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An Approach for Cross Layer Design Framework in MANET: Countermeasure and Analysis

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Abstract - **Video content has an impact on video quality under same network conditions. This feature has not been widely explored when developing reference-free video quality prediction model for streaming video over wireless or mobile communications.MANET has much significance in several areas but it is limited to support real time traffic transmission in efficient way. To minimize this limitation, numerous way or approach suggested by various authors. One of the author, suggested cross layer design to achieve video transmission using TCP Friendly Rate Control (TFRC) which uses four traffic class (TC) but the efficiency still not achieved as expected. In this research work, an approach is suggested to design cross layer architecture using TCP Friendly Rate Control (TFRC) mechanism with different classifier and priority queue. Here, priority queue is used to distinguish traffic type that enables separate transmission of non real time and real time traffic. Proposed approach has intended to minimize delay jitter, end to end delay, maximize packet delivery ratio and throughput. The implementation of the proposed approach of cross layer design framework for effectively transferred video packets is performed using NS 2 and when compared with conventional techniques in MANET, our results in average end-to-end delay, average PSNR, is considerably reduced with increase in high throughput and good delivery ratio. The experimental results shows the adoptable performance of the basic flow and improves the different performance parameters**

Keywords: MANETs, Multimedia, Video Transmission, TFRC, Cross-layer, Protocol, AODV

I. INTRODUCTION

Networking a large number of wireless devices in ad hoc mode will facilitate a wealth of applications not feasible under the conventional base station-to-network node communication model. An infrastructure less network has to rely on the collaboration among network nodes in implementing most, if not all, network operations. . For example, two nodes that are not within the direct communication range will have to rely on intermediate nodes to exchange messages, thus forming multichip networks [1]. The cross layer design (CLD) approach is a new dynamic area of research into MANET networks. This approach provides new possibilities to increase the performance and adaptability of MANET [2]. Evolutionary approach to cross layer design is based on extending the layered structure to maintain compatibility with existing systems and networks.

A. MANET

A Mobile Ad hoc NETwork (MANET) is a system of wireless mobile nodes that dynamically self-organize in arbitrary and temporary network topologies. People and vehicles can thus be internetworked in areas without a preexisting communication infrastructure or when the use of such infrastructure requires wireless extension [3]. In the mobile ad hoc network, nodes can directly communicate with all the other nodes within their radio ranges; whereas nodes that not in the direct communication range use intermediate node(s) to communicate with each other. In these two situations, all the nodes that have participated in the communication automatically form a wireless network, therefore this kind of wireless network can be viewed as mobile ad hoc network [4].

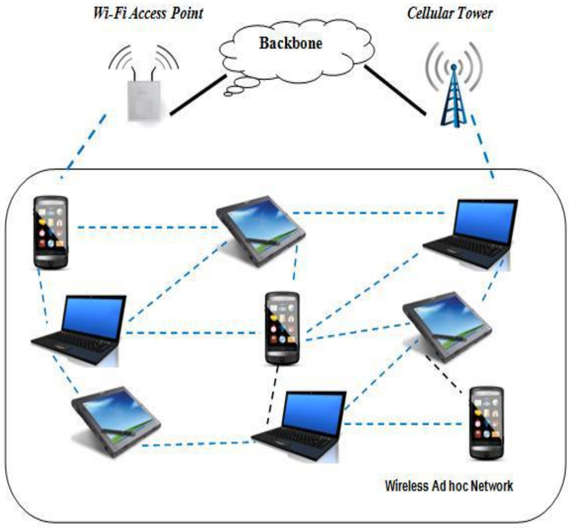

Figure 1: Mobile Ad-hoc Network

MANET can provide rapid connection between independent mobile users. Examples include establishing survivable, efficient, dynamic communication for emergency/rescue operations, disaster relief efforts, military networks, conference or campus networks, car networks, personal networks, etc.

B. *Video Streaming*

Transmission of video in continuous form is known as video streaming. Video streaming requires a camera to capture video, an encoder to encode video, a media publisher and a network to deliver and distribute video. It refers to the consumption of video data while it is being delivered. A video stream can be live or on demand. Live Streaming refers to streaming of data from a live event, where the media data is not stored for a period of time before it is distributed. Streaming is necessary to enable viewing of live multimedia content. On-demand Streaming refers to streaming of non-live media, where the media data is stored and transmitted to the user upon request. The demand for multimedia content in general has steadily increased since the introduction of broadband Internet services for home customers, and as bandwidth continues to increase, so does the demand for multimedia content [5].

