Cross Layer Design for Video Streaming in MANETs

George Adam
Computer Engineering and Informatics Dept., University of Patras, Rion, Greece
Email: adam@ceid.upatras.gr

Christos Bouras
Computer Technology Institute & Press "Diophantus", Rion, Greece
Computer Engineering and Informatics Dept., University of Patras, Rion, Greece
Email: bouras@cti.gr

Apostolos Gkamas
University Ecclesiastical Academy of Vella of Ioannina, Ioannina, Greece
Email: gkamas@aeavellas.gr

Vaggelis Kapoulas
Computer Technology Institute & Press "Diophantus", Rion, Greece
Email: kapoulas@cti.gr

Georgios Kioumourtzis
Center for Security Studies, Ministry of Public Order and Citizen Protection, Athens, Greece
Email: gkioumou@gmail.com

***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 design utilizes 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 classification and the SNR utilization lead to important improvements in terms of end-to-end Quality of Service (QoS).**

***Index Terms*—Cross Layer Design; Multimedia transmission; TCP Friendly; Media Friendly; Congestion Control; SNR; Quality of Service.**

I. Introduction

Mobile ad hoc networks (MANETs) are becoming more and more essential to wireless communications due to growing popularity of mobile devices. A node in MANETs could act as a router while having also the possibility of being the sender or receiver of data packets. Their ability to be self-configured and form a mobile mesh network using wireless links, makes them suitable for a number of cases that other type of networks cannot fulfill the necessary requirements. MANETs offer the freedom to use mobile devices and move independently of the location of base stations (and outside their coverage) with the help of other network devices. The lack of predefined infrastructure makes them suitable in a number of mission critical applications. 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, there are certain limitations when we consider MANETs for real time applications and especially for video streaming. First of all, routing becomes a very complicated task in dynamic topologies. The routing protocols that have been developed for MANETs are directly affecting data transmission and the performance of the underlying applications. Each protocol in fact has its own routing strategy that is used in order to discover a routing path between two ends. The performance varies, depending on network conditions like the density of nodes in a specific area, their speed and direction. Therefore, it is obvious that the selection of the proper routing protocol for a specific network topology plays a critical role regarding the network performance.

Video streaming applications, on the other hand, 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.

doi:10.4304/jnw.9.2.328-338

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, where under some conditions uncontrolled video transmission may lead to possible starvation of TCP-based applications running in the same network.

One promising approach to address the aforementioned issues is the so called "cross-layer" design [35]. There have been proposed so far a number of cross-layer schemes for MANETs. However, there is still a lack of such schemes to directly support multimedia data transmission in an effort to increase the applicability of such applications.

Therefore, our motivation in this article is to address all the aforementioned issues, making video streaming in MANETs a promising application area. To this direction, we implement a cross-layer design based on the utilization of information from lower layers, which combines the following features:

- Provision to provide priority to routing and video packets against other type of data packets.
- Provision to implement congestion and flow control mechanism for the video streaming applications.
- Provision to enhance routing operations with additional wireless medium-related metrics in order to improve the wireless transmission performance.

The above design can be applied in applications with bandwidth and delay constraints, while keeping at a minimum level the requirements imposed by intermediate stations. The main contributions in this work are the cross-layer mechanisms that combine the features of the IEEE 802.11e protocol [1], the implementation of the TCP Friendly Rate Control (TFRC) [2] in order to provide congestion control features for the video streaming application and the modification of the Ad hoc On-Demand Distance Vector (AODV) [3] protocol in order to improve the route discovery process [4]. Another important contribution is the combination of network related metrics and video-centric metrics in an effort to better evaluate the effectiveness of the proposed design under a controlled environment.

The simulation results show that the proposed design increases the average Peak to Signal Noise Ratio (PSNR), the throughput, and the packet delivery ratio of the received video. In addition, the proposed design reduces the average end-to-end delay. As the performance evaluation section shows, the combination of the above mentioned modules can lead to significant improvements in terms of the end user experience.

The rest of the paper is organized as follows: In Section II we discuss the related work. In Section III we provide the description of the cross-layer design. The performance evaluation for the selection for the routing protocol in use is presented in Section IV. In Section V we discuss the simulation environment under which we evaluate the proposed cross-layer mechanism. Performance evaluation results are presented in Section VI. We conclude this work in Section VII with plans for future work in Section VIII.

