Using Smoothed Playout Delay for Improving Video Quality over EDCA

Mohammad Mahdi Motahhari Kia, Marsa Rayani, Seyyed Hossein Raja, Mohammad Iman Ghiasi

1Department of Computer, Damavand Branch, Islamic Azad University, Damavand, Iran
2Faculty of Computer Science and Information Technology, University Putra Malaysia, 43400 UPM, Serdang, Selangor, Malaysia
3M.Sc. in Information technology of DSP Research Institute, Tehran, Iran
4Research Institute of Petroleum Industry (RIPI), Tehran, Iran
1damavandiau@mail.com, 2marsarayani1@gmail.com, 3hosseinraja@dspr.com, 4ghiasimi@ripi.ir

Abstract

In this paper a method for video packets retransmission management on 802.11e WLANs and Enhanced Distributed Channel Access (EDCA) mechanism namely Smoothed Playout Delay-base Retry Adaptation (SPD-RA) has been offered. It estimate deadline of video packets based on maximum playout delay in receiver and with regards to some effective parameters in deadline. Since this approach requires the knowledge of Data-link layer from Application layer information, regarded as a Cross-layer approach. The methods is presented till now, didn’t consist of these concepts and formulas which we notified. It is also demonstrated that the method which we presented improves the simplification and calculation speed. Results of simulations show that correct setting of SPD-RA parameters without statistical information collection make improvement of received video quality at receiver in bad network condition is possible up to 5 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 threatened severely due to contention between users, different data streams and existence of noise in channel. These factors and similar factors require thinking about resources 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 packet 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 SPD-RA algorithm will be described in it. Then smoothed playout delay formula will be stated which SPD-RA algorithm is based on it, that is to say this formula is used for packets retransmission and parameters of this formula will be described 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 the 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 states 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 use 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 described 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.

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 $\beta$ and has some uses. Real playout delay can’t be used directly 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 (SPD-RA).

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 SPD-RA mechanism.

Objectives of SPD-RA 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 SPD-RA mechanism is:

After an unsuccessful packet $P$ transmit do:

1. $T_{trans}(r, QN) = T_{cur} + T_{back}(r, QN)$
   - In the $(r-1)$'th try:
     - If $T_{trans}(r, QN) + \delta > D_{sm}(P_{ij}^{(k)}, QN)$
       - Drop $P_{ij}^{(k)}$
     - In $(r)$'th try:
       - If $T_{cur} > D_{sm}(P_{ij}^{(k)}, QN)$
         - Drop $P_{ij}^{(k)}$
       - Else
         - Transmit $P_{ij}^{(k)}$

2. 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
   - $\delta$: maximum propagation delay in wireless channel
   - $D_{sm}(P_{ij}^{(k)}, QN)$: packet $P_{ij}^{(k)}$ deadline according to queue $Q$ parameters

802.11c MAC Layer


Virtual collision handler

Wireless Network  Physical Layer

From receiver

Packet Drop

Ack

Fig.1. General scheme of SPD-RA mechanism

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

\[ D_{sm}'(P_{ij}^{(k)},QN) \] 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:

\[
D_{sm}'(P_{ij}^{(k)},QN) = \Delta_{sm} + \left( \left( \frac{i-1}{j-1} \right) \alpha + \lambda \right) + R(F_{ij})
\]  

(1)

- \( \Delta_{sm} \): smoothed playout delay 
- \( \alpha \): GoP size 
- \( \lambda \): inter-frame interval time 
- \( R \): retransmission extension period (added time according to packets importance)

Also:

\[
R(F_{ij}) = \lambda(M(F_{ij}) + 1)
\]

(2)

- \( M(F_{ij}) \): number of frames inter-coded with respect to \( F_{ij} \)

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. Time required for retransmission time of 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) playout delay can be larger. Indeed, smoothed playout delay (namely \( \Delta_{sm} \)) is a number between 0 and Maximum playout delay. We proposed \( \Delta_{max} \) 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 \( \Delta_{sm} \) 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 \( \Delta_{lost} \).

3. One of important issues for computing smoothed playout delay, is calculating amendment time (with proposed notation \( T_{amend} \)) 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 \( \Delta_{sm} \) 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 \( \Delta_{max} \) parameter is shared between different queues therefore only one \( \Delta_{lost} \) 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 retention and prevention from unfavorable results, it’s necessary to setting queue coefficient in favor of lower priority queues.