II. LITERATURE SURVEY

Supporting multimedia applications over wireless links has been one of the main fields of attention in the networking and video coding communities in the last decade. There are numerous research works going on in this area.

V. K. Goyal et al. [6] proposed Multiple Description Coding (MDC) scheme for video transmission. MDC encodes the source stream into multiple sub-bit streams. The sub streams are also known as descriptions. Each sub stream has equal importance because it carries some unique information. Sub streams are transmitted over multiple independent channels in a network. Because the loss of one sub-stream does not influence other sub-streams, a lost packet in any path does not need to be retransmitted.

J. G. Apostolopoulos et al. [7] proposed a scheme that is the combination of two sub-systems, multiple state video encoding and decoding and path diversity transmission system. Initially, a video stream is encoded into various independently decodable streams. Different network path is chosen for transmitting each stream. The advantage of this method is that if a stream is lost, then it can be recovered with the help of other received streams. The path diversity transmission system is used to send different packets over multiple paths. Since various streams are transmitted over different paths, there are fewer chances that all the streams are simultaneously corrupted.

S. Mao et al. [8] proposed three MCP-based video transmission techniques for MANET. The techniques are (1) Feedback Based Selection method (2) Layered Coding technique (3) Motion compensation coding using multiple descriptions. In feedback based selection, the reference frames are selected based on feedback and predicted path status. Reference frame is the last frame which is correctly received. Layered coding with selective ARQ makes the use of layered video coding in which video is encoded into two layers EL and BL. The video quality is further enhanced by applying multiple description motion compensation coding technique.

J. Kim et al. [9] proposed channel adaptive MDC technique for reliable video transmission in wireless networks. This system uses an optimized splitting algorithm. In this technique, MDC produces two correlated sub-streams from a single description coder using an optimized splitting algorithm. This system is robust and provides a relatively good value video at the receiver end.

M. A. Igartua et al. [10] proposed a multipath routing technique. In this technique, initially routes are discovered by using dynamic source routing and a probe message packet is transmitted from source to destination using these routes. Once the first probe message is received at the destination end, a time out is triggered. The probe message reply packet is sent to the source using the same path from which it is received. Routes are classified as best route, medium route and worst route. Various paths are used for sending data according to data priorities. Best path is used for sending high priority packets, medium path is used for transmitting medium priority packets and worst path is used for sending least priority packets

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III. PROPOSED WORK

A. *Domain Overview*

Emerging services in the context of the Future Internet pose new requirements on the underlying network infrastructure that derive from the need to operate on top of dynamic, heterogeneous and complex environments. In order to cope with these requirements, such services will be expected to (i) be aware of the context and the environment in which they operate, (ii) self-configure and self-adapt according to the network conditions that they sense and (iii) require minimum feedback from the end-user avoiding any explicit human intervention. Now a day there is tremendous growth in the use and applications requiring MANETs (Mobile Ad hoc NETworks) which will be very helpful in multimedia services. Enhancing and maintaining the video quality over IEEE 802.11e.

In this presented work, in order to maximize the performance between received and transmitted video signals of nodes, a cross layer framework is designed for the successful TCP video transmission then the necessary decisions to act on different layers' parameters dynamically. Preliminary simulation results show that our proposed network design can improve the performance of video-streaming transmissions over MANETs on different parameter basis like end to end delay, packet delivery ratio, PSNR, MSE, etc in spite of frequent changes in network topology and node conditions.

B. *Methodology*

Performing real-time video transmission over MANET introduces problems because of strict bandwidth and delay requirements. Live video transmission requires much less delay jitter as compared to other applications.

Delay jitter occurs in MANET because of following reasons:

- Delay Jitter at MAC layer due to interface queue.
- Delay Jitter due to frequent route breaks.
- Delay Jitter due to congestion in network:
    a) Congestion due to less receiving power of receiver node.
    b) Congestion due to poor range between two nodes.

