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Adaptive end-to-end QoS for multimedia over heterogeneous wireless networks

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ABSTRACT

There has been a surge of interest for multimedia applications over wireless networks in recent years. A colossal number of ways have been proposed to decrease delay, delay jitter and loss in wireless networks and good user-perceived quality in video over internet. This paper studies the multimedia over heterogeneous wireless networks, requirements and problems, and proposes a new scheme to overcome the obstacles. The proposed scheme, takes into account the effects of Application-Level Wireless Multilevel ECN marking (AWMECN), thus helps us overcome the congestion/loss mistake problems. For handoff, handover and lossy link problems, it is considered that a freezing mechanism is in use in application layer and assumed that the upper layers can be aware of disconnection periods to make the rate adaptation decisions. Also a new scheme has been added to receiver to gracefully degrade the quality when no other action is available to combat the long delays without data which is caused by handoffs and wireless temporary disconnections. The transport layer mechanism is chosen to be UDP for avoiding TCP regarding problems. We believe that obtaining a good quality of video depends on good performance of all layers and tried to use the best mechanism in each layer.

1. Introduction

Internet is growing rapidly and its growth has been successful in the wired networks over the past years but the process for wireless and mobile networks is not as successful as for traditional networks. The story is going even worse when considering the heterogeneous wireless environment. Today's internet challenge is multimedia and QoS support, especially over wireless networks. A number of methods have been proposed to combat the problem over Internet's best-effort nature. New research trend is going to attack the problem in wireless and mobile networks.

There are three main impediments in QoS support, especially video streaming over the internet. First, the variable network performance due to load changes. More specifically available bandwidth is highly variable, which can be dealt with, to some extent, by using a buffer; but this option is limited by the buffer space available at the receiver. Second obstacle is the network congestion, which results in packet loss in network interior nodes and low arrival rate at the receiver. Congestion induced frame loss may severely affect video quality. Finally, the bandwidth availability in the network is highly unpredictable which makes quality adaptation difficult.

When one is to support QoS and more specifically video streaming over wireless networks, there is a need to go further to consider the problems of wireless and mobile networks as well. For an end-to-end QoS support, there always has been a choice of TCP or UDP. UDP is connection-less and does not maintain packet numbering and timing. It also does not provide any kind of information about the network state. There have been a number of ways proposing UDP in companion with RTP

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for time stamping and sequence number and using RTCP for obtaining network information to support QoS over wireless networks which is the preferred method for multimedia transmission over wireless networks [10]. The main advantage of this way is that UDP does not maintain packet loss and retransmission and it does not include any kind of congestion control mechanisms which take a lot of time, especially with loss/congestion mistake scenarios in wireless networks. But as the compression volume increase, the need to retain lost packets increase. On the other hand TCP works effectively over traditional networks but it does not work effectively for multimedia services as its performance degrades due to window size decrease in the response to packet loss resulting from the congestion.

TCP shows even worse performance over the wireless networks due to five main problems. First, limited bandwidth, which is a bottleneck to improve the TCP performance over wireless networks. Second, long round-trip times which cause the growth rate of the TCP congestion window to be very low resulting in low sending rates. Third, random losses due to low signal strength in an area or duration of noise; since TCP reacts to packet loss with halving its congestion window, bursts of errors may cause the TCP congestion window to reduce TCP sending rate dramatically. Fourth, User Mobility and handoffs; when mobile user transfers from a cell to another a handoff occurs due to user's need to change base station (BTS). Handoffs usually take about 300 ms and all packets get lost in this duration. Fifth, short flows services; this service is offered in a very short time interval. In the beginning of the connection TCP is still in slow-start and the data transmission may stop before the window size has enough time to increase sending rate.

There are number of ways to solve these problems including pure end-to-end protocols, link layer protocols, split connection protocols, soft-state transport-layer caching protocols, cross layer signaling protocols [8,10] but none of the proposed methods could solve all the TCP regarding problems in wireless networks.

In this paper we propose a way to overcome the video streaming problems in mobile wireless networks and provide an algorithm that shows better QoS and multimedia delivery over wireless networks. The proposed mechanism aims at adaptive end-to-end QoS support so it is classified in pure end-to-end protocols category.

The proposed mechanism tries to attack the QoS support problems with the main adaptation scheme used in application layer and a mixture of best services for each underlying layer. The main adaptation scheme which consists of rate and quality adaptation mechanisms and the graceful degrade mechanism are all done in application level. The Wireless MECN marking is used in network layer, but the decision about the MECN values is also made at application level. For the transport layer, it is assumed that the UDP is used and sequence numbering and time stamping is also done in application level header.

The rest of this paper is organized as follows: in Section 2 we study the problems encountered in QoS support over wireless networks and the ones we are going to solve in this paper. The third section describes in detail the proposed scheme. Simulation results are summarized in section four. Finally, section five includes conclusion remarks and a way to future work.

