Adaptive Rate allocation Approach to improve video Quality in IEEE 802.11e

Dr H.S.N. Murthy
Retired Professor, Department of ECE, JNTUCE, Hyderabad, Telangana, India

Naveed Farhana
Lecturer, Department of Computer Engineering, College of CS&IT, Taif University
farhana_naveed@rediffmail.com

***Abstract*— IEEE 802.11 wireless local area networks (WLANs) are now a days used extensively in our daily life, which provides the wide variety of multimedia applications like IPTV, video Conferencing, VOIP etc. IEEE 802.11e was developed by IEEE working group as an enhancement to IEEE 802.11 and it provides a prioritized and good service differentiation but could not able to provide the desired quality of service ,and especially improving the quality of video is still considered as a challenge , Now a days active queue management is a current research topic of this era and structural similarity index(SSIM) based rate-distortion optimization (RDO) has been proved to be an active tool for H.264/AVC to support and enhance the perceptual video coding standard in WLAN, In our proposed research with jointly consider the active queue management with the SSIM –RDO to develop a rate adaptation Approach to enhance the quality of a video. A simulation study shows that RA- AQM , our proposed mechanism has improvised performance over the other two mechanisms.**

Keywords- Rate control, EDCF, delay factor, similarity index measure ,CA-AQM

I. INTRODUCTION AND MOTIVATION

Wireless communications have appreciated outstanding growth for the last few decades, as an outdated means of wired communications have proven to be insufficient in meeting the challenging needs of the end users. Installation and maintenance of the wired networks are quite expensive as it needs lots of labor and cost. As the number of end users and their needs in terms of bandwidth, speed, required for multimedia services, etc. are increasing and it is a burdensome task to frequently upgrade the network. Wireless networks offer many benefits over the wired networks. Currently a different variety of wireless technologies has been developed around the world such as Wireless local area network (WLAN), WiMAXs, Bluetooth, etc. Wireless LANs allows two or more devices to communicate using the wireless spectrum [1]. The IEEE 802.11 standard is originated by the IEEE LAN Standards Committee (IEEE 802) for WLAN [3]. Wireless Local area network (WLANs), Which operates in the range of typically less than 100 meters, which is adequate for small to the medium organization, Apartments, hospitals, etc. As WLANs operate in the unlicensed ISM bands, there can be other devices which use the same frequency band and may result to signal interference [2]. Initially, the IEEE 802.11 WLANs provided a best-effort service only, using the Distributed coordination function (DCF), and it does not discriminate between the non-real and real-time traffic such as data ,voice and video. It gives all traffic classes with equal priority in accessing the spectrum. Due to the inadequacy in providing the priority classes, a new enhance protocol called as IEEE 802.11e standard was developed called as Enhanced Distributed coordination function (EDCF)[3] and it provides a good service differentiated and Prioritized traffic classes such as data, voice and video. While Enhanced Distributes coordination function (EDCF) can improve the Quality of service of voice ,video traffic, but not adequate for the real-time multimedia applications[3], Since wireless channel conditions are time varying and wireless stations adjust itself transmission rate, according to the channel conditions by the link adaptation (LA) approach, When the wireless link condition weakens, a lower transmission rate will be adopted in order to decrease the error rate of the channel [18]. Also video traffic is also bursty in nature and considered by large packet sizes and requires maximum bandwidth requirements. WLANs are not appropriate for carrying video with a suitable QoS and it requires a special mechanism to have the required QoS, As EDCF use contention-based mechanism to gain access to the channel, i.e. stations require to contend for accessing the medium ,because the medium is shared among all the users[1,3]. As the number of stations grows, the time needed to efficiently access the medium also increase in turn, increase the time of a packet in the buffer also. An increased contention time also causes to an increased probability of collision , delay and packet loss and resulting in the probability of buffer overflow, there are may be other losses on a WLAN in various ways as transmission errors due to noise and interference present in the medium, buffer overflow due to an insufficient availability of transmission opportunities to satisfy the incoming video packets and MAC collisions due to contention for transmission opportunity[4].So it is very important to manage the transmission of the video traffic on a WLAN. Stream video traffic flow has different network requirements compared to other types of traffics in terms of bandwidth, delay, jitter, and loss to acquire the desired QoS[19].

