An Improved Video Packet Retry Adaptation in EDCA

Mehdi Gheisari

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Mehdi Gheisari[1]

department of computer, damavand branch, Islamic Azad University, Damavand, Iran

Abstract- Video transmission in wireless network is an important issue. One of the most important differences between video and ordinary data is emphasis on time and deadline so in video avoiding from delay and jitter is necessary because those causes uselessness of received data by receiver. Lossy nature of wireless networks causes requirement for retransmission of lost data thus retransmission management is very important because video data must be arrived on-time. Prior approaches in this issue mainly were based on 802.11 standard and would use statistical ways and statistical information collection for this purpose. We offer a method for retransmission of video packets on 802.11e WLANs and Enhanced Distributed Channel Access (EDCA) mechanism namely Retry Adaptation base-on Smoothed Playout Delay-base (RA-SPD). This method estimates deadline of video packets based on maximum playout delay in receiver and with regards to some effective parameters in deadline. This approach requires the knowledge of Datalink layer from Application layer so enumerate as a cross-layer approach. Results of simulations illustrate if setting of RA-SPD parameters be correct, improvement of received video quality at receiver without need of statistical information collection, in bad network condition is possible up to 4 percent rather than previous approaches and more than 10 percent rather than EDCA standard.

Keywords

Cross-layer Approaches, Video, EDCA, Deadline, Smoothed Playout Delay, Retry Adaptation

1. Introduction

Transmission of multimedia and especially video has been found many usages in recent years. In most of these usages multimedia is encountered with severe time restrictions and is used as real-time. Because of special condition of wireless environments and lack of stability in these channels, transmission of multimedia over those is encountered to fundamental challenges. Quality of multimedia is threated severely due to contention between users, different data streams and existence of noise in channel. These factors and similar factors require thinking about recourses which is said it providing Quality of Service (QoS) for multimedia transmission over wireless network usually and justify using of priority protocols such as IEEE 802.11e. But using of this protocol solely isn't able to provide desirable QoS for different multimedia applications and so usage of some methods for encounter to special conditions of wireless channel is necessary.

In video transmission over a network, important performance parameters are delay and jitter. In wireless networks other important subject is packet loss which has destructive effect on visual quality of video. Hence it is necessary thinking about solutions for packet loss and indeed error in video transmission over wireless networks in addition to delay control because these packet losses have wide effect on quality decrement of video and it causes transmission of video become very challenging.

802.11e standard and its EDCA mechanism perform a classification over packets and determine transmission priority according to it for providing QoS.

Base of QoS in this protocol is such classification over packets and powerful channel access management too. Also in EDCA there is a retransmission scheme after packet loss for encounter to this error which this retransmission perform by one of the Automatic Repeat reQuest (ARQ) error recovery techniques namely Stop and Wait ARQ (SW-ARQ). Thus one of main subjects for improvement quality of received video at receiver especially in bad wireless channel condition, is retransmission of lost packets. If these retransmissions performed correctly it can perform main role in received video quality improvement else quality decrease extremely.

For example with retransmission of packets which their deadline is bygone, time for next packets will be low too and this subject can have destructive effect over received video by receiver. For more effective retransmission other subjects are important too. One of them is amount of different packets transmission effect on received video quality.

Other is transmission rate and importance of queue which packet transmitted via it. Notice to these subjects can cause receiver video increase significantly.

Rest of paper organized as follow: first we look at prior works and matters which can be useful in context of packet retransmission will be reviewed in II. Expression of proposed mechanism is next section (III) which details of RA-SPD algorithm will be descripted in it. Then smoothed playout delay formula will be stated which RA-SPD algorithm is based on it, that is to say this formula is used for packets retransmission and parameters of this formula will be descripted in detail. Then in next paper section (IV), simulation and evaluation of mechanism will be argued and finally this paper will be ended with an overall conclusion and future work statement in VI.

2. Related Works

In context of transmission of multimedia over wireless networks specially 802.11 and 802.11e standards there are many works which mainly offer adaptive approach for encounter to special condition of wireless networks and also special requirements of video. In this section perform a short revision over some of these methods.

