Distortion Monitored Rate Allocation for Video

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Abstract

In the approach of resource constraint video streaming, remote device streaming were developed in recent years. Wherein multiple approaches were developed towards the video streaming, for the remote streaming application, the captured sequences need to be streamed to a remote location using wireless channels. Wherein wired communication is constraint with range and maintenance, the wireless streaming are more efficient in approach. However the concern of visual quality, rate of transmission, and resource utilizations are major concern in such approach. As most of the streaming are made with devices with low resources, the samples are to be transferred in a continuous manner, as well due to multiple communication units, the issue of interference arises which leads to degradation in video quality. To achieve the objective of accurate video streaming in remote streaming application, in this paper, a new video streaming approach based on routing streaming and rate of allocation (ROA) based on distortion streaming is proposed.

Keywords: *Automated video streaming, video streaming, error resilience, rate of allocation.*

I. INTRODUCTION

With the increase in video density, it has become a prime importance to monitor the video condition in running status. For the live streaming of video condition, multiple communication units are installed at video codecs or on the routes to capture the video conditions of moving data. As the communication units are installed at remote locations, it is required to gather all these observations at a centralized location to achieve common observation of a wide observation area. It is hence required to transfer the monitored data to centralized location so as to give a common observation. However, the streaming centres are located at a far distance from these communication units, and it is required to transmit the captured video sequence to this far location for streaming. Transmission of data from one point to other is achievable via wired mode or wireless mode. Where in wired mode of communication, give higher quality in data streaming, the installation and maintenance is expensive. Hence, wired communications are economically not preferred. For the transmission of captured data, hence wireless mode of communication is more suitable. In wireless mode of communication, it is required to transfer the data in open wireless channel, communicating with allocated resources.

However to transfer the data to a long range transmission, high resources are required. The allocated power and spectrum are to be governed, so at achieve highest degree of accuracy and processing efficiency in such application. With the objective of governing video condition various approaches were developed in past. To achieve the objective of error resilient coding in video streaming a rate distortion optimization (RDO) using the SSIM metric was proposed in [1]. This approach optimizes the wireless video streaming operation. This approach defines a Lagrange optimization for encoding to achieve the objective of error resilience coding in video streaming. The approach improves the conventional model of SSE-based error-resilient RDO for wireless video streaming. Thought the approach of SSIM-RDO gives a simple and higher degree of accuracy on coding, the work was more focused towards error control and no observation was made on transmission model. Where in the effect of video quality also depends on the network considered and the nodes used. Wherein in such network, where long range transmission is required, hop base communication is most preferred. The significance of self-deployment and no pre-infrastructure requirement, gives the significance of using such network for video streaming. It is also required to control the video overhead and to avoid congestion so as to achieve higher throughput improving visual quality and refreshment rate. During the hop based communication, as all intermediate node work as a router, the probability of developing congestion is major in such nodes. Towards congestion control in such network, an adaptive queue mechanism based on context modelling is proposed in [2]. The proposed CA-AQM presented an approach of streaming the flow of packet based on the context of video quality. To provide the demanded QoS in H.264 video transmission over IEEE 802.11e this mechanism is proposed. The approach of queue management streaming the flow of video sequence by dropping or forwarding the data based on the queue size is presented. Wherein SSIM-RDO is a simpler approach and quite effective in video transmission, with this coding the rate optimization using network condition and data flow conditions should be considered to achieve highest level of accuracy ion video streaming. To develop this objective in this paper, a joint scheme of rate allocation with network congestion level is proposed. The joint approach of SSIM with congestion based routing is developed for IEEE 802.11e network modelling. To define the proposed work, this paper is outlined in 7 sections. Wherein section 2 outlines the past works for the effective video streaming in wireless environment and its applications in real time usage. Section 3 presents the conventional approach of SSIM-RDO approach for error-resilience coding. Section 4 outlines the proposed approach of SSIM-RDO with data flow modelling. The experimental results for the developed approach are presented in section 5. Section 6 present the conclusion developed for the present work.