II. Related Work

The research community in order to address congestion control issues came with new proposals 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 between the sender and receiver pair. In wireless networks the cause of packet losses is mainly due to interference in the wireless medium. Therefore, one needs to differentiate congestion packet losses against random packet losses [5]. To this direction a number of various versions of TCP have been proposed including TCP Veno [6], TCP New Jersey [7] and TCP NCE [8]. In another work [9], 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 TFRC, which is already an international standard. According to [2], TFRC is a congestion control mechanism for unicast flows operating in a best-effort Internet environment. It is reasonably fair when competing for bandwidth with TCP flows, but has a much lower variation of throughput over time compared to TCP, making it more suitable for applications such as telephony or streaming media where a relatively smooth sending rate is important.

However, even TFRC is facing limitations in wireless environments and especially in MANETs. In [10] 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, the authors analyze several factors contributing to TFRC's conservative behavior, many of which are inherited to MANETs. 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 signaling and application's adaptation policies.

In order to overcome the above limitations an algorithm is proposed in [11], which is termed as Rate Estimation (RE) TFRC, and is designed to enhance TFRC performance in wireless Ad hoc networks.

In the area of the routing protocols for ad-hoc wireless networks, a sufficient number of proposals have been presented by the research community. Each protocol has its own routing strategy and its performance varies depending on network conditions like the density of nodes in a specific area, their speed and direction. Most of these protocols do not take into account the limitations and the special requirements posed by the applications.

In [12], the effects of various mobility models on the performance of Dynamic Source Routing (DSR) [13] and AODV routing protocols are studied. The experimental results illustrate that the performance of a routing protocol varies across different mobility models, node densities and the length of data paths.

Another performance evaluation of three widely used mobile ad hoc routing protocols (Destination-Sequenced Distance Vector DSDV [14], AODV and DSR) with respect to group and entity mobility models is presented in [15]. Simulation results indicate also that the relative ranking of routing protocols may vary, depending on the mobility model.

In [16], a QoS-aware self-configured adaptive framework is presented to provide video-streaming services over MANETs. The routing algorithm periodically updates a set of paths, classifies them according to a set of metrics, and arranges a multipath-forwarding scheme. This proposal operates in a different way under highly dynamic states than under more static situations, seeking to decrease the probability of having broken links and improving the service performance, while using lower signaling overhead.

Matin and Naaji [17] addresses the use of multi-hop as an alternative to conventional single hop transmission in order to increase the quality of real time video streaming over MANETs. The use of the IEEE 802.11e Enhanced Distributed Channel Access (EDCA) function improves the overall performance of the high priority traffic in MANETs, by using the access control mechanisms of the MAC layer.

Calafate, Malumbres, Oliver, Cano and Manzoni [18] propose a QoS architecture for MANETs that seeks to alleviate the effects of both congestion and mobility on real-time applications. The proposed architecture includes the usage of IEEE 802.11e protocol in order to offer soft QoS support to MANETs heavily loaded by both best effort and QoS traffic.

In [19], priority assignment mechanisms are considered for implementing priority treatment of packets in a MANET using the DSR routing protocol based on a modified IEEE 802.11 MAC layer operating in the distributed mode. The mechanism includes priority queuing and several methods for providing important messages an advantage in contenting for channel access. In [20] an integrated cross-layer optimization algorithm is proposed in order to maximize the decoded video quality in a multi-hop wireless mesh network with QoS guarantees. It is investigated in [21] whether or not the operating conditions in a city are likely to permit video streaming. It is found that AODV outperforms DSR over the Manhattan grid model.

Finally, a large variety of research has been conducted regarding the usefulness of the wireless medium-related metrics. In [22] a systematically measurement-based study on the capability of to characterize the channel quality is presented. Although it is confirmed that SNR is a good indicator for channel quality, there are also several practical challenges.

III. Cross Layer Design

A. Description

The proposed cross-layer design is based on the attitudes 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 than for example a file transfer application. A way to minimize delay is to prioritize traffic and 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 [1], based on QoS criteria. Therefore, the IP datagrams 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 the 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 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:

$$AIFS = SIFS + n * SlotTime \qquad (1)$$

The source code [23] 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 I outlines the different Quality of Service (QoS) parameters for the four TCs.