Also the performance of video streaming are very sensitive as uncontrollable packet losses due to overflow of the buffer, Medium access collisions and transmission losses and the hostile nature of the WLAN environment contribute to screen freeze, and distortion in the audio quality and the viewer's experience would be unsatisfactory, in drastically reducing the video quality in an uncontrollable fashion. Packets lost due to collisions can be retransmitted but at the expense of a higher probability buffer overflow, buffer overflow occurs when there is insufficient capacity in the transmit buffer to accommodate the arrival of new packets to be transmitted, leads to a disastrous drop in the QoS since the packets lost due to buffer overflow cannot be recovered [14]. To deliver video packets with acceptable QoS uncontrolled packet losses should be minimized and packets dropped at the Medium Access layer (MAC) layer due to buffer overflow need to be reduced. Hence it is rather challenging to guarantee satisfactory Quality of Service (QoS) for a streaming video over WLANs [16]. The Objective of streaming the Video packets would play continuously with as lesser a delay and packet loss to offer a better quality of service(QoS). Due to the development of new coding approaches, results in the real wireless streaming. To achieve the objective of improved visual quality in streaming in wireless local area network (WLANs), and to minimize the distortion in video quality [17], a rate distortion optimization (RDO) using the SSIM metric was proposed in [17], Where in the effect of data quality also depends on the network considered and the nodes used. It is also required to control the control congestion so as to achieve higher throughput to improve the visual quality in Wireless local area network, an adaptive queue mechanism based on context modeling is proposed in [18]. We propose a novel approach called as Rate Adaptation (RA-AQM), which controls the rate adaptation depending on the queue length of EDCF. Our proposed work, our paper is organized in 6 sections. Section 2 briefs about the past work, Section 3 presents the related work, Section 4 offers the proposed work. The simulated results are discussed in section 5 ; section 6 concludes the paper.

II. PAST WORK

Even though 802.11e EDCF can enhance the multimedia application performance to some extent, but still difficult to guarantee strict delay requirements and with packet losses which are unavoidable. To Enhance the performance of EDCF, two different types of techniques have been proposed by many different researchers, the first kind of solutions is to adjust the parameters of EDCF, such as Contention Window (CW), Transmission Opportunity (TXOP) to improve, prioritized and service differentiate services. Among them [6,7 and 8] proposes the improvement based on adjusting and varying contention window (CW) based upon the network condition to increase the Throughput, Researchers in [9,10] proposed a solution to increase the performance by prioritization and differentiated the multiple services through varying TXOP and in [11] discuss the super-slot mechanism to enhance the fairness in EDCF Protocol. The other kind of the solution to improve EDCF are cross-layer architecture. In [12], the characteristics of hierarchical coding in H.264 are taken into account. It shows that Static mapping frame I to AC [2], frame P to AC [1], and frame B to AC [0] can effectively improve the video quality in heavy loads. However, when the network loads are light, static mapping increases the delay and packet loss due to the overflow of the lower priority AC queue and leads to worsening the performance of EDCF. In [13] proposed a cross-layer Dynamic Mapping framework which dynamically maps MPEG-4 video stream to respective AC queues according to types of video frames used and network conditions. The saturated network loads and lower priority of video packet, the greatest opportunity for the packet to be mapped into a lower priority queue. Although, non-video services are ignored. Because the high priority queue, i.e., AC [2] is only accessed by video, the lower priority AC queues will be occupied by non-video content and lead to overflow of the video stream. To overcome the problems stated in [12, 13], various Active Queue Management (AQM) approaches[15,16] were proposed, to provide the satisfactory QoS by reducing the packet loss. The key challenge for video stream delivery quality in IEEE 802.11 wireless LANs is to maintain the required video quality with controlling the packet loss. with the development of AQM mechanism in [14,15], a Context-Aware AQM [18] approach was proposed which provides better QoS by dropping the packets based on their context. with the objective of improving the quality of stream video were developed.