In [1] is proposed Content-aware Adaptive Retry (CAR) which try to prevent from retransmission of packets which have many delay and so receiver isn't able to playout they by estimation of packets deadline according to their importance with using of exist concepts in [2]. This mechanism is designed for 802.11 wireless networks and it smooth condition for on-time transmission of next video packets with dropping useless packets. Because attention to packets importance and their dependencies, packets with higher priority and importance have higher chance for retransmission and this cause relatively high improvement in quality of video in receiver specially in bad condition of channels and high amount of loss.

The same algorithm proposed again called Time-base Adaptive Retry (TAR) in [3]. Difference between this work and prior work is consideration other aspects in account and completing previous work. In TAR authors try to estimate retransmission time of each packet in two states of network separately which one of them is noisy state and another is congestion state so as a result profitability of using TAR mechanism which do act of retransmission adaptively will be apparent.

In a proposed algorithm in [4] named Content-Aware Retry Limit Adaptation (CA-RLA), is tried to using results of prior works and more precise analysis on granting packets priority, supply more effective mechanism for packets retransmissions. In [4] in addition to deadline which was previously noteworthy, also is noticed to retry-limit. This retry-limit computes and uses for each packet separately with using of image processing methods. In this research paper is said it is possible to identify delay packets before decision to retransmission in according to estimated time of next retransmission so drop those packets one step ago. Also it proposed a greedy algorithm which use efficiently from time since successfully packet transmission. In this greedy algorithm important packets will obtain more retry limit after a quick successful retransmission. CA-RLA have more complexity from other mechanism so it is difficult completely implementation it.

All three methods explained hitherto use statistical approaches and statistical information collection for finding proper packets retry counts. So there is no distinct and exact formula for deciding and also there are some presumptions for determining final retransmission count.

Addition to instances cited above there are other variety works in context of video transmission over wireless networks which each of them consider some aspects of subject. Some of them will be descripted here.

In [5] proposed a range of methods for video specs analysis and utilize those for improvement of transmitted video quality in limited condition of network and low transmission rate. For example light, location, movement vector and energy specification analysis are some of them. In this paper only limited condition and very low rate is object of paper authors so they regard to different aspects of video and perform many analysis. Then more important parts of video determined and transmitted under better preparation rather than lower important sections to overcome bad network condition and so having effective video transmission.

For instant an adaptive method proposed which in good network condition whole frames are transmitted, in 1/3 rate B frames are dropped and in very low rate and very motionless video all frames are skipped except I frames. Displacement of video frames according to their importance is one of other adaptive approaches can deployed.

In [6] authors addressed quantization step size adjustment and proposed an adaptive approach which it use from a Video Rate Control Algorithm (VRCA). The quantization step size is the main parameter that controls the compression of the video. But the VRCA is implemented as a simple feedback control loop that consists of setting an initial quantization step size, encoding part of the picture, measuring the resulting intermediate bitrate, changing the step size accordingly and then continuing with the next part of the picture. So it is possible to change bitrate for each frame independently. Although this VRCA has been designed for constant bit rate (CBR) encoding, but can also be used to dynamically change the bit rate produced by the encoder.

This approach try to achieve an optimum rate by adjusting quantization step size and sender regarding to network condition determine this amount till get desire video quality with adjusting video rate. So video rate changes are completely depend on channel condition and are adapted withal. In this approach network condition is estimated in both short and long interval by some feedback methods.