II. PAST WORK

In [3] towards video streaming, a cloud process is proposed. The remote devices are interfaced to a centralized streaming unit, which performs a object detection and video condition for video streaming and streaming. The cloud approach is made to synchronize the detection of vehicle mobility under different scenarios. For such a objective in [4] a unmanned aerial vehicle (UAV) was proposed, which monitors the aerial streaming of video flow in road. The system is interface to centralized streaming unit via wireless link, where the captured video signals are captured for live streaming or for future off-line evaluation. Additional intermediate server units are used to maintain the data flow and throughput under variant channel condition. In [5] a video streaming model based on the combination of video density and video speed is evaluated. These measured parameters were used for the analysis of video congestion level and categorizes the video flow for free or congested level. The application of video streaming over the wireless channel using 3GPP, MBMS and IMS service is proposed. In [6] towards an application of video streaming, the video is monitored with the vehicle movement as well the driver driving pattern. This approach also provides a wireless mode of communication to monitor the video mobility in highway, as well it also transmit a video sequence of the driver, to monitor the driving pattern. This approach is a kind of approach towards secure video streaming. In [7] an analysis to power consumption in video coding based on constant bit rate over 3G service is presented. The approach of radio resource utilization based on control transition state machine was proposed for a 3G network. The approach of resource utilization based on power optimization to improve the power consumption over PSNR evaluated over a Nokia N900 device was evaluated. A relay based communication for video transmission was presented in [8]. The relay node selection called REDEC was developed. The approach uses the approach of receiver modelling considering the excessive collision and overhead due in exchange of video packets at high frequency. To achieve a higher precision data processing under distortion effects, in [9] a framework for video communication, streaming and analysis is proposed. The natural distortion of haze effect is studied. To improve the visual quality of the video sequence at the streaming location, new filtration process was developed. The simpler, yet effective approach of median filtration with histogram approach is proposed. To develop on to hardware level, FPGA modelling is also made out. In [10] a source rate control approach is proposed. The video streaming is controlled over a wireless channel by restoring on a reduced reference (RR) quality estimation approach. The process extracts the content feature of the video sequence, which are transmitted with the video sequence. The transmitted features are used to analyse the quality of video received. The source rate is then control to achieve the objective of higher throughput and visual quality. In [11] an approach for live streaming of video sequence for video streaming was developed. In the prospective approach of streaming IEEE 802.11p for automotive video applications were identified. The test bed simulation for the stated approach illustrates a higher visual quality under different channel conditions. In [12] an integrated model of multi-source video streaming for video streaming is proposed. A block partitioning of the video data into macroblocks for efficient streaming is proposed. The approach analyzes the impact of different multipath condition for data streaming over VANET, taking various network parameters under consideration. To add intelligence to the video streaming operation in [13] a neuro-fuzzy modelling for MPEG-4 video transmission over IEEE 802.15.4.zigbee wireless device operating at 2.4 GHz with data rate of 250kb/s was proposed. The approach defines two scheme for streaming input and output of the data storage in video regulation. The approach controls the bit rate coding for video application to overcome the picture loss quality in MPEG-4 video coding over zigbee application. In [14] a real time vision system, for automatic video streaming was presented. The system was designed for automated communication and processing of images from the pre-calibrated devices. The approach was developed under the framework of TRAVIS (Video Visual streaming) project work. The streaming unit is defined to provide target ID, position, velocity and classification of video movement for video streaming. During the process of video streaming, various routing protocols were developed, which are tailored for vehicular Ad Hoc networks (VANET). In the process of packet exchange in such network, the IEEE 802.11 format of communication is used which communicate using the RTS/CTS model of communication. To develop a efficient communication model in such network, in [15] a rate distortion model for video streaming video streaming is proposed. A shadowing model of DCF for a highway segment is proposed, which is designed for evaluating the density of the road video. Towards the seamless transmission in [16] a new protocol based on network-based localized mobility management group working of internet engineering task force was proposed. The protocol minimizes the issues for network switching under mobility scenario. The protocol focus on mobility management to minimize latency, jitter, packet loss in video streaming application. In [17] a delay optimized network coding for video streaming is proposed. The wireless network, increase the system throughput by mixing the packets from different flows into a single packet increasing the information content per transmission flow. The streaming of intermediate delay factor to optimize network coding for video streaming in video streaming application was proposed. A cross layer approach of video streaming over wireless network is presented in [18]. The approach is defined for IEEE 802.11 WLAN network. The scheme proposed a rate adaptation for data link and physical layer, whereas the quality adaptation in application layer. The rate adaptation is used for the adjustment of the allocated rate, whereas the quality adaptation scheme control the video quality offered. In [19] an approach of cross layer modelling of video streaming over wireless network is presented. The model is developed for centralized and distributed streaming of video streaming under automated streaming application. In [20] the approach of hierarchical wireless network with multiple layer cells is proposed. It is been focused for optimizing the load balancing over different cells to minimize frame drop ratio. The user assignment problem in cell communication is focused. The adaptive adjustment of the user detection results in improvement of various real time loads balancing characteristic in wireless video streaming. In [21] to improve the video quality a joint selection of quantization offset is presented. The statistical distribution of transformed coefficient in encoded video sequence is developed using a Laplacian model. The multi-level optimization solution results in optimal path selection for lower path failure probability. The seamless transmission of video sequence using H.264/AVC was developed. In the mode of burst mode communication to achieve high throughput communicating with error occurring during the channel fading process is proposed in [22]. The effective channel bandwidth and current channel state is analyzed for the automatic repeat request error streaming operation. The constraint of buffer and end-to-end delay is considered as a trans-coding parameter. It is illustrated that the approach of trans-coding results in improved video picture quality. A scalable mode of video coding is presented in [23]. The approach of vehicle streaming, in multi-Hop communication model is presented. The error of such coding is governed by a video quality of experience (QoE) metric. The approach obtains a good tradeoff between the video quality and the latency of startup.