TABLE I. 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. Based on the results of Section 4 in this paper, we use the AODV routing protocol which is among the most popular ad hoc routing protocols, is capable for both unicast and multicast routing and better facilitates the multimedia transmission mechanisms of the proposed mechanism. AODV is a reactive routing protocol that is based on the Bellman-Ford algorithm. It 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.

AODV 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 based on TFRC feedback mechanism.

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

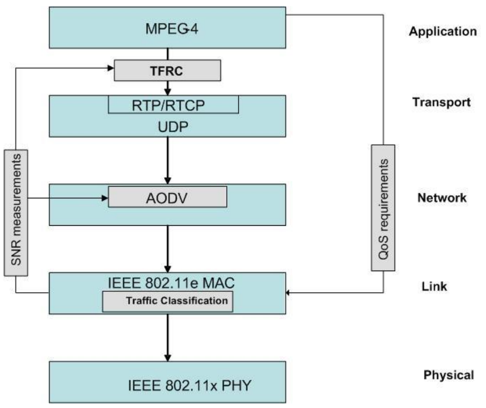

Figure 1. Proposed cross-layer design

B. 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 the information provided by the TFRC feedback messages 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 technical 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 issue, 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) at every transmission. 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 SNR measurement that is included in the feedback report is the latest one as it is assumed to be the most representative. Finally, 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 flooding the network with very frequent routing request messages. This means that a new discovery procedure is allowed to be executed only if the timer has expired. Although the timer threshold can be set to any value, it is suggested to be more than the expected RTT between sender and receiver and less than the maximum allowed latency. The default value that is used in our implementation is 2 seconds and was selected based on various simulations preformed and not included in this paper due to space limitation.

The routing path update function that is shown in Algorithm 1 initiates a discovery procedure. Typically, in AODV, a node disseminates a request message when it determines that it needs a route to a destination and an already existing route for this destination is not available at time. This happens when transmitting data to a destination for the very first time, or if an existing routing path is expired or invalid. By scheduling the routing discovery procedure earlier than AODV mechanism does, we can take advantage of the fact that the source and destination nodes can still communicate. This can be considered valuable especially for unreliable transport protocols.

```
ModifiedRecvTfrcFeedback(Feedback_packet) {
  Snr = get_snr(Feedback_packet)
  Receiver_address = get_source_address(Feedback_packet)

  If (NOW – last_discovery_time) > TIMER_THRESHOLD Then
    Timer_expired = True
  Else
    Timer_expired = False
 Endif

  If snr < SNR_THRESHOLD and Timer_expired = True Then
    Routing_record = routing_table.lookup(Receiver_address)
    schedule_update_routing_path(Routing_record)
  Endif

  X_recv = get_data_reception_rate(Feedback_packet)
  p = get_estimated_loss(Feedback_packet)
  adapt _transmission(Receiver_address, X_recv, p)
}
```

Figure 2. Modified TFRC feedback handling algorithm

IV. ROUTING PROTOCOL SELECTION

In this section we conduct a simulation-based performance evaluation in order to select the routing protocol that fits better to our design.

Routing protocols for ad hoc networks can be classified into three main categories. In Proactive routing protocols ([14], [24], and [25]), every node in the network has one or more routes to any possible destination in its routing table at any given time. Reactive routing protocols ([3], [13], and [26]) obtain a route to a destination on a demand fashion. When the upper transport layer has data to send, the protocol initiates a route discovery process, if such a route does not already exist, in order to find a path to the destination. In Hybrid routing protocols ([27] and [28]), every node acts reactively in the region close to its proximity and proactively outside of that region, or zone. Hybrid protocols take advantage of both reactive and proactive protocols, but may require additional hardware, such as GPS devices, separated or integrated into the communication device.

For the evaluation process, we select two reactive protocols, AODV [3] and DSR [13] and one reactive OLSR [25]. These protocols are selected as they are among the most popular routing Ad hoc protocols and good representatives of the respective categories. Another important factor that promotes the study of these protocols is that they have been already implemented and verified by the ns-2 user community.