Delay Jitter at Mac Layer Due to Interface Queue

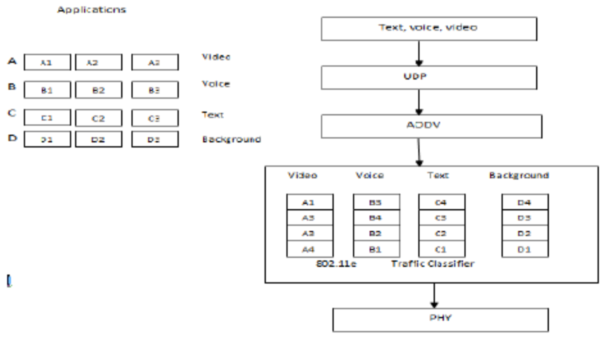

Figure 2: Separate Queue for each traffic category

Parameters for each traffic class are described below:

- Transmission Opportunity (TXOP) –A station can transmit frames only during the time interval defined by the EDCF transmission opportunity limit.
- Arbitration Interframe Space (AIFS) – When a station wants to transmit data, it first checks that the medium is idle for a duration defined by the arbitration interframe space (AIFS [TC]). If the medium is idle for AIFS [TC] then it performs a random backup procedure. AIFS is different for different traffic categories. The equation used to calculate AIFS is as follows:

$$AIFS = SIFS + n * \text{Slot time}$$

- Contention Window (CW) - Contention window is used to select the value of random backoff counter. Each station has a contention window. Backoff counter is an integer and its value is chosen randomly out of an interval between 0 and CW.
- Persistence Factor (PC) – After any unsuccessful transmission, contention window is increased to avoid further collision. Maximum value of contention window is CW_MAX. Persistence factor is different for each traffic class.

Table 3.1 IEEE 802.11e parameters

	TC [0] Video	TC [1] Voice	TC [2] Text	TC [3] Background applications
PF	4	4	4	4
AIFS	4	3	2	7
CW_MIN	5	12	27	27
CW_MAX	16	32	1024	1024
TXOP limit	0.008	0.004	0.01	0.1

Table 3.1 shows QoS parameters for various traffic categories. We consider four TCs for our proposed work**.** Figure 3 shows proposed cross layer design between the application layer and MAC layer. Application layer provides QoS parameters to the MAC layer. When the MAC layer receives packet from the network layer, the traffic classifier classifies the packets on the basis of QoS parameters received from the application layer. After classification, various application based packets are maintained in packet queues and each queue has its own priority to transmit data over the physical layer. QoS parameters are provided from the application layer to the MAC layer using a cross layer technique.

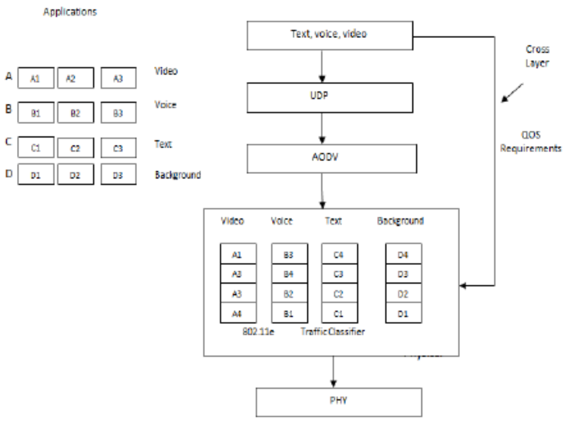

Figure 3 Cross layer between application layer and MAC laye

C. Proposed Model

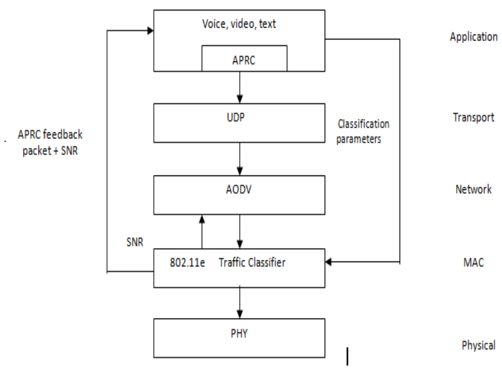

Figure 4: Integrated Cross layer architecture

Figure 4 shows proposed integrated cross layer architecture for MANET. This architecture provides real time video transmission with stable route, traffic classification and rate controller of transmission.

D. TCP-Friendly Rate Control

TFRC allows an application to transmit at a steady rate that is typically within a factor of two from the TCP rate in the similar conditions. TFRC does not halve the transmission rate after a single packet loss, but is also slow to increase the rate in the absence of congestion. The TFRC receiver is responsible for reporting the loss event rate p and average receive rate X_rcv to the sender. The sender computes the reference transmission rate X_calc based on p, X_rcv.