III. SSIM-RDO VIDEO STREAMING [1]

For error resilience coding in video streaming the approach of SSIM-RDO is used.In [1], the approach of SSIM-RDO based error-resilient scheme for H.264/AVC is proposed. To improve the wireless video streaming performance, a numerical relation was derived through Lagrange method to obtain minimum distortion. The SSIM is used as a distortion metric, and a low- complexity Lagrange multiplier for SSIM-based RDO video coding for error-free coding is derived initially. The SSIM-based decoding for distortion minimization is included at encoder to formulate error resilient video coding. Further, the Lagrange multiplier is theoretically derived to optimize the encoding mode selection in the error-resilient RDO process.

A. SSIM-based RDO Formulation Based on SSE-Based RDO

In the video processing, the optimal encoding mode can be determined by reaching the best trade-off between the coding bits amount and the obtained video quality. This problem can be modeled as [1];

$$\min_{\{m\}}\{D\}\ subject\ to\ R \le R_c \quad (1)$$

Which indicates that the video encoder should minimize the apparent distortion ‘D’ with the number of encoding bits ‘R’, following the constraint of bit rate of ‘R_c’ by appropriate selection of a encoding mode ‘m’. In video streaming, the Lagrange optimization approach is used to make the objective free as;

$$\min_{\{m\}}\{J\}\ with \quad J = D + \lambda.R \quad (2)$$

Where, ‘J’ is Lagrange cost and ‘λ’ is the Lagrange multiplier for RDO. Basically SSE and SAD are used to optimize the RDO process. Wherein, SSIM is used to measure the distortion unit. The SSIM index is derived to independently compute the relativeness of local luminance, contrast and structure between an original image and a distorted image. The SSIM index is calculated in windows with different block sizes for the two images. Given two images windows of block x and y, the local SSIM index of the two image is defined as;

$$SSIM(x,y) = \frac{(2\mu_x\mu_y + C_1)(2\sigma_{xy} + C_2)}{(\mu_x^2 + \mu_y^2 + C_1)(\sigma_x^2 + \sigma_y^2 + C_2)} \quad (3)$$

Where ‘μ_x’, ‘σ_x’ and ‘σ_{xy}’ are the mean, standard deviation and cross correlation between the two image windows respectively. ‘C_1’ and ‘C_2’ are used to maintain the stability when the means and variances are near to zero. When the coding distortion is measured using SSIM-based distortion, the Lagrange optimization scheme from (2) is then modelled as;