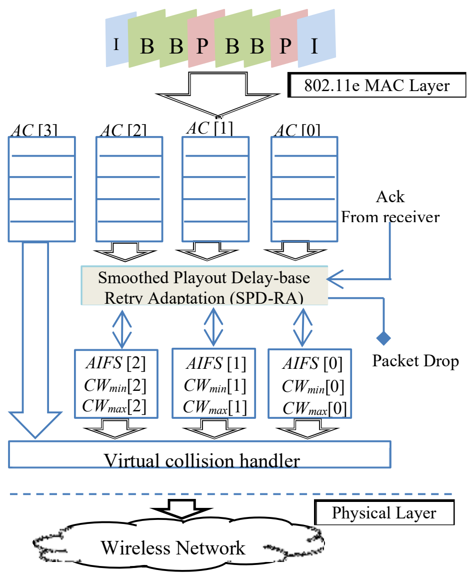

Fig.1- general scheme of SPD-RA mechanism

Proposed approach in [7] follows a retransmission algorithm in application layer and deploys packets perceptual importance and temporal dependencies for retransmission adjustment. Because this algorithm act in application layer and don't use link layer retransmission, is different from other expressed algorithm in this section roughly. However adaptation in link layer is faster usually so link layer algorithm can be more effective. Too some papers concentrate over Forward Error Correction or mixture it with ARQ. For example [8] propose an algorithm which deploys a proactive retransmission scheme for hybrid FEC/ARQ to transmission of video by feedback from receiver. If sender receives any feedback from receiver in a predefined interval, it will be informed from burst packet loss and take advisable decide.

3. Proposed Mechanism

In previous section, said prior works in packets retransmission over 802.11 standard determine retry count base on statistical information collection. Also these approaches was indifferent to real playout delay used in receiver so playout delay calculation in they is performed without having information about real playout delay in receiver. Though this issue is proper from viewpoint of having no requirement information from receiver but maybe in many cases causes no right estimation from playout delay. Whatever network condition is changed, but receiver finally set fix amount to its playout delay which it is without change along playout time. If sender can become aware from this playout delay, can use from it as maximum delay in calculations. Even sender can proposed playout delay itself to receiver. In addition sender can calculate this amount by common formula with receiver or guess it or distinct it according to application. So sender can simply calculate and enter this amount in its calculations. In [4] this amount is notated β and has some uses. Real playout delay can't be used dirctly and must adapt with network condition and traffic amount which called smoothed playout delay and so our mechanism named Smoothed Playout Delay-based Retry Adaptation (RA-SPD). In proposed mechanism before retransmission of each packet over the network, deadline of the packet is calculated according to smoothed playout delay and also parameters of each Access Category and then sender decide send the packet or no. This deadline which calculated according to smoothed playout delay is called smoothed deadline. Figure 1 depicts general scheme of RA-SPD mechanism.

Objectives of RA-SPD retransmission adaptation mechanism are:

- Prevention from ineffective retransmission.
- Sending packets while their deadline permits which cause high assurance for more important packets to be received accurately and on-time.

In [4] proposed before deciding to retransmission a packet, approximate transmission time be determined and if packet isn't received in its deadline, it be dropped. So according to mentioned objectives and proposed algorithm in [4], general algorithm of RA-SPD mechanism is:

After an unsuccessful packet P transmit do:

$T_{trans}(r, QN) = T_{cur} + T_{back}(r, QN)$

In the (r-1)'th try:

If ($T_{trans}(r, QN) + \delta > D'_{sm}\left(P_{i,j}^{(k)}, QN\right)$)

Drop $P_{i,j}^{(k)}$

In (r)'th try:

If ($T_{cur} > D'_{sm}\left(P_{i,j}^{(k)}, QN\right)$)

Drop $P_{i,j}^{(k)}$

Else

Transmit $P_{i,j}^{(k)}$

In this algorithm parameters are as follows:

- QN: queue number N
- $T_{trans}(r, QN)$: estimated transmission time in r'th retry according to queue Q parameters
- T_{cur}: current time
- $T_{back}(r, QN)$: estimated backoff time in queue Q for r'th retry
- δ: maximum propagation delay in wireless channel
- $D'_{sm}(P_{i,j}^{(k)}, QN)$: packet $P_{i,j}^{(k)}$ deadline according to queue Q parameters

In $P_{i,j}^{(k)}$, i denote number of packet's GOP, j express number of packet's frame and at last k is number of packet in its frame.

$D'_{sm}(P_{i,j}^{(k)}, QN)$ is calculated similar to D' in [1] and only its difference is using smoothed playout delay instead of statistical playout delay. Its formula is as follows:

$$D'_{sm}\left(P_{i,j}^{(k)}, QN\right) = \Delta_{sm} + \left((i-1)\alpha + (j-1)\right)\lambda + R\left(P_{i,j}^{(k)}\right) \qquad (1)$$

- Δ_{sm}: smoothed playout delay
- *α:* GoP size
- λ: inter-frame interval time
- *R:* retransmission extension period (added time according to packets importance)