Calculating the loss event rate rather than simply taking the packet loss rate is an important part of TFRC. TCP does not typically reduce the congestion window more than once per a window of data. The default method that TFRC uses for calculating the loss event rate is called the Average Loss Interval. In this section we define a proposed algorithm for video transmission based on TFRC. This algorithm shows that step by step procedure where we have been summarized concept in algorithmic structure:

Table 1: Proposed Algorithms

Input: Number of Nodes, Video Packet File;

Output: Improved Cross Layer Video Transmission Rate;

Process:

1: Initialize the Network, with N nodes where $N = 1, 2, 3, \ldots..$,, in ideal condition.

2: Initialize Route Discovery by Source Node N_s

3: N_s sends RREQ Packets to Destination N_d

4: Wait Until all Route Replies not received

5: For each routing in routing table:

- ModifiedRecvTFRCresponse (reply_packet)
- {
- snr = obtain_snr (reply_packet)
- destination_address = obtain_source_address(reply_packet)
- if (snr < $Average_{SNR}$ || $\mathrm{Timer}_{\mathrm{Expired}}$ = true)
- {
- routing_information = routing_table. visit (destination_address)
- list_update_routing_path (routing_record)
- }
- average_recv_rate = data_acceptance_rate (reply_packet)
- loss_event_rate (p) = estimated_loss (reply_packet)
- adapt_transmission (destination_address, average_recv_rate, loss_event_rate)
- }

7: ***end process***

IV. IMPLEMENTATION

The simulation is being implemented in the Network simulator [11]. Protocol used here is AODV.

Table 2: Simulation Scenarios

Parameters	***Values***
Antenna Model	Omni Antenna
Dimension	500 X 500
Radio-Propagation	Two Ray Ground
Channel Type	Wireless Channel
Traffic Model	CBR
Routing Protocol	AODV
Mobility Model	Random Waypoint
Number of Nodes	20, 40, 70, 100

1. Simulation using the AODV Routing for 802.11e: In this network recreation the network is configured using the AODV protocol to configuring of 802.11e. The given simulation screen shows that all nodes are spread in topography area where source nodes are sending video packets for retrieving error free transmission but there need to improved effective transmission rate of the communicating over the network. The simulation of traditional approach is implemented with different nodes 20, 40, 70, 100.

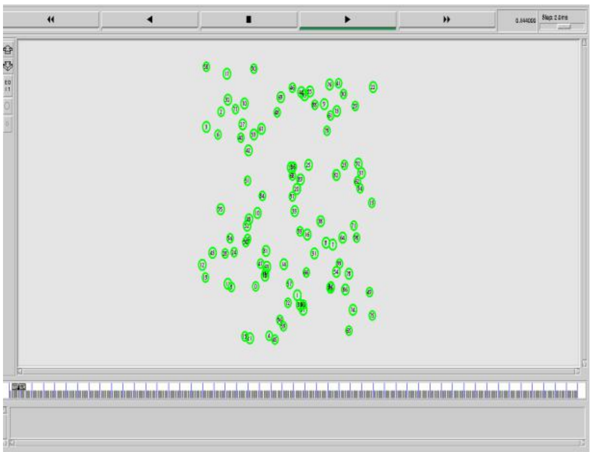

Figure 5: Traditional Approach of 802.11e

2. Simulation using proposed 802.11e with cross layer: For the proposed cross layer design framework we enhance the system accuracy in form of design parameter for evaluating transmission rate successfully using TFRC protocol. In this phase, proposed 802.11e with cross layer method is demonstrated in figure 6. In this simulation screen the green nodes show as normal network nodes with AODV modification. When the proposed method is deployed network performance is improve and large number of video packet is delivered to the destination to improved transmission rate. The proposed method is very efficient because of it produces result which is better than traditional scheme. Consequently, huge numbers of data packets are received on destination side.