$$\min_{\{m\}}\{J\}\ with \quad J = D_{SSIM} + \lambda_{SSIM}.R \quad (4)$$

Where 'D_{SSIM}' denotes the SSIM-based distortion and 'λ_{SSIM}' is the Lagrange multiplier for the SSIM-based RDO. As the distortion is observed in terms of SSIM metric, 'λ_{SSIM}' should be suitably chosen to obtain the optimal trade- off between the coding bits amount and the SSIM-based distortion. Thus, the core problem for SSIM-based RDO is to determine the SSIM-based Lagrange multiplier 'λ_{SSIM}'.The Lagrange multiplier 'λ_{SSIM}' for SSIM based RDO can be modelled by just scaling the 'λ_{SSE}' for a fixed scaling factor 'f'. Then, the SSIM-based Lagrange multiplier λ_{SSIM} can be obtained as;

$$\lambda_{SSIM} = \frac{-D_{SSIM}}{R} = -\frac{\left(\frac{D_{SSE}}{f}\right)}{R} = -\frac{1}{f}\cdot\frac{D_{SSE}}{R} = \frac{\lambda_{SSE}}{f} \quad (5)$$

Thus, for the SSIM-based Lagrange optimization process, it can be modelled by only scaling the existent SSE-based Lagrange optimization formulation with 'f' as;

$$\min_{\{m\}}\left\{\frac{\lambda_{SSE}}{f}\right\} with \quad \frac{\lambda_{SSE}}{f} = \frac{D_{SSE}}{f} + \frac{\lambda_{SSE}}{f}.R \quad (6)$$

B. SSIM-based Error-Resilient Video Coding

In this approach, to provide the network optimality, the video coding layer (VCL) and the network abstraction layer (NAL) are designed for H.264/AVC video coding standard. The VCL operates for the compression process wherein, the NAL operates at the network level to provided proper allocation of resources under network level. For wireless communication, the transmission channel is time-varying and erroneous in nature. For the minimization of error during signal propagation, independent channel model is used. Knowing the bit error rate (BER) of the transmission channel, the packet loss probability 'ρ' for a NAL unit containing 'L' bits is related as;

$$\rho = 1 - (1 - ber)^L \quad (7)$$

During the encoding process, the video stream are divided into frame slices represented as $s_{n,m}$. For the mth slice in the nth frame, the BER is defined by, $ber_{n,m}$ which is the channel BER for the transmission of m_{th} slice of nth frame and $\rho_{n,m}$ is the packet loss rate for slice $s_{n,m}$. The Lagrange multiplied 'λ_{SSIM}' was adjust to achieve the objective of error resilient to minimum. The Lagrange multiplier was developed based on distortion metric 'D_{SSIM}'. Since the distortion estimation is conducted at the encoder end, a module in the encoder is added, to simulate the decoding process with the help of the acknowledge message which informs the encoder whether the transmitted packet is received by the receiver or not. For a acknowledge message 'nr' received by the encoder while encoding the nth frame the encoding information is stored and the added decoding unit decode the 'nr' frame and get the expected decoded frames from nr^{+1} to n^{-1}frame. Given the decoded reference frames or the expectations of the decoded reference frames, the pixel values were obtained.

IV. FLOW CONTROL SSIM-RDO STREAMING (FL-SSIM-RDO)

Where in SSIM-RDO approach is observed to be very effective in video streaming, when applying to the video streaming over a wireless channel for video streaming, it is to be observed that the rate of transmission and the visual quality need to be large for better streaming . In the approach of video streaming, due to remote communication, the allocated resource of power and spectrum will be limited. In such a case proper utilization of resources and intermediate node support is highly required to achieve greater performance. Wherein observing for error resilience coding, the SSIM-RDO approach is simpler, and effective for video streaming, however without control of flow this resilience may not result in effective visualization at streaming units due to incurred latency issue. Hence with this error resilience, a rate allocation is required so as to achieve the objective of higher throughput with better visualization. While observing the SSIM-RDO approach, the SSIM-based video coding considered the SSIM as distortion metric between the original and received videos. The SSIM has given the structural similarity between the original and recovered videos. Higher the SSIM, higher the quality of service or vice versa. Where in error resilience coding is required to improve the visualization, a congestion in the channel will degrades it performance. Hence the access streaming is also needed with error streaming. With this objective in [2] a queue based management scheme to control the rate of allocation was proposed. As in the proposed work, each intermediate node is considered to be a router, each node encounters multiple video at this node and observe heavy congestion. To overcome this issue, queue management is applied. The cross layer optimization of video stream video at router level was proposed in [2]. The approach of coding was introduced at network abstraction layer (NAL), where the queue based congestion streaming following Active Queue Management (AQM) and relative quality of Service (QoS) is mapped to schedule the rate of video flow. The cross layer approach called as CA-AQM process on measured queue length and derives the packet enqueue or doping probability based on receiving data video. Wherein at the Video coding layer (VCL) the video source is blocked into slices and passed to NAL for rate allocation. The method computes the drop probability'd (t)' as;

