A Cross-layer Design for Video Transmission with TFRC in MANETs

George Adam[1,2], Christos Bouras[1,2], Apostolos Gkamas[3],
Vaggelis Kapoulas[1,2] and Georgios Kioumourtzis[4]
[1]*Computer Technology Institute & Press "Diophantus", Patras, Greece*
[2]*Computer Engineering & Informatics Dept, Univ. of Patras, Patras, Greece*
[3]*University Ecclesiastical Academy of Vella, Ioannina, Greece*
[4]*Center for Security Studies, P. Kanellopoulou 4, 10177, Athens, Greece*
adam@ceid.upatras.gr, {bouras, kapoulas}@cti.gr, gkamas@aeavellas.gr, gkioumou@gmail.com

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Abstract: Mobile Ad hoc NETworks (MANETs) are becoming more essential to wireless communications due to growing popularity of mobile devices. However, MANETs do not seem to effectively support multimedia applications and especially video transmission. In this work, we propose a cross-layer design that aims to improve the performance of video transmission using TCP Friendly Rate Control (TFRC). Our design provides priority to video packets and exploits information from the MAC layer in order to improve TFRC's performance. The proposed cross-layer mechanism utilizes Signal to Noise Ratio (SNR) measurements along the routing path, in order to make the route reconstruction procedure more efficient. Simulation results show that both the use of traffic categorization and the SNR utilization lead to important improvements of video transmission over the mobile Ad hoc network. More specifically, simulations indicate increased average Peak Signal to Noise Ratio (PSNR) for the received video, increased throughput and packet delivery ration, as well as reduced average end-to-end delay.

1 INTRODUCTION

A fundamental difference in MANETs, compared to other infrastructure-based wireless networks, is that a mobile node could act also as a router while having the possibility of being the sender or receiver of information. The ability of MANETs to be self-configured and form a mobile mesh network, by using wireless links, make them very suitable for a number of cases that other type of networks cannot operate. An important usage scenario of MANETs can be a disaster area or any kind of emergency, in which the fixed infrastructure has been destroyed or is very limited. However, this ability results in a very dynamic topology in which routing becomes a very complicated task. The impact of this dynamic topology on multimedia applications and especially on video streaming applications is high latency when a wireless link breaks, and as a result routing protocols should find alternate paths to serve applications. Therefore, under these constrains there should be in place additional mechanisms to minimize latency in video streaming applications.

On the other hand, video streaming applications use UDP as the transport protocol for video packets. Although this is an obvious solution to avoid latency caused by the retransmission and congestion control mechanisms of TCP, it may cause two major problems. The first one has to do with possible bandwidth limitations in which uncontrolled video transmission without any congestion or flow control may lead to increased packet losses. The second issue relates to TCP-friendliness. Under some conditions uncontrolled video transmission may lead to possible starvation of TCP-based applications running in the same network.

The research community in order to address these issues came with new proposals to provide congestion control schemes based on those that are already successfully implemented in TCP. However, the proposed congestion control schemes are mainly designed for use in wired networks, in which packet losses primarily occur due to congested links. In wireless networks the cause of packet losses is mainly due to interference in the wireless medium. Therefore, one needs to differentiate congestion packets losses from random packet losses (Vazão et al., 2008). To this direction a number of various versions of TCP have been proposed including TCP Veno (Cheng and Liew, 2003), TCP New Jersey

(Xu, Tian and Ansari, 2005) and TCP NCE (Sreekumari and Chung, 2011). In another work (Shagufta, 2009), the impact of TCP variants on the performance in MANETs routing protocols is investigated.

The most well-known congestion control mechanism that can be used on top of other transport protocols, such as UDP, is the TCP-friendly Congestion Control (TFRC) (Handley, Floyd, Padhye and Widmer, 2008), which is already an international standard. However, even TFRC is facing some limitations in wireless environments and especially in MANETs. In (Chen and Nahrstedt, 2004) these limitations are studied and it is shown that TFRC can be used in MANETs only when strict throughput fairness is not a major concern. Moreover, they analyze several factors contributing to TFRC's conservative behaviour, many of which are inherent to the MANET network. While their study reveals the limitations of applying TFRC to MANETs, they address the open problem of multimedia streaming in these networks and propose an alternative scheme based on router's explicit rate signalling and application's adaptation policies.

In order to overcome the above limitations an algorithm is proposed in (Li, Lee, Agu, Claypool and Kinicki, 2004), which is termed as Rate Estimation (RE) TFRC, and it is designed to enhance TFRC performance in wireless Ad hoc networks.