5. Packet importance \( (M(F_{ij})) \) which whatever be larger, \( \Delta_{sm} \) must be greater.

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

\[
\Delta_{sm}(AC[QN],P_{ij}^{(k)}) = \Delta_{max} \times x \quad (0 \leq x < 1)
\]

(3)

\[
\Delta_{sm}(AC[QN],P_{ij}^{(k)}) = \Delta_{max} \times h\left( \frac{M(F_{ij}) + 1 \times QW'[QN]}{T_{amend}[QN] + (\Delta_{lost} \times \delta)} \right)
\]

(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. \( M(F_{ij}) \) is a distinct parameter which it’s value is gotten according to packets dependencies and vary from 0 to (\( \alpha - 1 \)). 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. About which of them is better, we can argue, but if each of them is used corresponding and possible using of summation too. About which of them is better, we can argue but if each of them is used corresponding and possible using of summation too. About which of them is better, we can argue but if each of them is used corresponding and possible using of summation too. About which of them is better, we can argue but if each of them is used corresponding and possible using of summation too. About which of them is better, we can argue but if each of them is used corresponding and possible using of summation too.

\[ T_{amend}[QN] \] 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)

\[
\text{Effective Rate } (R_E) = \text{Current Rate} \times x + \text{Mean Rate} \times (1-x)(\text{bps}) \tag{5}
\]

\( x \): 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).

\[
\text{Mean Video Size } (Size_M) = \frac{\text{Video Size} + \text{Packets Count} \times \text{LEN}_{dat}}{\text{Video Time}} \text{(bit)} \tag{6}
\]

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

\[
\text{Required Time } (T_R) = \frac{Size_M}{R_E}(s) \tag{7}
\]

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

\[
\text{Extra Time } (T_E) = 1 - T_R (s) \tag{8}
\]

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

\[
T_{amend}[QN] = \frac{\Delta_{max}}{T_{E}[QN]}(s) \tag{9}
\]

Because \( T_E[QN] \) is less than 1, \( T_{amend}[QN] \) will be greater than \( \Delta_{max} \) but it is possible \( \Delta_{max} \) 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 \( \Delta_{max} \). 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 \( \Delta_{max} \) and \( 2 \times \Delta_{max} \).

Other parameter in proposed formula (4) is summation of lost time from \( \Delta_{max} \) namely \( \Delta_{lost} \). 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, \( R(P_l^{(k)}) \) must be minus from their difference and sum with \( \lambda \). If result be greater than 0 this value considered as \( \Delta_{lost} \) and else be valued with 0 (it’s better to say least value after 0). If \( \Delta_{lost} \) be greater than or equal with \( \Delta_{max} \) under some condition, \( \Delta_{sn} \) 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 produce random numbers between 1 and 2. So because \( \Delta_{lost} \) must be less than \( \Delta_{max} \) and on the other hand \( T_{amend} \) is between \( \Delta_{max} \) and \( 2 \times \Delta_{max} \), maximum value of \( \Delta_{lost} \) 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:
If \( \Delta_{sm} \) became determined there is still a problem. Problem is due to value achieved from (4) although is less than \( \Delta_{max} \) but there is the probability of (however low) its summation with \( \Delta_{lost} \) exceed from \( \Delta_{max} \) which it is an unacceptable issue. So before using \( \Delta_{sm} \), this matter must be investigated. Regarding to it smoothed playout delay calculation algorithm is:

- **smoothed playout delay calculation algorithm:**
  
  If \( (\Delta_{lost} \geq \Delta_{max}) \)
  \( \Delta_{sm} = 0 \)
  
  else
  \[
  \Delta_{sm} (AC[QN], P_{i,j}^{(k)}) = \Delta_{max} \times h(M[Fi,j]+1) \times QW'[QN] + (\Delta_{lost} \cdot \theta) 
  \]

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

  As be seen still it’s possible to initializing \( \Delta_{sm} \) with \( \Delta_{max} \) (in aspect of algorithm) and also which summation \( \Delta_{sm} \) with \( \Delta_{lost} \) be equal to \( \Delta_{max} \). Because never don’t arrive to \( \Delta_{max} \), using some Solution is possible. For example instead of using \( \Delta_{max} \), 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 \( \Delta_{max} \) 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 \( \Delta_{sm} \) with \( \Delta_{max} \) since exception occurrence.

- **Improved smoothed playout delay calculation**

  if \( (\Delta_{lost} \geq \Delta_{max}) \)
  \( \Delta_{sm} = 0 \)
  
  else
  \[
  \Delta_{sm} (AC[QN], P_{i,j}^{(k)}) = \Delta_{max} \times h(M[Fi,j]+1) \times QW'[QN] + (\Delta_{lost} \cdot \theta) 
  \]

  if \( \Delta_{sm} (AC[QN], P_{i,j}^{(k)}) + \Delta_{lost} \geq \Delta_{max} \)
  \[
  \Delta_{sm} (AC[QN], P_{i,j}^{(k)}) = \Delta_{max} - \Delta_{lost} - \text{rand} (1, \Delta_{max} - \Delta_{lost}) 
  \]

  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 augument and can be considered in future work.