$$d(t) = \begin{cases} 0; & q(t) < min_{th} \\ 1; & q(t) > min_{max} \\ max_p \times \dfrac{q(t) - min_{th}}{max_{th} - min_{th}}; & otherwise \end{cases} \quad (8)$$

Where min_{th} is the minimum queue threshold and max_{th} is the maximum queue limit.

In the approach of cross layer modelling (CA-AQM), the drop probability is modified as; $d(t) = 1 - \phi^{-p(t)}$, Where $p(t)$ The price in period t and ϕ constant value 1.001 defined as REM (random exponential marking).

The price is updated from time to time according to the average queue length, and the input rate and the output rate of the queue. The CA-AQM approach, controls the flow of video by accepting or dropping the video packets based on the probability index p (t) and the importance of the packet to drop. The price is incremented if the input rate exceeds the output rate, and decremented otherwise. The suggested streaming algorithm of CA-AQM [2] is as outlined below;

Algorithm: CA-AQM algorithm

Calculate the average queue length $q_1(t)$ in period t;
Receive packet σ;
Calculate the drop probability d(t);
Randomize a number μ;
If (d(t)<μ)
  En queue packet σ;
else
  Drop the packet σ with $U = arg_{i\in[l,L]} min U(i)$;
  En queue packet σ;
End if
Where,
$U_i(t)$ The important index of the i packet in the queue in period t

Though the above stated approaches of SSIM-RDO and CA-AQM approach are developed as rate control to video quality assessment, they lack in optimal rate streaming based on queue factor or error factor. As in [1] the SSIM-RDO the node property is neglected in [2] error factor is neglected. Hence, it is required to have a integrated approach of rate streaming with error control for better quality visualization in video streaming . With this objective in this work, a flow control model based on improved Queue management and error metric is derived.

C. Flow Control Based on Congestion Level

From the queue management approach as outlined in [2] it is observed that, the congestion level is governed at 2 levels and the dropping probability is then defined as of '1' or '0' as presented in (11). Wherein it is observed that video flow under min_{th} is considered as non-congestive zone and above max_{th} is considered as congestive zone. The region in between these two limits are taken as a random zone, where the packets are randomly been enqueued or dropped based on the probability of dropping 'p (t)'.

However, in consideration with error resilience and video flow, a Flow control based video streaming is proposed termed as "FL-SSIM-RDO". The suggested approach is defined as; under the constraint of node congestion apply queue management, the allocated rate is defined as

$$R_{alloc}(t) = \begin{cases} R(t) + \Delta t \, if \, Q_{current} < Q_{min} \\ R(t) + (\Delta t - d(t)) if \, Q_{min} < Q_{current} < Q_{max} \\ R(t) - \dfrac{R(t)}{d(t)} if \, Q_{current} \geq Q_{max} \end{cases} \quad (9)$$

Where,
$R_{alloc}(t)$ = Allocated data rate
$R(t)$ = full rate
Δt = Incremental data rate
$Q_{current}$ = Current length of queue
Q_{min} = Minimum limit of queue
Q_{max} = Maximum limit of queue

In the above equation (12), the allocated data rate is varying with respect to node congestion level. If the current queue length is minimum level the data will be allocated with an increment of 'Δt'. If the current queue length is in between the minimum and maximum levels, the allocated data rate depends on the dropping probability according to eq. (11). Similarly, if the current queue length exceeds the maximum queue length which represents the congestive level, the allocated data rate will vary according to dropping probability. i.e., the allocated data rate will be too less.