With the fast deployment of hierarchical coding technology, a resilient error coding in data streaming , a rate distortion optimization (RDO) using the SSIM metric was proposed in [17], which optimizes the wireless streaming using a Lagrange optimization, and this approach improves the conventional model of SSE-based error-resilient RDO for wireless video streaming . In [19] an integrated model of multi-source video streaming for video streaming is proposed. A block partitioning of the video data into Macro-blocks for efficient streaming is proposed. The approach analyzes the impact of the different multipath condition for data streaming over VANET, taking various network parameters under consideration. In the process of the packet exchange in such a network, the IEEE 802.11 format of communication is used which communicate using the RTS/CTS model of communication. To develop an efficient communication model in such network, in [20] a rate distortion model for video streaming video streaming is proposed. In [22] a delay optimized network coding for video streaming is proposed. A cross-layer approach of video streaming over wireless network is presented in [21]. The approach is defined for the IEEE 802.11 WLAN network. The scheme proposed [23] a rate adaptation for the data link and physical layer, whereas the quality adaptation is considered in the application layer. The rate adaptation is used for the adjustment of the allocated data rate, whereas the quality adaptation scheme controls the video quality offered.

III. RELATED WORK

A. Enhanced distributed coordination function (EDCF)

EDCF [3,4] is a feasible scheme defined in IEEE 802.11 standard is developed to fulfill the need for providing QoS in Wireless LAN by supporting 8 users priorities ,which are mapped to 4 Access categories (AC_i) as AC_VO (for voice traffic), AC_VI (for video traffic) , AC_BE (for best effort traffic) and AC_BK (for Background traffic), Each (AC_i) within every station behaves like a virtual node independently contents for access of the medium. The back- off process is also carried out individually after detecting the medium is ideal for a time equal to arbitrary interface space (AIFS). Each (AC_i) uses (*AIFS* [ACi]), CWmin [ACi] and CWmax[ACi] and Persistence Factor(PF) instead of the DIFS time, CWmin and CWmax of the DCF. The conflicts between the virtual nodes within each node are resolved by granting access to the higher priority class with EDCF , After each successful transmission of ACi the corresponding CW_i will be set to CW_i min ,Once the transmission fails CWi will be calculated as

$$CW_i = \min (CW_i \max, CW_i * PF)$$

After waiting for Arbitrary inter-frame space(AIFS) period. Each back off timer is set to a random number from (1 , CWi +1) with the unit of time slot.

B. Active Queue Management (AQM)

It is a mechanism to support congestion control at the station Queue buffer; the active queue management is a recent research topic. The fundamental idea of AQM is to avoid congestion before the buffer is completely full by dropping the packets in advance [14,15and 16]. Most popular AQM techniques are queue-based such as the random early detection (RED) algorithm. In [16] , observers the average queue size and dropping of packets based upon the statistical Probability. If the buffer is almost empty, then all incoming packets are buffered in. As it grows, the probability of reducing an incoming packet increases too. When the buffer is full, the probability has reached one ; all incoming packets are discarded as

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

There are many RED variants were developed such as WRED etc ,which falls the packets according to the packet type, The PI (Proportional Integral) controller ,uses the information of the queue size to embrace the steady value of queue length to the specif ed reference value [15]. It procedures is the favorite idea of control theory.

Context-Aware Active queue Management (CA-AQM) : The cross-layer optimization of the video stream at router level known as in **CA-AQM** was proposed in [18]. The method of coding was presented at the network abstraction layer (NAL), where the queues are mapped to schedule the rate of a video stream. This cross-layer approach called as CA-AQM process on measured queue length and derives the packet enqueue or dropping 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. In the approach of cross-layer modeling (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 price index p(t) and the context of the packet .The price p(t) is incremented if the input rate goes beyond the output rate, and decremented otherwise. The Algorithm of CA-AQM [18] is as outlined below figure below.

```
Calculate the average queue length q1(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
  En queue packet σ;
   End if
   Where,
Ui(t) The important index of the i packet in the queue in period t
```

Figure 1: CA-AQM Algorithm

C. SSIM-RDO video streaming

For the error resilience coding in video streams. the method of SSIM-RDO is adopted, a based error-resilient scheme for H.264/AVC video flows is proposed in [17]. The performance of the video streaming can be increased by a numerical equation using Lagrange method to achieve minimum distortion. The structural similarity index (SSIM) can be used as a distortion metric, and a low- complexity Lagrange multiplier for SSIM-based RDO video coding for error-free coding is derived first. The SSIM-based decoding for distortion minimization is built-in at the encoder to frame error resilient video coding. Additionally, the Lagrange multiplier is being calculated theoretically with optimize the encoding method in the error-resilient RDO process.