2. QoS support obstacles in wireless networks

We assumed the main problems of QoS support over wireless networks are random losses and handoffs as well as congestion. There are several proposed methods to overcome these problems [7,9]. The main problems and shortcomings of proposed features available till now are discussed here in detail. Then we study the multimedia characteristics and major obstacles in multimedia over heterogeneous wireless networks and show that how wireless disconnection durations affect the multimedia and its user-perceived quality over the mobile wireless networks.

2.1. Congestion

In the traditional TCP, packet loss is the main indicator of congestion events. In such networks, where the link error is very low and links are considered to be reliable, this mechanism would show acceptable results but when used in wireless networks, it may cause starting the congestion control mechanisms in the case of non-congestion related losses. In other words, random packet loss is one of the important obstacles in wireless networks that impose big bursts of packet loss in wireless networks. Therefore offering a good end-to-end QoS supporting scheme, requires considering an environment in which packets experience congestion related losses and also random losses due to wireless link. One of the problems to attack in this stage is to detect random losses from those originated from congestion events. One of the famous ways to solve this problem is explicit congestion notification. ECN [6,8,3] marking allows one to explicitly notify the receiver and therefore in ACK packets the sender, about the congestion and how to act accordingly. But for supporting QoS over the wireless networks it is not enough to have one bit indicator of congestion. A multi-level ECN mechanism as proposed in [2] best fits the requirements of multimedia applications that are the main group of applications needing QoS support. We used previously proposed method, Application Layer Wireless Multi-level ECN for obtaining a feedback from network backlog [3]. This multi-level ECN-like mechanism is done in the internal hops in the network and its worse value along the way is remained unchanged.

2.2. Handoff and temporary disconnections

The second problem in support of QoS in wireless networks is the handoff and temporary disconnection periods caused by user mobility. As mentioned earlier, in the mobile networks there is a disconnection gap while a mobile host moves from one

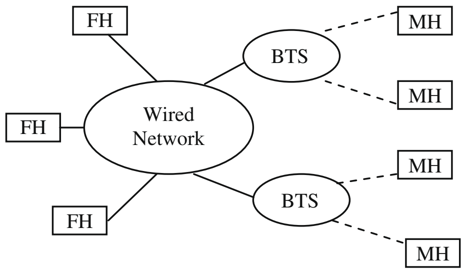

Fig. 1. Wireless network model.

cell to another. In this disconnection period, all of the packets in transition will get lost. ECN-like mechanisms are going to combat the rate-quality adjustment problem in presence of wireless losses and congestion. But there is a need to handoff and disconnection detecting mechanism in the underlying layers. In the disconnection period the packets or their ACK packets are going to get lost and sender may want to resent them that will take some of the sender's time; or there may be the possibility that sender decrease its congestion window size in transport layer that would affect video quality tremendously. The problem gets worst when considering heterogeneous wireless environment. In such an environment, it is considered that a handoff would occur not only between two adjacent cells of the same service, but also among different service areas which would cause longer handoffs. This means that heterogeneous wireless networks need handoff and disconnection aware mechanisms for QoS support. This problem is been solved in this paper by a freezing mechanism running in application level [1]. We here assume the possibility that the receiver's buffer run out of packets to show when it has no knowledge about disconnection event and therefore there exists a need to add some additional states to mechanism showing handoff possibilities before it happens. Therefore the application layer needs to be aware about the handoff and disconnection events.

2.3. Multimedia over heterogeneous wireless networks

Multimedia and more specifically video is very sensitive to network QoS parameters. User-perceived quality is one of the meters used as an indicator of quality of video transmitted video over the wireless networks. It is measured by availability of frames to playback at the receiver. In the case of long temporary disconnections caused by handoffs and handovers, the playback buffer used to maintain delay and jitter, runs out of packets to show and this will affect the user-perceived quality dramatically. So there is a need for a mechanism doing a graceful degradation in these situations prohibiting the quality to fall down suddenly (Fig. 1).

3. AWEQ: adaptive wireless end-to-end QoS

Our proposed algorithm (AWEQ) tries to solve the problems discussed above with Adaptive Wireless Multi-level ECN [3] which is an application layer usage of network layer data, UDP mechanism in transport layer, disconnection signaling mechanism in link layer, and graceful degrade, rate adaptation and quality adaptation in the application layer. The main contribution of this research in this area is using the AWMECN, graceful degrade and the decision making algorithm proposed in Table 2. Putting these appropriate algorithms in different layers to cooperate to make a good performance together is also considered as an important contribution of this paper. We believe we have made good choice of mechanisms in each later which will ultimately result in improved performance.

3.1. Multilevel ECN-like mechanism

For getting a precise feedback from the network, Application-level Wireless Multi-level ECN marking is used [3]. The used process is much like WMECN discussed in [5,2]. As it is proposed in our previous work in [3], AWMECN uses two bits that are being specified for the use of ECN in the IP header to indicate four different levels of congestion instead of binary feedback provided by ECN. Four levels obtained by two bits are assumed for following feedbacks: "00" for non-ECN capable feedback, "01" for no congestion notification, "10" for mild congestion notification and "11" for severe congestion notification. Table 1 summarizes the values used for Congestion feedback.

Table 1
Different ECN values and how to interpret them.