A large variety of research has been conducted regarding the usefulness of the wireless medium-related metrics. In (Zhang et al., 2008) a systematically measurement-based study on the capability of SNR is performed to characterize the channel quality. Although it is confirmed that SNR is a good prediction tool for channel quality, there are also several practical challenges.

Our motivation in this paper is to address the aforementioned technical issues, making video streaming in MANETs a promising application area. To this direction, we design a cross-layer mechanism that:

- Provides priority to video packets against other data packets.
- Implements TFRC to provide congestion and transmission rate control to video applications.
- Enhances routing operation with additional wireless medium-related metrics in order to improve the wireless transmission performance.

The proposed cross-layer mechanism is tested through simulations.

The rest of the paper is organized as follows: In section 2 we discuss the cross-layer design. In section 3 we provide an analysis of the proposed mechanisms. In Section 4 we briefly discuss the simulation environment under which we evaluate our cross-layer design. In Section 5 we present the simulation results. Finally, we conclude the paper in Section 6 with plans for future work.

2 CROSS-LAYER DESIGN

The proposed cross-layer design is based on the attributes of voice and video streaming applications, which are characterized by different tolerance in terms of end-to-end delay. A real time service, like video transmission, requires much less delay jitter values than a file transfer application. A way to minimize delay jitter is to prioritize traffic and to adapt the routing procedures depending on application requirements. The proposed cross-layer design invokes three layers in which we apply our adaptations

At the MAC layer, we differentiate the access of various applications with the use of the IEEE 802.11e protocol (IEEE Std., 2005), based on Quality of Service (QoS) criteria. Therefore, the IP packets are marked based on the underlying application type. This is a simpler task in mesh networks than in wired networks with fixed infrastructure, in which different administrative domains may exist in a path between video sender and receiver(s). Ad hoc networks provide this flexibility as every node in the network acts also as router. The main function for providing QoS support in IEEE 802.11e protocol is the Enhanced Distributed Coordination Function (EDCF). This function is responsible for managing the wireless medium in the Contention Period (CP) and enhances the Distributed Coordination Function (DCF) function of the legacy IEEE 802.11 protocol. The priority of each Traffic Class (TC) is defined by the following parameters:

- The transmission opportunity (TXOP), which stands for "*the time interval when a station has the right to initiate transmission, defined by a starting time and the maximum duration*". It is measured in milliseconds.
- The Arbitration Interframe Space (AIFS), which is at least DCF Interframe Space (DIFS) long. When the AIFS is represented by a number n instead of time, it is calculated according to the following equation:

$$AIFS = SIFS + n * SlotTime \tag{1}$$

- The minimum value of the Contention Window (CW)
- The Persistence Factor (PC), which is used to increase the CW after any unsuccessful retransmission and this CW is different for each TC.

The source code (Wiethölter and Hoene, 2003) used in this work is compliant with the specifications of the IEEE 802.11e protocol but supports only up to four different data Traffic Categories (TCs). In the latest IEEE 802.11e standard, the protocol can support up to eight different TCs but we regard the current implementation with four TCs for our work as sufficient enough. Table 1 outlines the different QoS parameters for the four TCs.

Table 1: QoS parameters for the four TCs in IEEE 802.11e.

	TC[0]	TC[1]	TC[2]	TC[3]
PF	2	2	2	2
AIFS	2	2	3	7
CW_MIN	7	15	31	31
CW_MAX	15	31	1023	1023
TXOP limit	0.003	0.006	0	0

At the network (routing) layer we utilize SNR information for improving the routing performance. We use the Ad hoc On-Demand Distance Vector (AODV) (Perkins and Belding-Royer, 2003) routing protocol which is among the most popular ad hoc routing protocols and is capable for both unicast and multicast routing. AODV is a reactive routing protocol that is based on the Bellman-Form algorithm. In general, the reactive protocols search for a routing path between nodes only on demand. The advantage of this method is that utilizes low network bandwidth and does not introduce routing overhead when data transmission is not required. In contrast, proactive protocols establish and maintain routing paths for nodes even if there is no need to transfer data. This allows lower latency but requires higher network management cost.