A. Simulation set-up

Simulations were carried out by taking into account realistic conditions and using the ns-2.34 [29] network simulator. 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 simplified 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. The moving speed varies from 0 to 20m/sec, having a mean value of 15m/sec. For each connection, data traffic is generated at a constant bit rate, using packets of 512 bytes. The traffic is assumed to use Real-time Transport Protocol (RTP) [30] that is designed for audio and video delivery over IP networks. Table II summarizes the simulation parameters.

TABLE II. QOS PARAMETERS FOR THE FOUR TCS IN IEEE 802.11E

Routing Protocols	AODV, DSR, OLSR
Mobility model	Manhattan Grid Model
Simulation duration	900 seconds
Number of nodes	50
Simulation area	500 x 500m
Node speed	0 – 20 m/sec (random)
Antenna	Omni Antenna
Propagation Model	Two Ray ground
MAC	802.11g
Traffic	CBR
Application	RTP
Data packet size	512 bytes
Rate	64 packets/sec

B. Performance Evaluation Metrics

In our evaluation, we use four quantitative metrics which indicate the efficiency of the tested protocols especially with focus on multimedia data transmission. The selection is as follows:

Packet delivery ratio (PDR)

PDR is defined as the fraction of all the received data packets at the destinations over the number of data packets sent by the sources. This is an important metric in networks. If the application uses TCP as the layer 4 protocol, high packet loss at the intermediate nodes will result in retransmissions by the sources that will result in network congestion. If the application is using UDP, like multimedia applications, high packet loss can reduce the quality of end user experience.

Average end-to-end delay

End-to-end delay includes all possible delays in the network caused by route discovery latency, retransmission by the intermediate nodes, processing delay, queuing delay, and propagation delay. To average the end-to-end delay we add every delay for each successful data packet delivery and divide that sum by the number of successfully received data packets. This metric is important in delay sensitive applications such as video and voice transmission.

Packet delay variation

Packet delay variation (PDV), or jitter, is defined as the difference in end-to-end delay between selected packets in a single connection. Any lost packets are ignored from this metric. Like end-to-end delay, PDV is also important in the case of multimedia transmission and other delay sensitive applications.

Routing overhead

The routing overhead is defined as the number of all routing control packets sent by all nodes. This metric discloses how efficient the routing protocol is. Proactive protocols are expected to transmit higher number of control packets than reactive ones. The bigger the number of control packets is, the less efficient the protocol is.

C. Simulation Results

In the following set of simulations, we evaluate the performance when streaming with 256 Kbps data rate. Fig. 2 shows the packet delivery ratio of AODV, DSR and OLSR as a function of the number of connections.

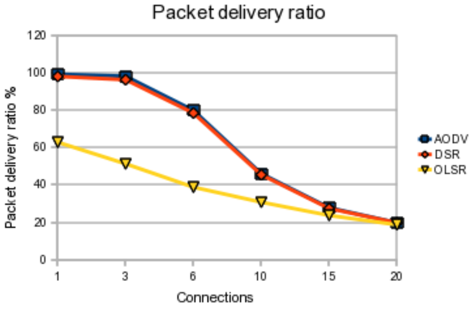

Figure 3. Delivery ratio over different maximum connections

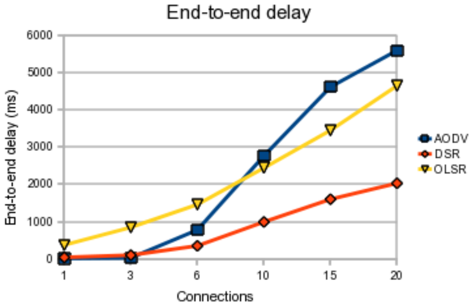

Figure 4. Average end to end delay (measured in milliseconds)

We can observe that the packet delivery ratio decreases when increasing the transmission sessions. AODV and DSR present identical performance while OLSR has the lowest performance. In the case of multimedia transmission, the OLSR does not seem to be suitable, as the packed delivery ratio is very low even when having only one stream. However, the reactive protocols present an acceptable ratio for up to 6 connections.

In parallel, the end-to-end delay is investigated with different number of connections. This metric is very essential when transmitting multimedia data as it affects the quality of the streaming video. For real-time multimedia services, the accepted threshold of delay can be considered to be approximately 150 milliseconds. As it is obvious from Fig. 3, the delay depends on the number of simultaneous connections.