ECN value	How to interpret
00	Non ECN capable
01	No congestion
10	Mild congestion
11	Severe congestion

Table 2
Decision making based on MECN value, handoff possibility and receiver buffer state.

Ar < Tr	ECN value	Handoff	*Buff*(*t* + *RTT*)	Action	
				Receiver	Sender
–	–	Yes	Overflow	None	Rate = 0
–	–	Yes	Underflow	Graceful degrade	Rate = 0t and Dec quality
–	–	Yes	OK	Graceful degrade	Rate = 0
False	00	No	Overflow	None	Dec rate $r \to r_l$
False	00	No	Ok	None	Inc rate ($r \to r_h$ or $r \to$ previous r)
False	00	No	Underflow	None	Inc rate ($r \to r_h$ or $r \to$ previous r)
False	00	No	OK	None	Inc Quality or
True	00	No	OK	None	None
True	00	No	Underflow	Graceful degrade	None
True	10	No	OK	None	None
True	10	No	Underflow	Graceful degrade	Dec quality
True	11	No	–	Graceful degrade	Dec quality
True	–	No	Overflow	None	Dec rate ($r \to r_l$)

Three different thresholds have been proposed on the red queue, minimum threshold, maximum threshold and middle threshold. When the queue size is lower than minimum threshold no packet is marked. When the queue size is between minimum and middle threshold, packets are marked with "10" with maximum probability $P_{\max}$. The rest of packets remain unmarked. When the queue size is between middle and maximum thresholds, packets are marked with "11" with maximum probability $P_{\max}$ and the rest of packets are marked with "10". After the maximum threshold, all the packets are marked with "11". All these values are set in an application level header of packet to be analyzed by the application as proposed in [3].

MECN is set in the router when the queue size reaches the defined thresholds. This is done using 7 and 8 bits in IP header. Then these values are set into application layer header and sent to the receiver.

Using network backlog feedback in transport layer will only help us in rate adaptation by increasing or decreasing the window size. This will help to avoid congestion, but huge shrinks in window size have bad effects on video data which are very sensitive to rate variations. This leads to one of the important benefits of using the feedback in application level, which is the fact that it will enable us to consider more parameters than congestion in increasing or decreasing the sending rate as well as adapting the rate and quality in a way that have the least degrading effect on video stream. For example a feedback that adds disconnection duration, notified by the receiver to the sender, could be added. This feedback could help the sender to avoid sending packets in the short wireless disconnection periods in which all the packets are subject to loss. Another parameter that could be used in adaptation is the playback buffer occupancy which is beneficial in better understanding of video on demand situation. Adding these parameters would also cause in adding a quality adaptation scheme, which is a valuable character we gain in addition to the rate adaptation. Fig. 2 shows the parameters that could be brought into account in the rate and quality adaptation schemes.

Since the decision making relies on different parameters rather than only a network backlog and loss, this kind of decision making avoids two mistakes scenarios for loss; identification of congestion loss as wireless link loss and identification of wireless link loss as congestion loss. The proposed mechanism offers coexistence with competing traffics. It also offers coexistence possibility with old routers with different definitions of ECN fields in IP header, as proposed by Floyd et al. in RFC 3168 and discussed in [5]. This will help using it with old routers as well as avoiding mistakes scenarios in routers.

3.2. Handoff and freeze mechanism

We assume that the UDP is used for transport layer mechanism. Sequence numbering and time stamping is done in application level. We assume a freezing mechanism run at application level. The first step to this was to use a freeze-TCP like

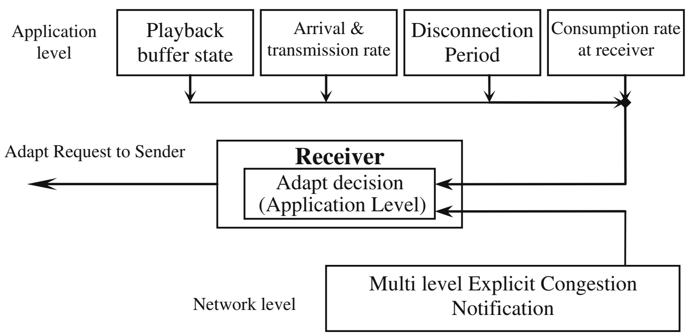

Fig. 2. Adaptation mechanism used.

mechanism, namely ATCP, run at the transport layer as discussed [1] to overcome the handoff and long disconnections of mobile wireless networks. The ATCP improves the performance not only when the TCP sender is a fixed host, but also when the TCP sender is a mobile host. This mechanism involves modifications to the network stack only at the mobile host and requires network layer feedback regarding the status of the connectivity. The algorithm assumes that the network layer sends a connection event signal to TCP when mobile host gets connected to the network and a disconnection event signal when the mobile host gets disconnected from the network. The algorithm uses these signals to appropriately freeze/continue ongoing data transfer and changes the action taken at retransmission timeout event, leading to enhanced TCP throughput. Fig. 3 shows the ATCP mechanism for different scenarios.