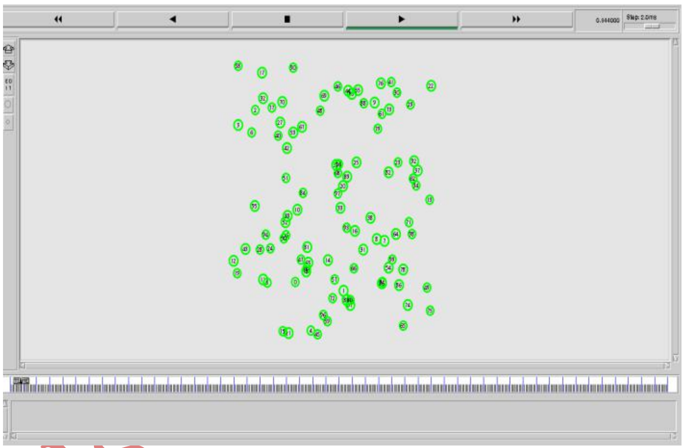

Figure 6: Proposed Approach of 802.11e with CL

V. RESULT ANALYSIS

We examine routing overhead, end-to-end delay and packet delivery ratio, etc. for overall network performance:

A. End to End delay

End to end day on network refers to the time taken, for a video packet to be transmitted across a network from source to destination device, this delay is calculated using the below given formula.

$$\text{E2E Delay} = \text{Receiving Time} - \text{Sending Time}$$

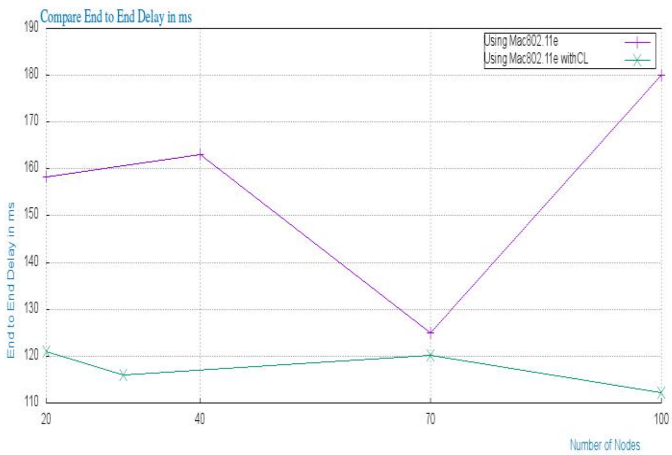

Figure 7: End to End Delays

Figure 7 shows the comparative average End to End Delay of the traditional 802.11e with AODV and the proposed cross layer framework. In this figure 5.1 the X axis contains the number of nodes in network and the Y axis shows the performance of network delay in terms of milliseconds. According to the obtained results the proposed technique is produces less end to end delay as compared to traditional technique under different nodes. We have to mention that the above improvement in average end-to-end delay is very important for video streaming applications.

B. Packet Delivery Ratio

The performance parameter Packet delivery ratio sometimes termed as the PDR ratio provides information about the performance of any routing protocols by the successfully delivered packets to the destination, where PDR can be estimated using the formula given:

$$\text{Packet Delivery Ratio} = \frac{\text{Total Delivered Video Packets}}{\text{Total Sent Video Packets}}$$

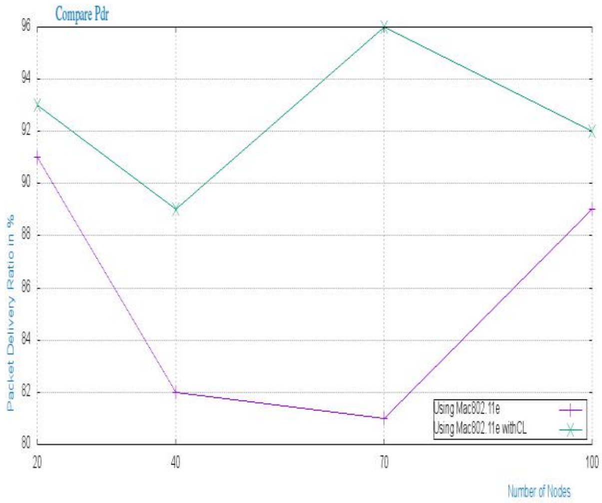

Figure 8: Packet Delivery Ratios

The comparative packet delivery ratio of the networks is given using figure 8, in this diagram the X axis shows the number of nodes in the network and the Y axis shows the amount of video packets successfully delivered in terms of the percentage. The blue line of diagram represents the performance of the traditional 802.11e scenario and the green line shows the performance of the proposed technique. According to the obtained results the proposed technique delivers more packets as compared to the traditional technique even when the network eliminates the transmission error and produces accurate data transmission. The utilization of the SNR mechanism leads to small additional significant in 802.11e and this work improve the highly delivered packet ratio.

