbai, Maharashtra,India since 2006. His research interests include Wireless Communication, Mobile Cellular Networks.

Uday Pandit Khot received his B. E.degree in Industrial Electronics Engineering from the Amravati University, Amravati, M.Tech. in (Electronic Systems, Spec. in Control and Instrumentation), and Ph. D.in Electronic Systems, IIT Bombay,Mumbai, Maharashtra, India. He has been 27 years of Teaching and 10 years of research Experience. He has been Professor of Electronics and Telecommunication,Saint Francis Institute of Technology, Borivali (W), Mumbai, Maharashtra, India since 2012. He received Minor Research Grant from University of Mumbai. He has Professional Memberships of IEEE (USA) Member (Since 2002) Fellow:IETE (India) (Since 2000) Member: IE (India) (Since 2006) Member: ISTE (India) (Since 1994) His research interests include analog and digital circuits, current mode circuits, Wireless Communication, and Mobile Networks.includes the biography here.

How to cite this paper: Anand D. Mane, Uday Pandit Khot, " QOE Improvement for Dynamic Adaptive Streaming of Multimedia in LTE Cellular Network Using Cross-layer Communication", International Journal of Wireless and Microwave Technologies(IJWMT), Vol.12, No.2, pp. 51-59, 2022.DOI: 10.5815/ijwmt.2022.02.05