Then we modified the algorithm and changed the decision making center to be in application layer. We just assumed that the link layer would signal and aware the adaptation algorithm in application layer from the probable disconnection event and the application layer would adapt its sending rate and quality accordingly. Note that the possibility of future disconnections which could be estimated by signal strength in mobile host is used instead of disconnection event signal. The transport layer mechanism would be a simple UDP. The freeze mechanism in application layer would result in bigger throughput due to the fact that it will never encounter the slow start, retransmission and congestion window problems.

The proposed mechanism would just decrease the sending rate at the sender to zero in the probable handoff and disconnection events. Then it will restart sending the video with the previous rate when the disconnection duration is over. All the decisions are made in application level in the receiver so the sender would need to be aware of what to do. Since the disconnection probability is announced in advance to the occurrence of event, receiver could inform the sender about the disconnection in the last talk.

It is more acceptable that the sending side of the connection to be a server which is in a fixed host but for the future needs we assume that both the sender and the receiver could be mobile and wireless hosts and assumed that in the event of disconnection origination from sender side, sender would also inform the receiver from the handoff and disconnection possibility and would decide by itself to decrease the sending rate to zero. After the disconnection is resolved, the sender would check new state with receiver and act accordingly.

3.3. Rate and bandwidth evaluation

Available bandwidth is the minimum unused bandwidth of any of the links along the path between sender and receiver. If the transmission rate of the flow is lower than the available bandwidth or equal to it, then the arrival rate is equal to transmission rate. Otherwise, the arrival rate will be lower than transmission rate. There is only one value of the transmission rate that the two are equal. For other cases, the link capacity must be know to assess the available bandwidth.

The arrival rate at the destination is a function of transmission rate, the capacity and the available bandwidth of all links along the path [2]. No single bottleneck could determine the arrival rate. For end-to-end measurement of available bandwidth a series of periodic flows is transmitted between the end points. For each flow the receiver analyzes one-way delay variations of the packets to determine whether the transmission rate is higher or lower than the available bandwidth. A

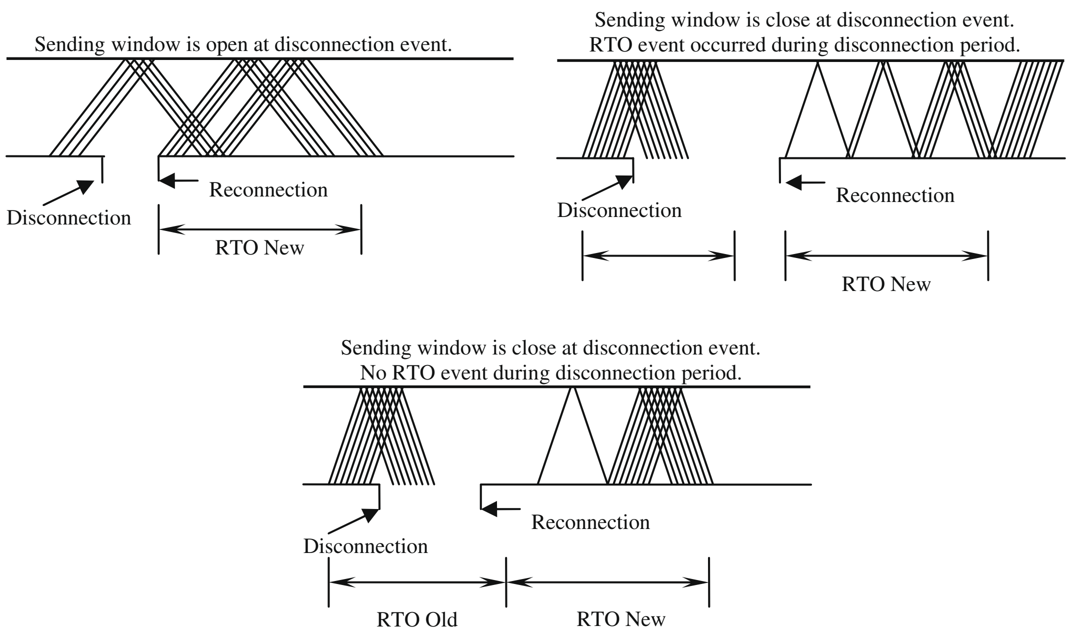

Fig. 3. ATCP freeze mechanism for different Scenarios.

set of video packets can be treated as a packet train and used to obtain an estimate of asymptotic dispersion rate, which are referred to as arrival rate at the client. The measurement is done over a moving window of certain size. Thus, the number of packets may vary from one estimate to another. This estimate is then used to draw a conclusion about the available bandwidth. We assumed the rate of the packets received is calculated using (1) which adds together all the received packets p_i during the Δt_r time interval.

$$r_r = \frac{1}{\Delta t_r} \sum_i p_i \tag{1}$$

The calculation is then modified and enhanced with the $0 \leqslant \alpha \leqslant 1$ factor which is used for taking into account the previous values of received packers in the present calculation.

$$\hat{r}_r = (1 - \alpha) r_r + \alpha r_r \tag{2}$$

The video arrival rate is compared against the transmission rate and conclusion is drawn. The conclusion is in two state domains which show one of followings: available bandwidth is lower than needed by the video stream or is equal or higher then needed by video stream.