V. EXPERIMENTAL RESULTS

To simulate the proposed approach a video compression algorithm at video coding layer (VCL) is developed. At the video coding layer the captured video is processed for compression. The motion elements are extracted using a recurrent block matching approach. The extracted motion vectors are compressed using entropy encoder and stream out to Network abstraction layer (NAL). At each node the NAL computes the current congestion level and computes the allocable rate of transmission in consideration with error factor as briefed earlier. To evaluate the proposed approach a subjective and objective analysis of the developed system is analyzed with the approach of SSIM based Rate allocation method. The performance of the proposed approach were evaluated with respect to SSIM, throughput, node overhead, end-to-end delay and allocated data rate. The illustration of the suggested approach is as presented below;

Fig 1: Testing Video sequence

Figure illustrates the captured sequence from a video codec. The communication unit was installed at the existing video light poles with a rotation of 360^0 orientation. The video are captured from a high resolution device at a frame rate of 120fps, with 256 x 512 pixel resolution.

Fig 2: Processing frames for the Captured video Sequence

The recovered sample for the developed approach of SSIM-RDO and FL-SSIM-RDO are shown in figure 6 and 7 respectively. The observed visuality of the two methods are relatively equal, as the error resilience coding of SSIM is used. However the proposed approach of FL-SSIM-RDO provides the video more faster than SSIM-RDO for observation. The observed performance for the developed approaches is illustrated below.

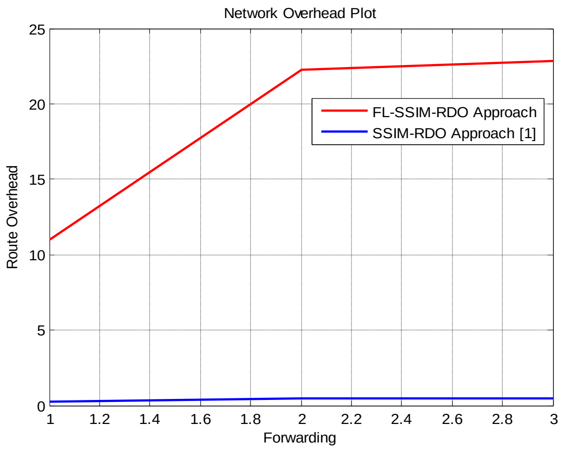

Fig 3: Network Overhead with Packet Forwarding

On the process of communication, data are forward for the path selected from communication unit to streaming unit via intermediated nodes as shown in figure 3. During the process of data exchange the packets are buffered and based on the computed congestion level, packets are forwarded. To transfer the packets per node forwarding data rate is computed and based on the congestion level and the SSIM factor, allocated data rate packets are released or queued up in the node. The queued up packets intern builds an overhead in the network. This overhead is defined as the number of packets queued up for processing at each node. With the forwarding of packets from source to destination, at each intermediate node, packets are buffered; the overhead observed is presented in figure 8. Due to flow streaming with error coding, the congestion is controlled at a lower limit of buffering in contrast to buffering to a higher level of buffer queue. Due to streaming of congestion at each node level with error streaming, the overhead to such network is observed to be minimized with the each forwarding of packet.

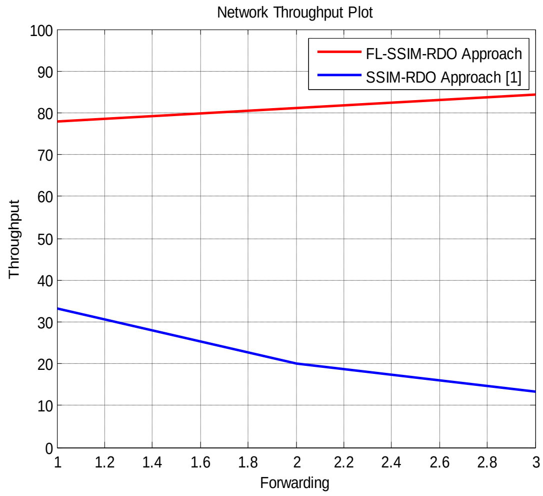

Fig 4: Network Throughput with Forwarding of Packet

The network throughout for the developed network is illustrated in figure 9. Due to higher data rate of packet forwarding in the proposed approach, it is observed that more number of packets is received and hence resulting in higher throughput in the network. The throughput is defined by the number of packets generated over packets received for an observing communication time period. The throughput is observed to be higher for the first 1-hop forwarding, as in such case the buffering is observed to be minimized and hence allocated data rate is improved. However as the number of packet forwarding increases the congestion level increased resulting in decrement of throughout, which is observed to be higher and increased in case of propose FL-SSIM-RDO approach.