AODV uses originator and destination sequence numbers to avoid both "*loops*" and the "*count to infinity*" problems that may occur during the routing calculation process. As a reactive routing protocol, it does not explicitly maintain a route for any possible destination in the network. However, its routing table maintains routing information for any route that has been recently used, so a node is able to send data packets to any destination that exists in its routing table without flooding the network with new Route Request messages. In cases where the mobility is high, the routing paths need to be reconstructed frequently. For this purpose, we introduce a mechanism that utilizes the SNR measurements along the routing path, in order to make the reconstruction procedure more efficient. In essence, the reduction of the measured SNR, may signify that the relative nodes are travelling further apart from each other and a disconnection of the link between them is eminent. At this stage the cross-layer design enables in advance the route reconstruction process to avoid the temporary disconnection.

At the application (APP) layer we implement TFRC for congestion control with enhanced functions to improve the estimations of TFRC and to better utilize the available bandwidth. To do so, we use feedback information from the receiver. The TFRC feedback packet is modified in order to include the SNR measurements along the routing path. Moreover, we consider rate adaptive video transmission for scaling among different qualities to achieve better bandwidth utilization. This adaptation is also achieved by utilizing the reception rate and packet loss estimation that TFRC feedback provides.

The proposed cross-layer design with adaptations at the MAC, Network and Application layers is depicted in Fig. 1.

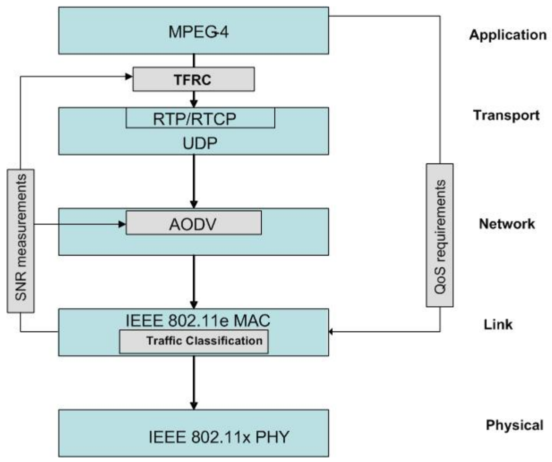

Figure 1: Proposed cross-layer design.

3 MECHANISM ANALYSIS

TFRC is a congestion control mechanism which is designed for unicast flows that compete with TCP traffic. Compared to TCP, TFRC has lower variation of throughput over time, so in many cases is more suitable for multimedia applications. However, TFRC should be used when there is a need for smooth throughput as it responds slower than TCP to changes in the network conditions. It is designed for rate adaptive applications that use fixed size packets and can increase or decrease the sending rate. TFRC is a receiver-based mechanism which means that the congestion control information is calculated at the receiver side and then it is sent to the sender using a feedback message. Our proposed adaptation extends this feedback message with SNR information.

The use of the SNR measurements in MANETs is not straightforward. A transmission path in a multi-hop topology consists of many single links with different quality. This heterogeneity is affected by nodes' hardware or the distance of each one wireless link. Therefore, one difficulty is that there are more than one SNR measurements that can be exploited, but there is no provision in the existing protocols to “carry” this information along with other information to the sending and receiving nodes. However, once a routing path is established then the transmission quality can be degraded even if only a single link of the multi-hop communication is degraded. In this environment, the link with the lowest quality directly affects the total quality of the routing path. To overcome the above difficulty, the proposed mechanism maintains only the minimum SNR measurement along the multi-hop path (which can be more easily attached to a packet with video information). This information is then made available to TFRC protocol and it is included in the next feedback report, so that both sender and receiver are aware of the link quality. The feedback message contains the following information:

- The timestamp of the last data packet received.
- The delay between the last received data packet and the generation of the feedback report.
- The rate at which the receiver estimates that data was received since the last sent feedback report.
- The receiver’s current estimate of the loss event rate.
- Minimum SNR along the routing path.

The TFRC feedback report is utilized to adapt the rate of the video transmission and also to maintain the routing path quality to high levels. For this purpose, the proposed mechanism implements a TFRC feedback handling algorithm (Algorithm. 1). Firstly, the mechanism extracts the receiver address and the minimum found SNR and then a comparison with a predefined SNR threshold is made. If the received SNR is found to be lower than the threshold, meaning that the end to end connection is likely to be lost, then a new route discovery procedure is initiated. Moreover, a simple timer is exploited in order to avoid very frequent routing path discoveries. This means that a new discovery procedure is allowed to be executed only if the timer has expired.