It is interesting to observe that the increment on the OLSR protocol is almost linear, while in AODV is exponential. This is an expected behavior of a reactive protocol because AODV needs to update the routing table when a new connection is established. OLSR periodically updates its routing table and therefore seems to be a more efficient solution for delay-sensitive applications, like multimedia streaming. On the other hand, we can observe that DSR is a much more efficient reactive routing protocol than AODV for multimedia data transmission. Even DSR seems to be more efficient than OLSR for delay-sensitive applications.

Packet delay variation, or delay jitter, is used to measure the variance of the packet delay. In this metric, it is possible to have both positive and negatives values depending on the variation of the end to end delay. However, Fig. 4 shows the average packet delay variation in absolute values.

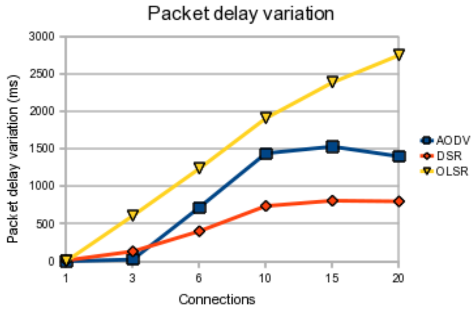

Figure 5. Packet delay variation (jitter)

The packet delay variation of the reactive routing protocols converges into an upper limit when increasing the connections above 10. In order to have high quality video and audio streaming it is important to have low packet delay variation. It is also interesting to observe that OLSR presents the lowest performance. One could expect that a proactive routing protocol like OLSR would reduce the packet delay variation at the destination node. However, our simulation results disclose that the existence of an up-to-date routing table cannot necessarily guarantee better performance in terms of delay. The main reason is that the periodic exchange of control packets occupies a noticeable portion of the available bandwidth and as a result, the transmission time for data packets increases. DSR and AODV leave more space for data packets and their performance seems to be independent from the number of connections in terms of packet delay variation. Once again DSR presents the best performance.

Fig. 5 depicts the routing overhead in terms of the number of routing packets that are transmitted. The comparison of the routing overhead that each protocol adds to the network shows that the proactive protocol OLSR has different behavior than the two reactive protocols. In OLSR, the number of routing packets depends only on the network size and not on the number of connections. We can also observe that DSR clearly outperforms AODV.

As the above results indicate, DSR and AODV perform better than OLSR having in mind the transmission of multimedia data over MANETs. OLSR seems to be ineligible for multimedia data transmission. For this reason at the next set of simulations, the focus is put on areas with high packet delivery ratio and acceptable values for end-to-end delay. As it is shown in Fig. 1, OLSR has very low packet delivery ratio; thus, the next comparison is conducted only by using AODV and

DSR, with a number between 3 and 6 connections. Therefore, we investigate the performance of AODV and DSR, when transmitting at different data rates.

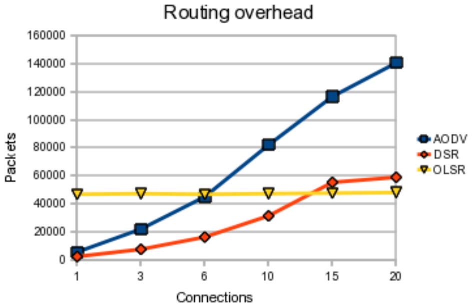

Figure 6. Routing overhead

Fig. 6 depicts the simulation results. We can observe that both reactive protocols AODV and DSR succeed similar performance.

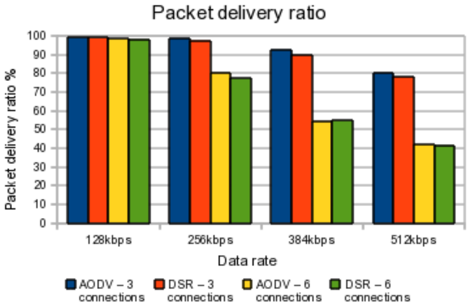

Figure 7. Delivery ratio over different data rates

Therefore we reach to the conclusion that the ratio is decreased when increasing the data rate or the number of connections. That means that either the number of simultaneous connections has to be limited, or the multimedia streaming has to be adapted (e.g. using lower rates) to the number of connections in order to succeed high packet delivery ratio.