SSIM-based RDO –Conventional Method

To improve the performance of the many coding standards are used such H.264 coding for video processing, the optimal encoding method may be decided by reaching the best trade-off between the coding bits amount and the achieved video quality. This statement can be represented in the form as equation (1) ;

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

Which shows that the video encoder must minimize the observed distortion 'D' with the number of encoding bits 'R', following the constraint of bit rate of 'R_c' by suitable choice of an encoding mode 'm'. In case of video flow, the Lagrange optimization approach is used to make the procedure independent as;

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

Where 'J' is Lagrange cost and 'λ' is the multiplier called as Lagrange multiplier for Rate Distortion Optimization (RDO). Basically SSE and SAD are used as an optimization method, Whereas, SSIM is a unit to measure distortion. In the RDO Approach, The SSIM index is used as a unit to compute independently, the dependence of local luminance, contrast and structured similarity between an original image and a distorted image. The SSIM index is computed in windows of different sizes for the resulting images. Now, consider the two images in a window of block sizes x and y respectively, the SSIM index of the two video is well-defined as;

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

The symbols.'μ_x', 'σ_x' and 'σ_{xy}' are stands for mean, standard deviation and cross correlation among the given two image windows respectively, and the constant 'C_1' and 'C_2' are used to conserve the stability when the means and variances becomes approximately equal to zero. When the video coding in distortion is measured using SSIM-based distortion, the Lagrange optimization scheme from (2) is then demonstrated as;

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

Where 'D_{SSIM}' and 'λ_{SSIM}' are the symbols used to characterize the SSIM-based distortion and the Lagrange multiplier respectively , for the SSIM-based Rate Distortion Optimization (RDO). As the distortion is measured in the form of SSIM unit, 'λ_{SSIM}' must be appropriately selected to achieve the optimal trade- off between the coding bits and the SSIM-based distortion. Therefore, the main problem for SSIM-based RDO is to measure the SSIM-based Lagrange multiplier.'λ_{SSIM}'.

IV. PROPOSED ALGORITHM

In this paper, a new rate adaptation , mechanism is proposed to enhance the video stream transmission quality over IEEE 802.11 WLAN[5] without disturbing the existing Enhanced Distribution Coordination function (EDCF) protocol ,which is used by the stations to contend for the wireless medium , In which structure similarity-Rate distortion optimization (SSIM-RDO) method used which is found to be very effective in video streaming, when transmitting over a wireless channel, and also it is observed that the rate of transmission and the visual quality required to be better for video streaming. Wherein witnessing for error resilience coding, the SSIM-RDO approach is simpler, and real for video streaming[5], on the other hand, without the flow control, resilience approach may not result in quality visualization at streaming divisions due to experienced latency issue. Hence, with the error resilience coding, a rate allocation is required to apply for obtaining the high throughput with better visual quality. Here, SSIM is used as a metric to measure the difference between the original and recovered videos, a high value of the the SSIM, higher the quality of the video or vice versa. Where, error resilience coding is necessary to enhance the video quality, but congestion in the network and interference in the medium degrades its performance and hence the quality of service, in [17] the similarity-Rate distortion optimization (SSIM-RDO) the node property is neglected in [18] error factor is ignored. Hence, it is required to have a Joint approach of rate streaming with error control for better quality visualization in video streaming. Based upon this objective , our proposed mechanism depends called as rate adaptation Approach.

From the queue management approach as outlined in [18], it is observed that the congestion level is controlled at the two levels and the dropping probability is then well-defined as of '1' or '0' depending upon the queue length(15). Wherein it is witnessed that video flow under min_{th} is considered as non-congestive and above max_{th} is considered as congestive. The area in between these two limits is considered as Otherwise, where the packets are being enqueued or dropped based on the dropping probability 'd (t)'.