Algorithm 1: Modified TFRC feedback handling algorithm.

```
ModifiedRecvTfrcFeedback(feedback_packet) {
   snr = get_snr(feedback_packet);
   receiver_address = get_source_address(feedback_packet);
   if (snr < SNR_THRESHOLD and
       TIMER_EXPIRED = true) {
     routing_record = routing_table.lookup(receiver_address);
     schedule_update_routing_path(routing_record);
   }
   X_recv = data_reception_rate(feedback_packet);
   p = estimated_loss(feedback_packet);
   adapt _transmission(receiver_address, X_recv, p);
}
```

4 SIMULATION ENVIRONMENT

For the simulation experiments the ns-2 simulator is used. The simulation environment is extended in order to support the mechanisms which described in the previous section.

In order to conduct a number of realistic experiments with real video files we use the Evalvid-RA (Lie and Klaue, 2008) tool-set in conjunction with ns-2. Evalvid-RA is a framework and tool-set to enable simulation of rate adaptive VBR video. It has the capability to generate true rate adaptive MPEG-4 video traffic with variable bit rate. The tool-set includes an online (at simulation time) rate controller that, based on network congestion signals, chooses video quality and bit rates from corresponding pre-processed trace files.

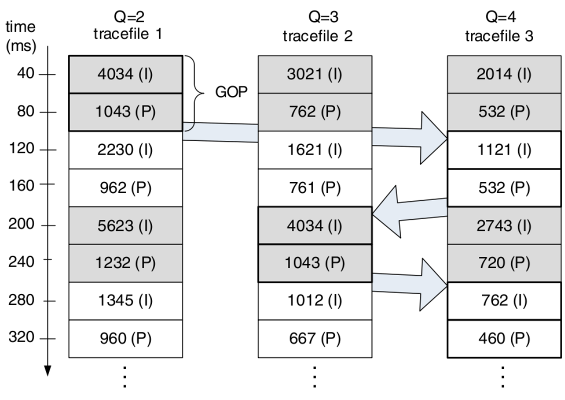

Figure 2: Simulation-time rate controller of Evalvid-RA.

As shown in Figure 2, the Evalvid-RA rate controller that is executed at simulation time chooses correct frame sizes (emphasized boxes) from different trace files. These files represent different video qualities for each quantizer scale. The same figure shows an example of a video transmission with 25 fps and three pre-processed qualities. The GOP size is 2 with the sequence of one I and one P frame.

For our simulations, we use a YUV raw video, which consists of 9144 frames and has duration of 366 seconds. We encode this raw video with the ffmpeg (Tomar, 2006) video encoder to produce an MPEG-4 video file. The frame size is set to 176x144 pixels, which is known as the Quarter Common Intermediate Format (QCIF). The temporal resolution is set to 25 frames per second with Group of Pictures (GoP) size equal to 12. After the simulation, we reconstruct the received video file and perform a frame-by-frame comparison between the original transmitted and the received video file in order to evaluate the quality of the received video.

The mobility model that is studied is based on the Manhattan city model with uniform sized building blocks. Manhattan grid mobility model can be considered as an ideal model to represent the topology of a big city. The simulation area is 500x500 meters in a 5x5 grid. Inside this area, there are 50 mobile nodes representing moving vehicles that are actually the transmitters and receivers of the information. The moving speed varies from 0 to 10m/sec, having a mean value of 4m/sec.

The simulations include some low rate background traffic between the moving nodes. Each node transmits in Constant Bit Rate (CBR) mode an amount of 2,560 bytes per second. Table 2 summarizes the simulation parameters that are used.

Table 2: Simulation parameters.

Mobility model	Manhattan Grid Model
Simulation duration	366 seconds
Number of nodes	50
Simulation area	500 x 500m
Node speed	0 – 10 m/sec (random)
Antenna	OmniAntenna
Data rate	2Mbps
Video bitrate	32kbps – 2Mbps (variable)

5 PERFORMANCE EVALUATION

The performance of the proposed cross-layer mechanism is evaluated in three scenarios.

- First, we evaluate the video transmission without any traffic prioritization in the MAC layer.
- Then, we introduce the IEEE 802.11e protocol in order to prioritize the video traffic against the background traffic.
- The last simulation utilizes the SNR mechanism for further performance enhancement.