3.4. Rate adaptation

After the available bandwidth and transmission and arrival rates have been calculated and ECN is taken to know about the network state and handoff and disconnection events are tested. Similar to formulation done in [2], the playback buffer occupancy for the next Round Trip Time is estimated using the sum of current value of buffer size plus the received packets in next RTT minus the used packets in next RTT:

After the available bandwidth and transmission and arrival rates have been calculated and ECN is taken to know about the network state and handoff and disconnection events are tested. After all, the playback buffer occupancy for the next Round Trip Time is estimated using the sum of current value of buffer size plus the received packets in next RTT minus the used packets in next RTT:

$$buf(t + RTT) = buf(t) + \sum_{i=t+1}^{t+RTT} \hat{r}_r(i) - \sum_{i=t+1}^{t+RTT} u_i \tag{3}$$

After gaining all required information decision is made and quality adaptation and then rate adaptation is made if needed. For a given level of quality, a default schedule using a fixed transmission rate and minimizing the client buffer requirement is prepared ahead of time based on knowledge of video content. The receiver can toggle its need between r_h,a rate higher than the default rate and r_l, a rate less than the default rate without need to change quality level. One can simply toggle or smoothly go the way between these two values.

The time Δt_h in which the sender could increase the sending rate is related to the size of network backlog in time t in which the request is made, and the playback buffer occupancy:

$$\Delta t_h = \max\left(t \leqslant i \leqslant N - 1 : buf(t) + (i - 1) r_h - \sum_{j=t}^{i} u_j \leqslant B\right) - t \tag{4}$$

The lower rate is requested when the playback buffer occupancy is reached a predefined value. In this time the lower rate is used to compensate the higher rate before the playback buffer would overflow. In contrast to the mechanism used in [2], in our mechanism there is no need to use this value after all high rate send periods and using this low rate value is only under the circumstances shown in Table 1.

$$(r_h - r)\Delta t_h = (r - r_l)\Delta t_l \tag{5}$$

3.5. Quality adaptation

There are four different known ways for quality adaptation: adjusting the compression ratio of an on-line encoder, switching among different pre-encoded versions, transcoding a pre-encoded version and dropping a layer of a hierarchical encoding, scheme. In this paper, quality adaptation is assumed to be done in the sender side of the connection by switching between different pre-encoded versions of video requested [2].

Available bandwidth detection is done with the transport layer mechanism, more specifically the wireless dedicated ATCP for getting network information.

3.6. Graceful degradation

In this research, it is assumed to be better to have a graceful degradation in user-perceived quality of multimedia in the case that no action will help to provide needed frames with the pre-encoded quality used. The way used for this graceful degradation is to use a multimedia buffer in a controlled rate. In this way we do the rate and quality adaptation in a way that we could have the enough frames in the playback buffer to show to the user, but when the video size increases and the time interval of the connection becomes long, in the cases that the playback buffer underflow, the problem of running out of frames to show becomes serious. In this case, when the buffer underflow and there seems to be a congestion or a wireless oriented disconnection in the network, the algorithm tries to impose a maximum playback buffer use rate on the user. This can be done with a delay between using two subsequent frames which is calculated in (6).

$$\text{Playback Delay}(i) = [TH - (Np(i) + Ra/Ru)] \times C \tag{6}$$

TH is the minimum threshold used for playback buffer. Np is the number of packets in playback buffer while using the ith packet. Ru is the usage rate of playback buffer frames. Ra represents the arrival rate of packets at the user and C is the playback delay constant which is considered to be 1 ms for our scheme. The delay should be found such that affect the video quality in an extremely controlled way. For this purpose an upper bound for this delay is assumed to be 10 ms, slightly more than 5ms which is acceptable in 12 fps playback, and if the delay value after calculation is more than this value, it is set to be equal to this value. So the final calculation of playback delay would be:

$$\text{Playback Delay}(i) = \min(10\text{ ms}, [TH - (Np(i) + Ra/Ru)] \times C) \tag{6}$$

The calculations are assumed to be done in a different thread in parallel with adaptation scheme in application level to decrease processing time. Table 2 shows the decision making outcomes in the application level for all possible combinations of previously discussed parameters.

4. Simulation and results

We implemented the proposed algorithm using NS-2 network simulation environment. We tried to meet the actual delay, bandwidth and rates used in mobile communications in our simulation. The added parts to the NS-2 network simulation environment to implement this characteristics and the AWEQ method are shown in Fig. 4.

We considered the following network topology and a two-part scenario for the test. The first part of the test scenario was to evaluate the AWMECN performance. We run the simulations using network simulator, NS-2 [4]. The general setting for the network feedback test is assumed to be like the one in Fig. 5. We tested the performance with different numbers of source and destination nodes.