C. Throughput

Network throughput is the average rate of successful message delivery over a communication channel. This data may be delivered over a physical or logical link, or pass through a certain network node. The throughput is usually measured in bits per second (bit/s or bps), and sometimes in data packets per second or data packets per time slot.

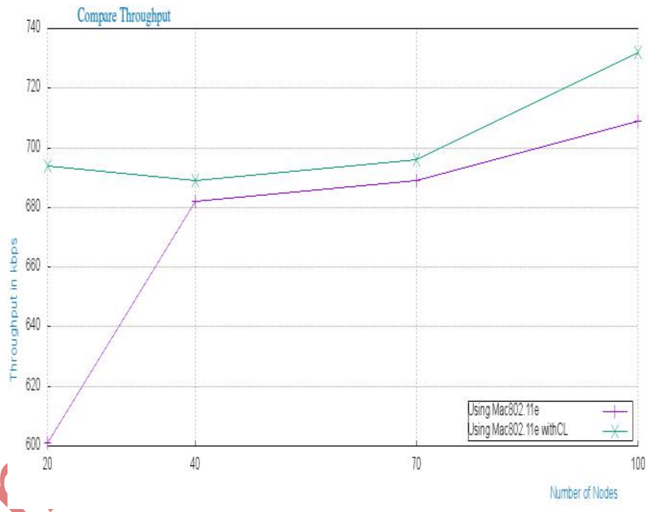

Figure 9: Compare Throughput

The comparative throughput of the network is demonstrated using figure 9 in this diagram the X axis shows the number of nodes in network and the Y axis shows the throughput of the network in terms of KBPS. The green line in this diagram shows the performance of the proposed technique and the blue line shows the performance of the old scenario. We have to mention that the improvement in throughput is important in terms of QoS from the end user viewpoint because a small increase in throughput can lead to significant enhancement of the end user knowledge.

D. Routing Overhead

During the communication scenarios it is required to exchange the packets for different tracking and monitoring purpose. Therefore the additional control messages in network is termed as the routing overhead of the network. The comparative routing overhead of both the video streaming method for video packet transmission i.e. traditional 802.11e and the proposed 802.11e with cross layer technique is given using figure 10. In this diagram the X axis shows the amount of network nodes exist during the experimentation and the Y axis shows the routing overhead of the network. In this diagram for demonstrating the performance of the proposed technique the green line is used and for traditional technique the blue line is used. According to the obtained performance of the techniques the proposed technique produces less routing overhead as compared to the traditional technique. Therefore the proposed technique offers higher bandwidth consumption as compared to the traditional routing technique under TFRC protocol.

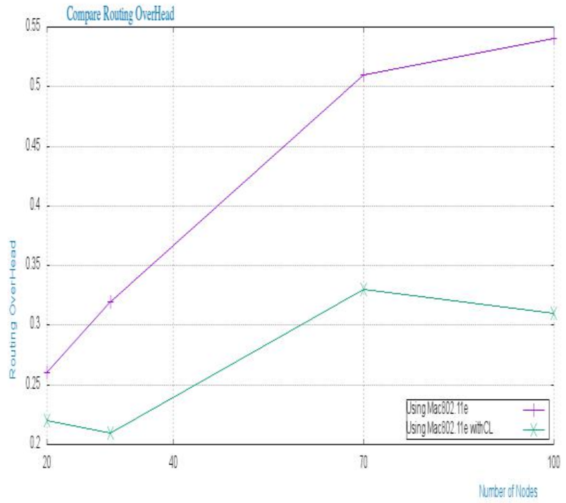

Figure 10: Routing Overhead

E. Peak Signal to Noise Ratio

The average peak-signal-to-noise ratio (PSNR) is used as a distortion measure of objective quality. PSNR is an indicator of picture quality that is derived from the root mean squared error (RMSE). The PSNR for a degraded N1 x N2 video v' with respect to the original video v is computed as follows:

$$PSNR = 20log_{10}\frac{255}{MSE}$$

Or

$$PSNR = 20log_{10}\frac{255}{(\frac{1}{N_1N_2}\sum_{x=0}^{N_1-1}\sum_{y=0}^{N_2-1}[v\,(x,y) - v'\,(x,y)]^2)^{1/2}}$$