However, in consideration with error resilience and Queue management, an adaptive Rate control based on video streaming is proposed termed as "RA-SSIM-RDO". The Proposed approach is presented for the video Queue as , the allocated rate can be represented as

$$R_{alloc}(t) = \begin{cases} R(t) + \Delta t \quad if \ \ Q_{current} < Q_{min} \\ R(t) + \left(\Delta t - d(t)\right) \quad if \ \ Q_{min} < Q_{current} < Q_{max} \\ R(t) - \dfrac{R(t)}{d(t)} \ if \ Q_{current} \geq Q_{max} \end{cases} \tag{5}$$

Where,

$R_{alloc}(t)$= Allocated data rate

$R(t)$ = full Date rate

Δt = increment in data rate

$Q_{current}$ = Present Queue length

Q_{min} = minimum thresold

Q_{max} = maximum thresold

In equation (5), the allocated data rate is adjusted with respect to congestion occurred on the network. If the current queue length is below minimum threshold value, the queue length will be an incremented with 'Δt'. If the current queue length is between the minimum and maximum levels, the allocated data rate will be varied depending on the dropping probability d(t) with respect to eq.(5),and if the current queue length goes beyond the maximum queue length, the allocated data rate is adjusted again with respect to the dropping probability d(t). i.e., the allocated data rate will be too low.

Refer to the eq.(2), the data rate allocated is variable , It also depends the of Lagrange multiplier in eq.(3), can be modified as

$$\lambda_{SSIM-FL}{}' = \frac{DSSIM(R_{alloc}(t))}{R_{alloc}(t)} = \frac{(1-\rho_{n,m}.SSIM\left(b_{n,m,k}, b_{n,m,k}^{e_c}\right) - (1-\rho_{n,m}).SSIM\left(b_{n,m,k}, b_{n,m,k}^{n_l}\right))}{R_{alloc(n,m,k)}(t)} = \frac{\left(\rho_{n,m}.SSIM\left(b_{n,m,k}, b_{n,m,k}^{e_c}\right)\right)}{R_{alloc(n,m,k)}(t)} + \frac{((1-\rho_{n,m}).SSIM\left(b_{n,m,k}, b_{n,m,k}^{n\_l}\right))}{R_{alloc(n,m,k)}(t)} \tag{6}$$

Nearly represented as

$$\frac{(\rho_{n,m}.SIM\left(b_{n,m,k}, b_{n,m,k}^{e\_c}\right))}{R_{alloc(n,m,k)}(t)} = -\frac{((1-\rho_{n,m}).SSIM\left(b_{n,m,k}, b_{n,m,k}^{n_l}\right))}{R_{alloc(n,m,k)}(t)} \approx \lambda_{SSIM} \tag{7}$$

From the equation (7), it can be noted that, the Lagrange Multiplier be determined by the rate allocated for a particular n_{th} slice in the m_{th} frame. Thus, by varying the allocated data rate, the Lagrange multiplier is also varied and delivers an efficient error resilient and queue management. From the eq.(8), the SSIM-based Lagrange multiplier λ_{SSIM} in eq.(5) is represented shown below;

$$\lambda_{SSIM\_FL} = \frac{-D_{SSIM}}{R_{alloc}(t)} = \begin{cases} \dfrac{-D_{SSIM}}{R(t)+\Delta t} & if \quad Q_{current} < Q_{min} \\ \dfrac{-D_{SSIM}}{R(t)+(\Delta t - d(t))} & if \quad Q_{min} < Q_{current} < Q_{max} \\ \dfrac{-D_{SSIM}}{R(t)-\dfrac{R(t)}{d(t)}} & if \; Q_{current} \geq Q_{max} \end{cases} \tag{8}$$

Form the eq.(8); it is shown as the adjustment of Lagrange multiplier in Lagrange optimization is determined by an allocated data rate. And the measured distortion DSSIM and the Lagrange multiplier $\lambda_{SSIM-FL}$, the Lagrange Optimization scheme (2) can be adapted as

$$\min_{\{m\}}\{J\}\, with \quad J = D_{SSIM} + \lambda_{SSIM\_FL}.R \tag{9}$$

Where 'D_{SSIM}' denotes the SSIM-based distortion and '$\lambda_{SSIM-FL}$' is the Lagrange multiplier for the RA-AQM based RDO. As the distortion is noted with SSIM metric and the allocated rate is optimized for the video queue with the dropping probability to obtain the required video quality.

```
Calculate Queue Length Qcurrent at 't'
    if (Qcurrent < Qmin )
       d(t) =0
        Ralloc(t)=R(t)+Δ(t)
    if ( Qmin < Qcurrent < Qmax )
       Calculate d(t)
       Ralloc(t)=R(t)+ [Δ(t)- d(t)]
    else
    if ( Qcurrent > Qmax)
      Ralloc(t)=R(t)+ R(t) / d(t)
     endif
    end
    end
```