A loss generating module was used to generate loss function and disconnection durations with variable intervals of disconnection. This module is implemented in NS-2 and is used to simulate the lossy link state and disconnection durations caused by mobility and handoffs. The user can change the lossy link state simulation parameters by altering the loss probability and mean disconnection duration input to this function. The receiver has no additional information about the possibility, time and duration of disconnection.

The AWMECN mechanism helps to avoid transmission of extra data which is lost and retransmitted in the other similar algorithms. It also helps to reduce the transmission time by avoiding the effect of drastic shrinks of window size in TCP-like mechanisms on video stream. This would help in better user perceived video quality as well as reduce transmission duration which is a factor in energy consumption in wireless devices.

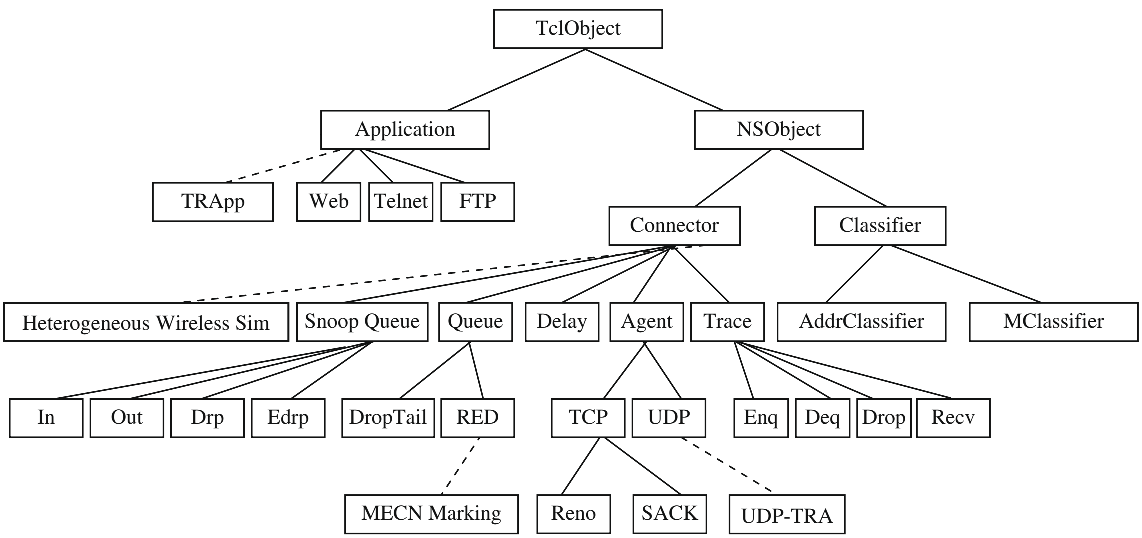

Fig. 4. Parts added to NS-2 for simulation of the AWEQ algorithm.

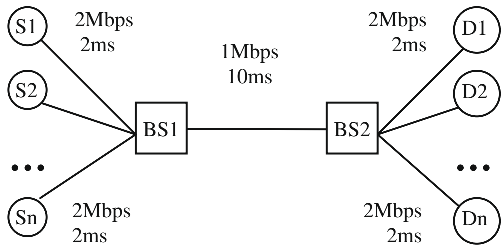

Fig. 5. Simulation network setting.

Based on the facts discussed above avoidable retransmitted data and connection duration are used to be the main metrics used for evaluation of the mechanism. These parameters are compared with those of ECN and WMECN, similar mechanisms for explicit notification of network state. Evaluation of total avoidable data in ECN-using schemes is shown in comparison with the proposed method in Fig. 6. It is obvious that the proposed mechanism shows a great enhancement in avoiding retransmissions.

Transmission duration is also used for both evaluations of performance and average power consumption estimations. Different scenarios used for evaluation of the mechanism, show an average reduce of 1.9% in transmission duration compared with similar methods. Although the protocol is used in the application level and it may be sensible to think that application level adaptation will cause a prolonged communication, the proposed method shows a good communication duration and low extra data so leading to an efficient power consumption scheme. Data transmission duration curves for AWMECN and WMECN [5] are shown in Fig. 7.

As discussed before, the main advantage of this method is in the fact that it adds features that could not be supported in transport layer which result in short communication duration and less retransmitted data as well as improved quality achieved by the user.

The second part of the test was the evaluation of the whole mechanism. We considered the video traffic is also CBR traffic with the rate of 0.1 Mbps, as discussed in [2]. Playback buffer occupancy and variations are used as indicator for acquiring the user-perceived quality. Some other sources of traffic are used just to congest the network. Arrival rate at the user and network backlog are used as indicators of network status. All these information together are used to have a good sense of what is happening in the network.

Application Level MECN, is using the WMECN [5] marking mechanism in network layer in companion to decision making algorithm which works in application layer. For this purpose, an application layer header is added on UDP packets. This header contains the sequence number, timing fields and MECN field as well as an ACK field used for application level acknowledgement for received packets.