4. 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. 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

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value (Values)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transmitting video</td>
<td>foreman-qcif</td>
</tr>
<tr>
<td>Video rate</td>
<td>30fps</td>
</tr>
<tr>
<td>GoP size</td>
<td>9</td>
</tr>
<tr>
<td>Video packets size</td>
<td>Up to 1024</td>
</tr>
<tr>
<td>Receiver playout delay</td>
<td>500ms</td>
</tr>
<tr>
<td>Retry limit</td>
<td>3 in two higher priority queues/1 in two lower priority queues</td>
</tr>
<tr>
<td>Transmission rate</td>
<td>Base rate 1Mbps/Maximum rate 11Mbps</td>
</tr>
<tr>
<td>Sender queues capacity</td>
<td>Each queue 50 packets</td>
</tr>
<tr>
<td>Queue size control algorithm</td>
<td>Drop-tail</td>
</tr>
<tr>
<td>Receiver buffer capacity</td>
<td>500packets</td>
</tr>
<tr>
<td>QWP1[1], QWP2[2], QWP1[3]*</td>
<td>0.02, 0.11, 0.21</td>
</tr>
<tr>
<td>QWP2[1], QWP2[2], QWP2[3]*</td>
<td>0.4, 1.5, 5</td>
</tr>
<tr>
<td>fraction_sum*</td>
<td>1</td>
</tr>
<tr>
<td>x*</td>
<td>0.7</td>
</tr>
<tr>
<td>ep*</td>
<td>5ms</td>
</tr>
</tbody>
</table>

\(^1\)Evaluation video
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. \( x \) parameter in line 4 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 (\( \Delta_{\text{lost}} \)).

<table>
<thead>
<tr>
<th>Frame type</th>
<th>I</th>
<th>P</th>
<th>B</th>
<th>Total</th>
</tr>
</thead>
<tbody>
<tr>
<td>Frame number</td>
<td>45</td>
<td>89</td>
<td>266</td>
<td>400</td>
</tr>
<tr>
<td>Packet number</td>
<td>237</td>
<td>149</td>
<td>273</td>
<td>659</td>
</tr>
</tbody>
</table>

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

Fig.3- APSNR at receiver while streaming only two video

Fig.4- APSNR at receiver in light traffic state

Fig.5- APSNR at receiver in normal traffic state

Fig.6- APSNR at receiver in heavy traffic state

Table 2- precise number of video frames in simulation
In simulation, we study proposed mechanism SPD-RA in four states: 1- only video traffic 2- light traffic 3- normal traffic 4- 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.

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 SPD-RA 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 SPD-RA. 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 SPD-RA which is observed at receiver even can reach up to 30 percent (for example in light traffic) but if survey with more precision, improvement rather than CA-RLA is about 5 percent and minimum 11 percent rather than standard EDCA often.

5. Conclusion and future works

In this document is proposed a method for retransmission of video packets over EDCA mechanism which use from a concept namely smoothed playout delay. Playout delay is a parameter which is used in deadline calculations for a packet while deciding for retransmission. Smoothed playout delay is obtained from maximum playout delay used at receiver and applying some effective parameters. So playout delay which before was obtained via statistical approaches now has been replaced by smoothed playout delay. Thus deadline calculation is performed easily in proposed mechanism SPD-RA. This mechanism has high consistency with channel condition changes and different traffic existence. Improvement of video quality in this mechanism arises up to 4 percent than prior approaches and minimum 10 percent than EDCA standard.

In future work can perform better settings for SPD-RA as possible and also study effect of changing different parameter on SPD-RA 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. Too using SPD-RA mechanism in condition which is different from simulation in this paper can complete researches in this context. One of other issues which aid to improvement of this approach is applying image processing ideas so can identify more important packets with more accuracy and retransmit those further according to their importance.

6. References

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7. Biography

Mohammad Mahdi Motahhari Kia was born 1984 in Tehran, Iran. He is Master of IT Engineering in Computer Networks ground studied at Islamic Azad University of Qazvin branch. Her master thesis is about video transmission in IEEE 802.11e wireless LANs and so he has many studies in this context. Also he has many experience and history in teaching at different universities like Islamic Azad University of Firoozkooh branch and Islamic Azad University of Damavand branch.

Marsa Rayani Faculty of Computer Science and Information Technology, University Putra Malaysia, 43400 UPM, Serdang, Selangor, Malaysia.

Seyyed Hossein Raja is Master of IT Engineering in DSP research institute Tehran, Iran. He is experienced about 14 years in different fields of IT, including teaching, programming, networking, network security, database, virtualization, Linux, Windows, and Mail server. He has scientific degree in CCIE Routing and Switching, CCNP, CCSP, CCIP, MCSE, RHCSS and LPI 3(Mail, Security, Open Ldap, Mixed Environment). He has been published dozens of articles in journals and international conferences. He published a book entitled E-Mail Security in Persian language with ISBN 978-600-6529-00-4 in 2012 with Pendare Pars publisher.

Mohammad Iman Ghiasi Was born 1982 in Iran. He received the M.Sc. degree in electrical engineering from K.N.Toosi University of Technology, Tehran, Iran in 2006. He is currently senior energy researcher in Research Institute of Petroleum Industry (RIPI). His areas of interests include power system transient, power quality and energy consumption.