Also:

$$R\left(P_{i,j}^{(k)}\right) = \lambda\left(M\left(F_{i,j}\right) + 1\right) \qquad (2)$$

- $M(F_{i,j})$: number of frames inter-coded with respect to *Fi,j*

Effective items in smoothed playout delay for a given packet are:

1. Maximum playout delay (real delay in receiver)
2. Amount of smoothed playout delay for prior packet or packets (used amount of playout delay hitherto)
3. required time for amending lost playout delay time (regarding to transmission rate)
4. Queue which packet is therein
5. Packet importance (dependency of other packet to given packet)

For offering effective formula, investigation impact of each factor over estimated playout delay for a given packet is necessary:

1. Maximum playout delay is a fixed parameter which is specified by user or receiver. Surely whatever it be more, estimated (smoothed) palyout delay can be larger. Indeed smoothed playout delay (namely Δ_{sm}) is a number between 0 and Maximum playout delay. We proposed Δ_{max} notation for this parameter.
2. Second parameter was mentioned is playout delay amount for prior packet or packets which for prevention from any ambiguity its better this parameter be calculated as summation of lost playout delay times yet. It is obviously distinctive whatever lost time by prior packets be more, restriction will be harder so Δ_{sm} must be lesser. But using this parameter thus cause bad effect due to primary packets waste much time and always time for next packets will be low and as a result probability of packet lost will be greater gradually. Surely there are some important packets too which won't have much time for retransmission. So it's better uppermost impact of this factor is considered partly random which is more effective because lost playout delay won't be cumulative. Proposed notation for this factor is Δ_{lost}.
3. One of important issues for computing smoothed playout delay, is calculating amendment time (with proposed natation T_{amend}) meaning if whole time be used for a packet namely how much time required this lost time be amended while each packet be transmitted only once. For example if playout delay be 500ms, for amending this 500ms how much time required or how many packet must be transmitted (only once) till this time be recreated for next packets. Amendment time is calculated according to queue parameters specially transmission rate. Whatever Amendment time is greater Δ_{sm} must be smaller.
4. Queue coefficient (QW') which consider more importance for lower priority queues! It's because making relative fairness between different queues. Because lower priority queues essential have low transmission rate it's required assign higher priority to them here for several reasons. One of reasons is low bit rate in these queues cause they need to more time for retransmission and with granting this priority, they obtain chance to retransmission. Second issue that Δ_{max} parameter is shared between different queues therefore only one Δ_{lost} must be considered for all queues but calculating of T_{amend} is independent and for each queue is performed regarding to its parameters. Because transmission rate from higher priority queues is more, more time is assigned to them from playout delay in ordinary state which isn't desirable and its effect should be neutral due to before mentioned reason. Other matter which cause this situation be worse is existence of more important packets in higher priority queues. So for fairness retainment and prevention from unfavorable results, it's necessary to setting queue coefficient in favor of lower priority queues.
5. Packet importance ($M(Fi,j)$) which whatever be larger, Δ_{sm} must be greater.

According to subjects which mentioned yet, proposed formula expressed as:

$$\Delta_{sm}\ (AC[QN], P_{i,j}^{(k)}) = \Delta_{max} \times x \quad (0 <= x < 1) \qquad (3)$$

$$\Delta_{sm}\ (AC[QN], P_{i,j}^{(k)}) = \Delta_{max} \times h\left(\frac{(M(F_{i,j})+1) \times QW'[QN]}{T_{amend}[QN] + (\Delta_{lost} \times g)}\right) \qquad (4)$$

- h: a function for mapping fraction to a number between 0 and 1
- g: a function to generate random number between 1 and 2 which reason expressed afterwards

Now it is necessary to survey how to calculate functions and variables. $M(F_{i,j})$ is a distinct parameter which it's value is gotten according to packets dependencies and vary from 0 to (a-1). Because this value don't be 0, it sum with 1. $QW'[QN]$ is adjustable for each queue and of course should be selected accurately due to using smoothed playout delay have proper impact. Here used from multiplication of two parameter but it's possible using of summation too. Abaut which of them is better, we can argue but if each of them is used corresponding and proper values must be select and much accuracy is necessary for suitable effect.