The overall performance of the proposed algorithm was measured by means of the playback buffer as an indicator of user-perceived quality. As it is obvious in Figs. 8 and 9, the variations in playback buffer are very low in the proposed mechanism and it has not underflow below a definite threshold while the non-adaptive algorithm used in comparison, which is assumed to be the normal non-adaptive application over the same underlying layers, let the playback buffer totally underflow to the value of zero. The comparison is considered to be fair, since it assumes same underlying layers and it adds the adaptive mechanism on the top of the same basic settings.

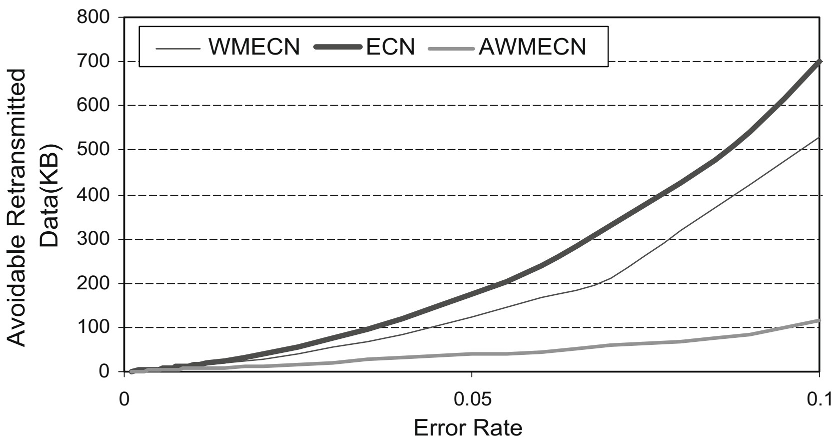

Fig. 6. Avoidable retransmitted Data.

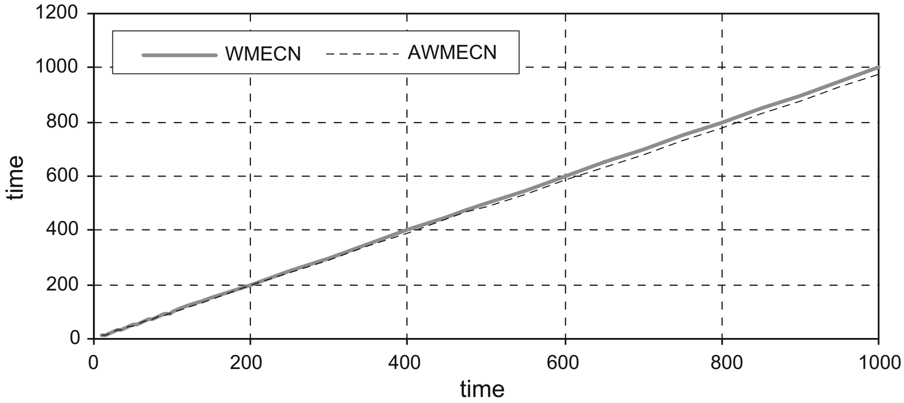

Fig. 7. Data transmission duration.