A number of simulations have been conducted, in order to investigate the affect of the SNR threshold on the perceived video quality by the end user. For this purpose we calculate the Peak Signal to Noise Ratio (PSNR) by directly comparing the video file sent by the sender with the same file at the end user on a frame-by-frame basis. Equation 2 gives the definition of PSNR between the luminance component Y of source image S and the destination image D:

$$PSNR(n)_{dB} = 20\log_{10}\left(\frac{V_{peak}}{\sqrt{\frac{1}{N_{col}N_{row}}\sum_{i=0}^{N_{col}}\sum_{j=0}^{N_{row}}\left[Y_S(n,i,j)-Y_D(n,i,j)\right]^2}}\right) \quad (2)$$

where

$V_{peak} = 2^k - 1$, k = number of bits per pixel (luminance component)

The selection of SNR threshold affects the efficiency of the routing path reconstruction. Choosing a low threshold may result to very late reconstruction, while choosing a high threshold may result to very frequent route discovery processes that will add routing overhead to the ad-hoc network.

For the evaluation of the performance of the proposed cross-layer design, we examine the PSNR of the received video, with respect to the original video, the average throughput, the packet delivery ratio, and the average end-to-end delay. The simulation results show that both the use of traffic categorization and the utilization of SNR mechanism lead to important improvements, in all the above metrics, during the video transmission over the MANET network.

More specifically, Figure 3 shows the PSNR measurements among different SNR thresholds. For the rest of the simulation, the SNR threshold is chosen to be 33.0 dB. It should be mentioned that the above PSNR measurements suggest a SNR threshold which may not be suitable for all network topologies and network conditions. The above PSNR measurements must be used in order to estimate the SNR threshold when someone plans to used the proposed mechanism in a network.

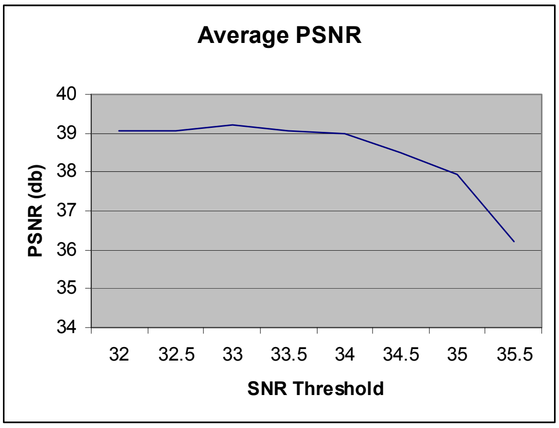

Figure 3: Average PSNR among different SNR thresholds.

As the cross-layer design intends to improve video transmission, the performance evaluation is focused in video related metrics.

In figure 4 the average PSNR is displayed for the three simulated scenarios. We can observe that the use of traffic categorization (with the use of 802.11e) leads to a small improvement of average PSNR but the utilization of the SNR mechanism leads to a significant improvement (more than 1.5 dB comparing with 802.11g) which is an important result.

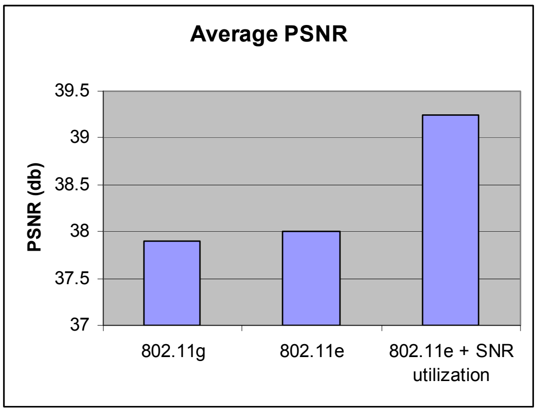

Figure 4: Average PSNR.

This means that for all kinds of frames the implementation of the proposed cross-layer design, that includes SNR measurements to better estimate the link quality, greatly reduces the video frame losses, and thus allows for a better video reconstruction at the receiver side. It is worth noting that without the implementation of the cross-layer design, the frame losses are at a level in which video reconstruction may not be possible at all at the receiver side. In contrast, the frame losses when the proposed cross-layer design is implemented are at level where video reconstruction can be done with only a few disruptions.

Figure 5 shows the average throughput during the three evaluation scenarios. Again the use of traffic categorization (with the use of 802.11e) leads to an improvement of throughput and the utilization of the SNR mechanism further leads to a significant additional improvement of throughput (more than 100Kbps comparing with 802.11g). We have to mention that the improvement in throughput is significant in terms of QoS from the end user perspective (PSNR measurements in Figure. 4) because a small increase in throughput can lead to significant improvement of the end user experience.