$T_{amend}[QN]$ for each time deciding to transmission must be calculated. For calculation it there is need to transmission rate estimation for each queue which is out of this paper's space but generally for consideration both short time (variant) and long time network condition in transmission rate, using from (5) formula is possible to estimation it. (for more simplicity and intelligibility, [QN] notation is deleted from some formulas for example (5) formula)

$$Effective\ Rate\ (R_E) = Current\ Rate \times\ x + Mean\ Rate \times\ (1-x) \text{ (bps)} \quad (5)$$

- x: a parameter with arbitrary value for setting current rate importance versus mean rate.

Estimation of current rate for a short interval time is possible to perform.

Having transmission rate now estimation of amendment time is possible: firstly will be calculated how size of video need to be transmitted (6 formula).

$$Mean\ Video\ Size\ (Size_M) = \frac{Video\ Size + Packets\ Count \times\ LEN_{hdr}}{Video\ Time} \text{ (bit)} \quad (6)$$

Now must be knowed for transmission this mean size of data how much time is required regarding to transmission rate (formula 7).

$$Required\ Time\ (T_R) = \frac{Size_M}{R_E} \text{ (s)} \quad (7)$$

This amount must be less than 1. Extra time which we have in one second is (8 formula):

$$Extra\ Time\ (T_E) = 1 - \ T_R \text{ (s)} \quad (8)$$

This value is certainly less than 1 too. Finally amendment time be calculated by 9 formula:

$$T_{amend}[QN] = \frac{\Delta_{max}}{T_E[QN]} \text{ (s)} \quad (9)$$

Because $T_E[QN]$ is less than 1, $T_{amend}[QN]$ will be greater than Δ_{max} but it is possible Δ_{max} be considered as low limit of $T_{amend}[QN]$. Too for one high limit because transmission rate (assuming once transmission of each packet and irrespective to which it is successful or no) usually is greater than video rate (with unit of frame per second), likely $T_{amend}[QN]$ time isn't greater than twice Δ_{max}. This supposition is for which if transmission rate don't be greater twice video rate probability of retransmission will be low and near to zero. But this is an assumption and is considered for high limit determination of $T_{amend}[QN]$. In practice it is possible to be achieved greater values which must be normalized. According to expressed subject value of this parameter will be limited between Δ_{max} and $2 \times \Delta_{max}$.

Other parameter in proposed formula (4) is summation of lost time from Δ_{max} namely Δ_{lost}. Maybe at first glance its calculation be looked very hard but with a little precision we can take to account a simple way for calculation it. With having start time of video stream and calculation of its difference from current time, it is distinct packets belonging to what time of video must be transmitted. If sending packet have time less than or equal with two values difference, it be used any amount of playout delay but if it's time be greater regarding to (1) formula, $R(P_{i,j}^{(k)})$ must be minus from their difference and sum with λ. If result be greater than 0 this value considered as Δ_{lost} and else be valued with 0 (it's better to say least value after 0). If Δ_{lost} be greater than or equal with Δ_{max} under some condition, Δ_{sm} will be 0 due to it be dropped with high probability. Here according to the expression before there is a point which with considering short value for this parameter, first packets have the chance of wasting much amount of playout delay which is undesirable matter. For prevention from this problem that's needed its value firstly be greater than 0 and secondly be random somewhat. For this reason it multiplied by g function which product random numbers between 1 and 2. So because Δ_{lost} must be less than Δ_{max} and on the other hand T_{amend} is between Δ_{max} $2\times\Delta_{max}$, maximum value of Δ_{lost} will be like to T_{amend}. Fraction which is seen in (4) must be converted into a number between 0 and 1. For this purpose that's needed to be specified minimum and maximum values of this fraction. Whole of the parameters of this fraction have distinct minimum and maximum values thus determination of minimum and maximum is possible for it. *h* function is for mapping numbers of this range to a number between 0 and 1. Assuming amount of the fraction be *x*, h function is:

$$h(x): \frac{x - min(x)}{max(x) - min(x)} \qquad (10)$$

When Δ_{sm} became determined there is still a problem. Problem is due to value achieved from (4) although is less than Δ_{max} but there is the probability of (however low) its summation with Δ_{lost} exceed from Δ_{max} which it is an unacceptable issue. So before using Δ_{sm}, this matter must be investigated. Regarding to it smoothed playout delay calculation algorithm is:

- **smoothed playout delay calculation algorithm:**

If ($\Delta_{lost} >= \Delta_{max}$)

$\Delta_{sm} = 0$

else

Δ_{sm} (*AC[QN]*, $P_{i,j}^{(k)}$) $= \Delta_{max} \times h(\frac{(M(F_{i,j})+1) \times QW'[QN]}{T_{amend}[QN] + (\Delta_{lost} \cdot g)})$

If (Δ_{sm} (*AC[QN]*, $P_{i,j}^{(k)}$) $+ \Delta_{lost} > \Delta_{max}$)

Δ_{sm} (*AC[QN]*, $P_{i,j}^{(k)}$) $= \Delta_{max} - \Delta_{lost}$

As be seen still it's possible to initializing Δ_{sm} with Δ_{max} (in aspect of algorithm) and also which summation Δ_{sm} with Δ_{lost} be equal to Δ_{max}. Because never don't arrive to Δ_{max}, using some Solution is possible. For example instead of using Δ_{max}, a less threshold limit be used which isn't a good idea because it is a static solution and spite of creating a safe margin, sometimes will be wasted. Also it doesn't help to lack of severe limitation for next packets. Better idea is using a random number between two values and with condition of it be less than Δ_{max} and such this algorithm will be improved. Also it's better to be added an equal sign to second condition in improved algorithm till in addition to satisfying summation of two values condition, prevent from initializing Δ_{sm} with Δ_{max} since exception occurrence.

- **Improved smoothed playout delay calculation algorithm:**

Other issue is impact amount of each parameter in ultimate value for each video packet and this formula can be more optimum using coefficients for different parameters. This subject is required extensive agument and can be considered in future work.

Simulation and evaluating of proposed mechanism

For simulating proposed mechanism have been used from NS-2[9,10] and Evalvid[1] tool which is employed for transmission and evaluating video over NS. Evalvid drawback in context of video transmission is lack of video buffering and buffering management in the receiver side. Regarding need to considering playout delay and packets deadline, evaluating without using it was impossible. So we add buffering ability and buffering management to Evalvid till researchers can do simulations and evaluation of videos related issues effectively in future works.

Before simulation scenario expression, declaration of some assumptions is required.

- **Simulation assumptions:**

[1] Evaluation video

1. Two nodes in the network are defined: 1- video sender or server, 2- video receiver or client.
2. Wireless part of network is considered.
3. Both nodes are fixed and motionless.
4. Video coding is MPEG-4.
5. Channel error follows from uniform distribution.

In table 1 some important simulation parameters and in table 2 precise numbers of video frames is observed.

Table 1- simulation parameters values

Parameter	Value (Values)
Transmitting video	foreman-qcif
Video rate	30fps
GoP size	9
Video packets size	Up to 1024
Receiver playout delay	500ms
Retry limit	3 in two higher priority queues/1 in two lower priority queues
Transmission rate	Base rate 1Mbps/Maximum rate 11Mbps
Sender queues capacity	Each queue 50 packets
Queue size control algorithm	Drop-tail
Receiver buffer capacity	500 packets
$QWp1[1]$, $QWp1[2]$, $QWp1[3]^*$	0.02, 0.11, 0.21
$QWp2[1]$, $QWp2[2]$, $QWp2[3]^*$	0.4, 1.5, 5
$fraction\_sum^*$	1
x^*	0.7
ep^*	5ms

* Last five lines are related to proposed mechanism parameters. First and second lines show weight of different queues in multiplication and summation states respectively and third line represent multiplication or summation selection in calculation of playout delay calculation. In line 4, *x* parameter is a value which is used in transmission rate calculation for weighting mean value versus current value. Also ***ep*** parameter in last line shows minimum value of lost playout delay (Δ_{lost}).

Table 2- precise number of video frames in simulation

Total	B	P	I	Frame type / Parameter
400	266	89	45	Frame number
659	273	149	237	Packet number

In simulation, we study proposed mechanism SPD-RA in three states: 1- only video traffic 2- light traffic 3- heavy traffic. In these states changing two important parameters number of video packets received via receiver and also Average PSNR (APSNR) versus network error increment is evaluated. APSNR is a useful parameter for evaluation of video quality and get more precise quality estimation rather than PSNR. For briefness observance, changing trend for number of received video packets only is showed in first state (figure 2). Charts are acquired from 15 times run of simulation.