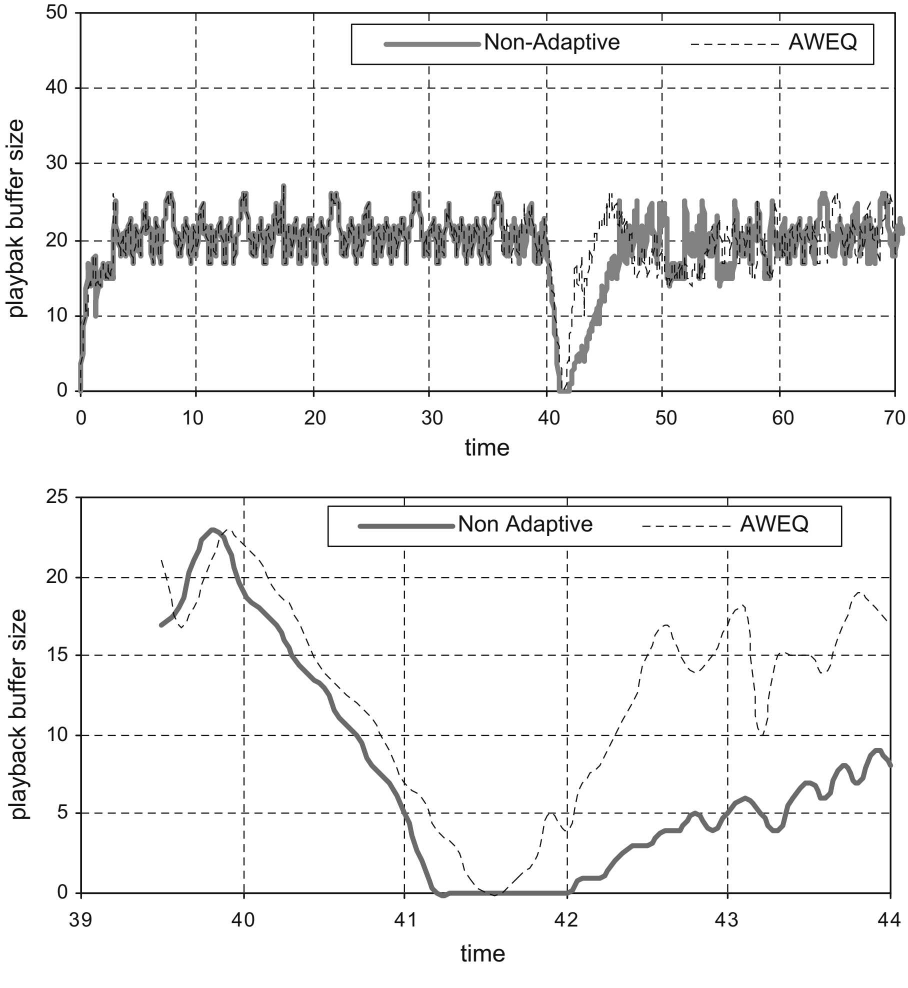

Fig. 8. Playback buffer occupancy during a disconnection.

Frame size is considered to be 500 bytes for simplicity. The other way to avoid the underflow in the playback buffer would be to either request a lower quality or to increase start-up delay and accumulate more frames before the playback starts. We did this by just having a graceful degrade in the receiver and requesting a lower rate from the sender. This avoids the stop of the video in the receiver and having a start-up delay and a long pause in play. The proposed algorithm is also more realistic

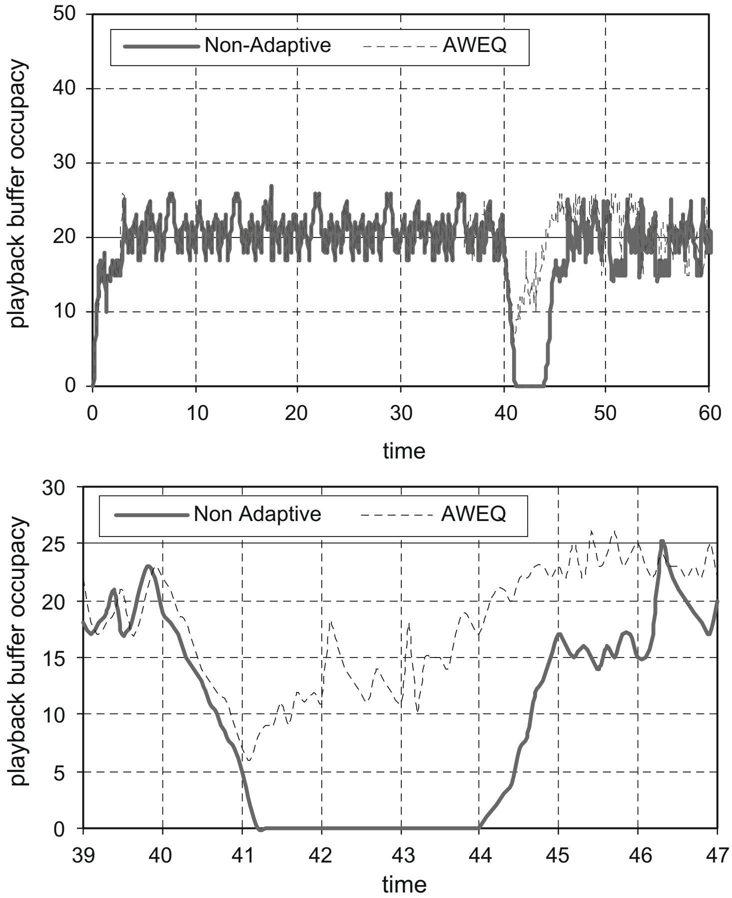

Fig. 9. Playback buffer occupancy during a congestion.

than the other proposed algorithms in that it has taken into account all the possible problems and the implementation has no special circumstances to be useful.

5. Conclusion

In this paper we studied a way to overcome the video streaming problems in mobile wireless networks and provided an algorithm that shows better QoS and multimedia delivery over wireless networks. The proposed mechanism aims at adaptive end-to-end QoS support so it is classified in pure end-to-end protocols category.

All the simulations are done using CBR traffic. The simulation settings are tried to meet real values and conditions.

We used a modification of WMECN which is used in application level to provide an application level notification of loss, congestion, and lossy link state. Coupling this feedback with an efficient adaptation mechanism in application layer was used for obtaining a reliable multimedia transmission over wireless networks. All of the adaptive decisions are made in the receiver in application level. Then the results of these decision making procedure is transmitted to sender via communication messages in the form of small acknowledge packets send for each block of data. The sender then applies the requested changes in rate and quality. The transport layer protocol used for this communication is UDP which is the best option for multimedia transmission over wireless networks. Sequence numbering, frame acknowledge, time stamping and adaptation schemes are added to have a perfect protocol over UDP. The proposed scheme has shown a better performance than other video transmission mechanisms in transport layer.

Finally, we conclude that it is a better engineering decision to choose best protocols in each layer and using adaptation in application layer than to require multimedia applications to use a separate specialized transport protocol other than TCP, given the many benefits of using TCP and also having same QoS problems unsolved for different scenarios. We see AWEQ as a promising definition of such an algorithm and hope to see it incorporated in the standard implementations for video delivery over heterogeneous wireless networks.

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