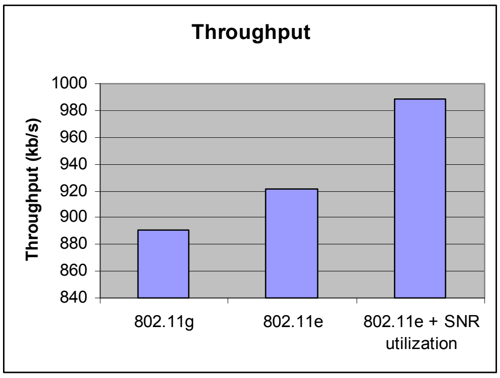

Figure 5: Average throughput.

Figure 6 shows the packet delivery ratio during the three evaluation scenarios. Similar conclusions as in the case of the average throughput can be inferred. Again, the use of traffic categorization (with the use of 802.11e) leads to a significant improvement of packet delivery ratio and the utilization of SNR mechanism leads to a small additional significant improvement of the packet delivery ration.

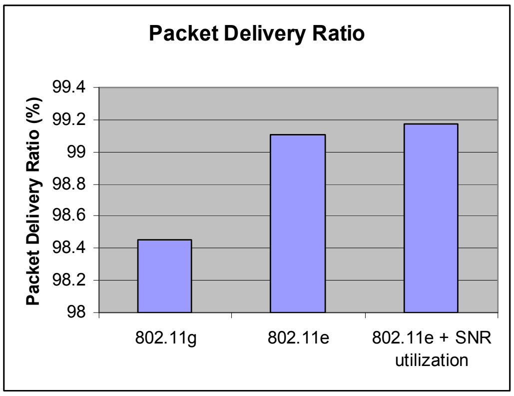

Figure 6: Packet delivery ratio.

Finally, figure 7 shows the average end-to-end delay during the three evaluation scenarios. Both the use of traffic categorization (with the use of 802.11e) and the utilization of the SNR mechanism lead to a significant improvement of average end-to-end delay. We have to mention that the above improvement in average end-to-end delay is very important for video streaming applications.

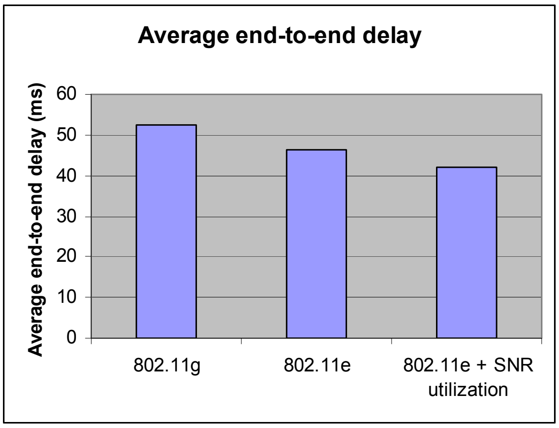

Figure 7: Average end-to-end delay.

The above results show that the use of both the traffic categorization and the SNR-utilizing cross-layer mechanism lead to important improvements during the transmission of multimedia data over the MANET network. This improvement can lead to a noticeable quality improvement of the video received, as subjectively judged by some viewers. This judgment verifies that the improvement can also be perceived by the users.

In addition the above results indicate that the use of the cross-layer design, can lead to significant improvements in video transmission in MANETs. These improvements can help make the difference in MANETs between an interrupted, low-quality video transmission and a usable video transmission service without perceived annoyances for the users..

6 CONCLUSION AND FUTURE WORK

We presented in this work a cross-layer design that aimed to improve the performance of video transmission with the use of TFRC. Our design provided priority to video packets and exploited information from the MAC layer (SNR) in order to improve the TFRC performance. Simulation results showed that the proposed cross-layer design led to improving performance, under several metrics, and could result in perceived improvements of the received video quality.

We also showed how a cross-layer design involving the Application, Network and the MAC layers can improve QoS in MANETs by sharing information between non-adjacent layers.

Our future work includes the implementation of an adaptive estimation for the appropriate SNR threshold based on network metrics. In addition, we plan to implement a prototype of the proposed cross layer design and evaluate it in a real MANET.

Another interesting extension of this work is to use the SNR measurements in order to provide TFRC with estimations of whether the observed packet loss is due to network disruption or due to congestion. This is expected to have a positive impact on the performance of TFRC and the video transmission in MANETs, as well.

Furthermore, we plan to investigate the (combined) use of new cross-layer designs and mechanisms in order to come up with a balanced set of improvements that could provide the best outcome. Finally, we plan to investigate the effect of the proposed design, and especially the use of SNR, in the performance of other routing protocols in MANETs.

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