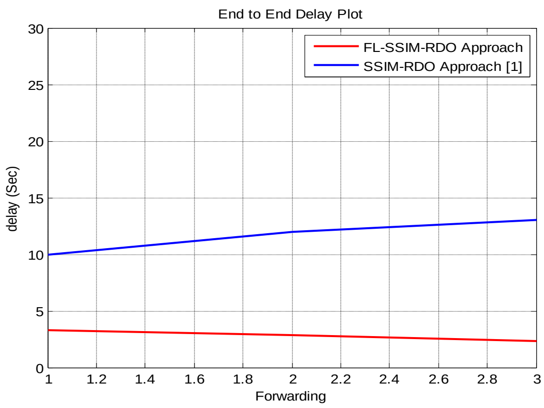

Fig 5: Observed Delay for the Simulated Network with Node Density of 30

The observed end-end delay factor for the developed system is presented in figure 10. The delay for the SSIM rate control approach is observed higher than the FL-SSIM coding, as the buffering of data at each node is minimized at the node level the packets are released faster. The congestion level in such coding is developed at a slower rate, wherein packets buffered in queue based on SSIM approach is high, which builds the packet forwarding delay at each node. The constant rate allocation, irrelevant of the congestion factor induces this delay per node.

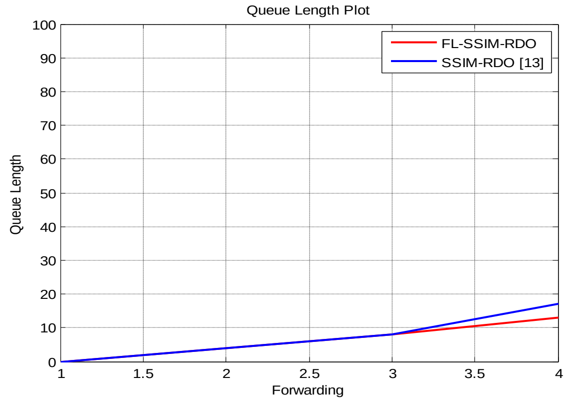

Fig 6: Buffered Q_Length with Increase in Forwarding Packets

The Q lengths are measured as the volume of data packets buffered with increase in forwarding of data packets. It is observed that, the Q length of buffering is lower for FL-SSIM coding due to optimal allocation of data rate. The queuing is however observed to be equal in the initial communication phase and gradually increased with forwarding of packets.

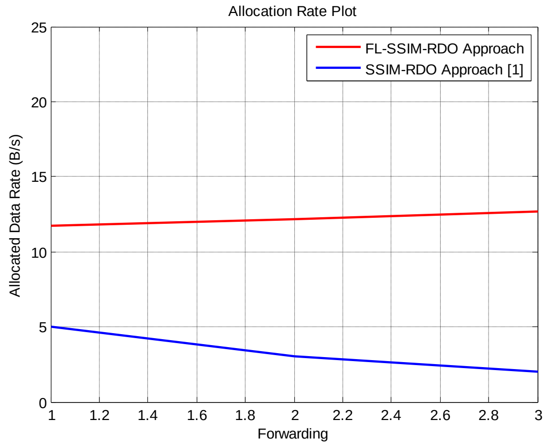

Fig 7: Allocated Data Rate with Forwarding of Packet

The allocated data rate for the developed methods for the simulating network is shown in figure 12. The nodes are communicated for the given data packet and each level of buffering these nodes compute the allocable data rate for forwarding. This forwarding results in proper control of congestion and hence results in efficient network performance.

VI. CONCLUSION

An integrated model of video congestion metric with SSIM-RDO is proposed. The approach of error resilience with high video flow is presented. A dynamic data flow control with probabilistic route density is developed, to control the flow of captured video data over a multi-hop IEEE 802.11e network model. In this approach, the video quality improvement is achieved with error resilience coding using, SSIM factor. The error resilience coding is then improved for high throughput using data flow streaming using rate allocation approach. From the experimental results an improvement in system throughput and video quality was observed.

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