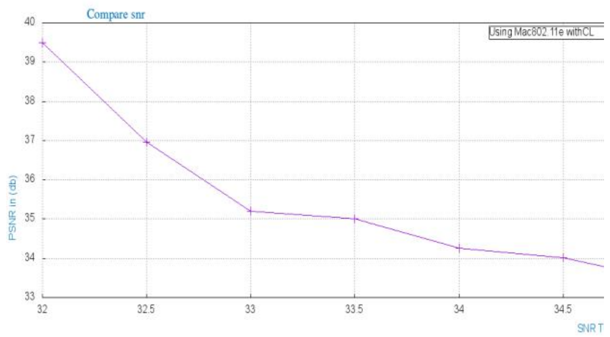

Figure 11: SVR vs PSNR

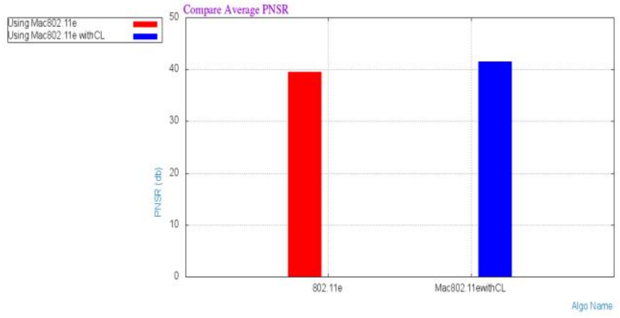

Figure 12: PSNR

The demonstration of the figure 11 is SNR threshold vs PSNR value of the proposed 80.2.11e with CL. In this X axis obtained different threshold value and in respect to Y-axis depicts PSNR values. Average Peak signal to noise ratio for both scenario of video packet transmission is given using figure 12. In this diagram the X axis shows the method name which is implemented and the Y axis shows the obtained PSNR values in db. The figure contains the red and blue column to show the performance of the 802.11e and 802.11e with CL respectively. The amount of computed PSNR is fluctuating with the video packets quality therefore that is not depends on the size of the video packet that is depends on the quality of video. From the visualization above graph, PSNR value of the proposed scheme is maximized as compared to old one.

F. Jitter

Jitter is defined as a variation in the delay of received packets. At the sending side, packets are sent in a continuous stream with the packets spaced evenly apart. Due to network congestion, improper queuing, or configuration errors, this steady stream can become lumpy, or the delay between each packet can vary instead of remaining constant.

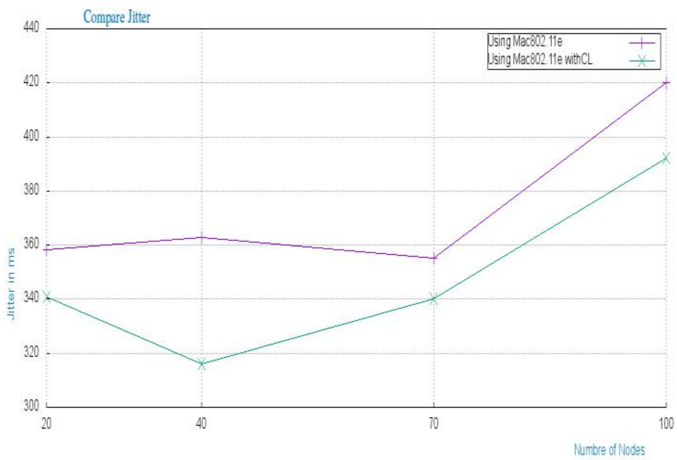

Figure 13: Jitter

In the figure 13 shows that jitter performance of the both scenario i.e. 802.11e and 802.11e with CL. X – axis contain node value and y-axis contain jitter value which show that variation of the incoming video packets. Form the result we can conclude jitter performance is higher as compared to proposed cross layer framework and packets are arrived at destination in similar manner.

VI. CONCLUSION

In this paper, we explore the prospective synergies of swap data information between different layers to support real-time video streaming. The concept of Quality of Service (QoS) in communication systems is closely related to the network performance of the underlying routing system. The APRC is used to maintain the data flow in between sender and receiver, using the feedback mechanism of the MAC layer to the application layer. The real time video transmission requires a jitter free delay in packets and the proposed architecture meets all the requirements without any aid of additional hardware and complex algorithms. Results show that the proposed strategy leads to performance improvement under several metrics, i.e. routing overhead, packet delivery ratio and end-to-end delay, Jitter and hence an improved video quality video is received at the destination side.

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