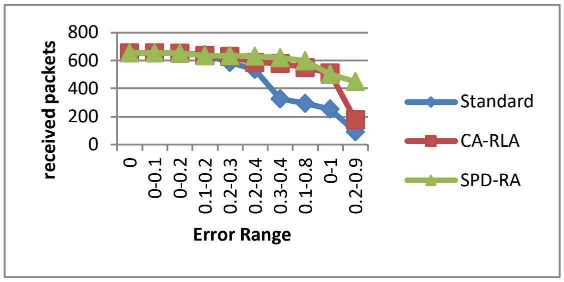

Fig.2- number of received video packets for different retransmission methods while streaming only two video

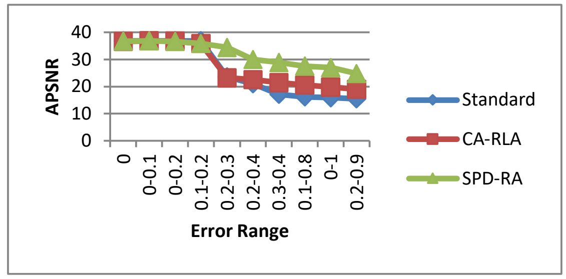

Fig.3- APSNR at receiver in light traffic state

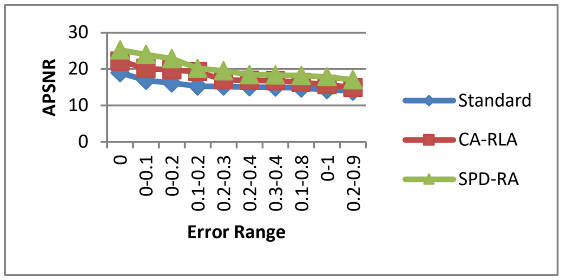

Fig.4- APSNR at receiver in heavy traffic state

In [4] is said CA-RLA has better performance rather than other mechanisms. So here we try to implement it correctly and use it for comparison with proposed mechanism SPD-RA. Of course a part of CA-RLA is image processing which we ignored it for simplicity because our mechanism can be improved by image processing too.

As be observed in many times proposed mechanism RA-SPD is better than other mechanisms and rarely has lower performance than CA-RLA or EDCA standard with little difference. This is somewhat natural because in changing and uncertain condition of wireless network, any mechanism can have best operation but maybe with better setting even can be achieved better results by RA-SPD. This issue is required more studies but in this paper has been tried simulation settings be proper as possible.

For numerical investigation of video quality improvement must be understood each 1db increase in APSNR about is equal with 3 percent quality improvement. So according to charts video quality improvement by using RA-SPD which is observed at receiver even can reach up to 30 percent (for example in light traffic) but if we measure with more precision, improvement rather than CA-RLA is about 5 percent and minimum 10 percent rather than standard EDCA on average.

5. Conclusion and future works

This paper offer an effective approach for retransmission decision of video packets in IEEE 802.11e standard which use from a concept namely Smoothed Playout Delay (SPD). Retransmission of packets in video application over network must perform according to playout delay amount. SPD is a calculated playout delay from maximum playout delay at receiver which has been applied it some effective parameters. Before playout delay was obtained from statistical approaches but now can be obtain via smoothed playout delay which it is easier and lighter way for this mean and can be very effective. Also this mechanism has high consistency with channel condition changes and different traffic existence. Video quality in this mechanism has been raised up to 5 percent than prior approaches and minimum 10 percent than EDCA standard.

One of future works is performing better settings for RA-SPD as possible and also studying effect of changing different parameters on RA-SPD performance. Using this approach with different queuing methods which do queuing of video frames with awareness and dynamically also can be a useful issue for transmitted video quality improvement. Also using RA-SPD mechanism in condition which is different from simulation in this paper can complete researches in this context. Applying image processing ideas is one of other issues which aid to improvement of this approach so can help to identify more important packets with more accuracy and retransmit further the most important packets.

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