Figure 2: RA-AQM Algorithm

V. SIMULATION RESULTS AND DISCUSSION

To show the performance of rate adaptation approach, a simulation study is conducted using the following steps ,firstly a video coding layer (VCL) has been developed to capture the video, The motion components are extracted using a recurrent block matching mechanism and are compressed using the entropy encoder and send to Network abstraction layer (NAL). At the NAL, current congestion level is computed and the allocated rate is varied with respect to the queue length and which is directly proportional to the distortion using Lagrange multiplier. In our simulation, the same video streams are taken with the different numbers, video source for the simulation used is yuv cif (260 frames) CIF format named football, all the video streams last for 10 to 20s,compare the quality of the video with different network condition .The enhancement of our approach is shown using the metrics as throughput, end-to-end delay and allocated data rate and the perceptual video quality are shown using Average PSNR.

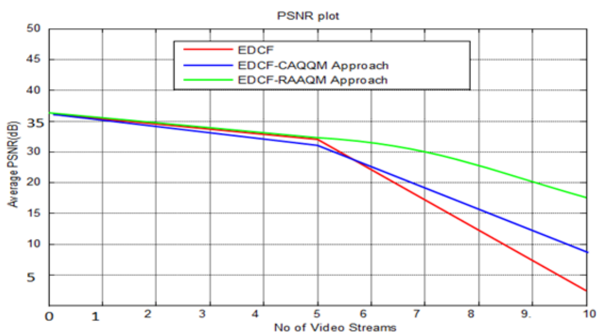

Figure 3 : Comparison of the mean PSNR in the 'Football ' sequence

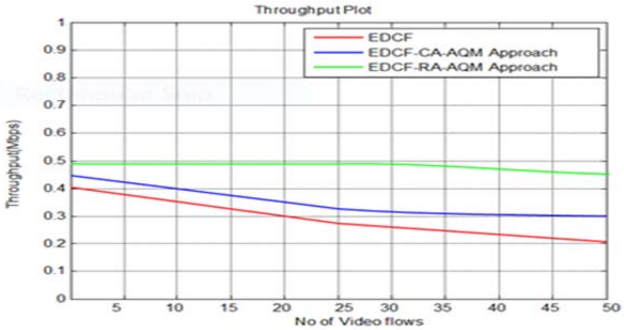

Figure 4: throughput Vs No of videos Streams

The throughout Plot for the simulated network is shown in figure 4, Due to the higher data rate of the video streams with our proposed approach, it is observed that as the number of video streams increases resulting in higher throughput compared to the conventional EDCF and the CA-AQM mechanism .

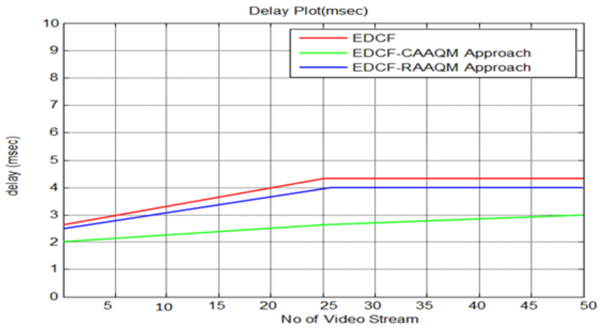

Figure 5: Observed delay Vs no of videos Streams

The observed end-end delay is shown in figure 5, which is high for the other two approaches compare to RA-AQM.

Figure 6a: EDCF

Figure 6b: EDCF+CA-AQM

Figure 6c: EDCF+RA-AQM

Figure 6 : Comparison of video quality in 'football'shots using the three methods

It is observed that the perceptual video quality as we see in figure 6(a, b and c) for the video quality is improved for RA-AQM approach compared to the conventional EDCF and CA-AQM.

CONCLUSION

A Joint approach using the SSIM-RDO to minimize the distortion due to interference and fading and Active queue management to control the congestion has been proposed to enhance the video quality. With our approach, the video quality improvement is enhanced compared to the conventional EDCF mechanism and CAAQM, which is proved using the simulation studies.

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