The above comparative evaluations suggest that both AODV and DSR can be used for multimedia transmission over MANETs. DSR has a small advance comparing to AODV in the presented test-bed which consist of relative small number of nodes, low load, and mobility. In more complicated test-beds ([31], [32]) DSR outperforms AODV in less “stressful” situations, i.e., smaller number of nodes and lower network load, and/or mobility. However, AODV outperforms DSR in more stressful situations, with widening performance gaps in increasing stress (e.g., more network load, higher mobility).

It is worth to mention that AODV supports multicast communication which is considered an important factor for multimedia transmission in mobile environments. Moreover, AODV is considered very popular as it is implemented in many mobile devices and widely studied by the research community having inspired many later routing protocols. For the above reasons we select AODV as the MANET routing protocol for the implementation of our cross-layer design.

V. Simulation Environment

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

In order to conduct a number of realistic experiments with real video files we use the Evalvid-RA [33] 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.

As shown in Fig. 7, 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.

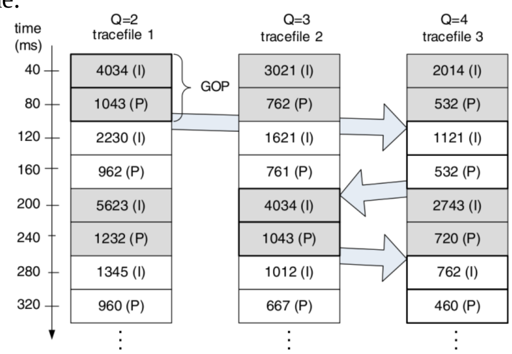

Figure 8. Simulation-time rate controller of Evalvid-RA

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 [34] video encoder to produce an MPEG-4 standard 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. 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. The introduction of background traffic is intended to trigger the routing processes in order to evaluate the routing performance. For that reason we introduce low rate background traffic to all nodes. Each node transmits in Constant Bit Rate

(CBR) mode an amount of 2,560 bytes per second. Table III summarizes the simulation parameters that are used.

TABLE III. 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)

VI. PERFORMANCE EVALUATION

The performance of the proposed cross-layer design is evaluated under three different scenarios.

- In the first scenario, we evaluate the video transmission without any traffic prioritization at the MAC layer.
- In the second scenario, we introduce the IEEE 802.11e protocol in order to prioritize the video traffic against the background traffic.
- In the last simulation scenario we utilize 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:

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

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

The selection of SNR threshold affects the efficiency of the routing path reconstruction. Choosing a low threshold may result in 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 mobile ad hoc network.

More specifically, Fig. 8 shows the PSNR measurements in conjunction with different SNR thresholds. We choose the SNR threshold to be 33.0 dB based on these simulation results. However, 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. Therefore, additional research is needed in order to dynamically define the suitable SNR threshold based on specific network attributes. This part left for future work.

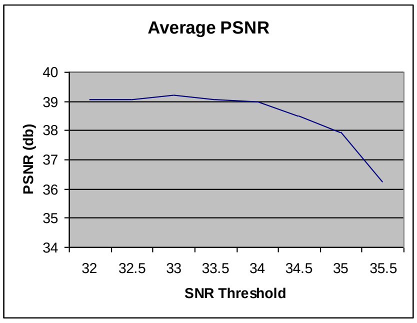

Figure 9. Average PSNR among different SNR thresholds.

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

In Fig. 9 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.

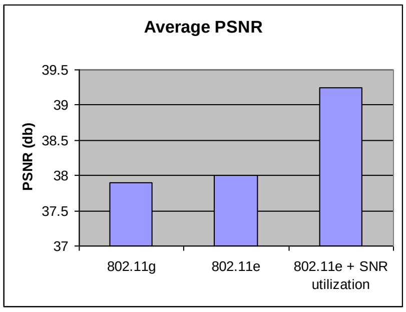

Figure 10. Average PSNR

This means that for all different type of frames the implementation of the proposed cross-layer design greatly reduces 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, in 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.

Fig. 10 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 Quality of Service (QoS) from the end user perspective

(PSNR measurements in Fig. 9) because a small increase in throughput can lead to significant improvement of the perceived end user experience.

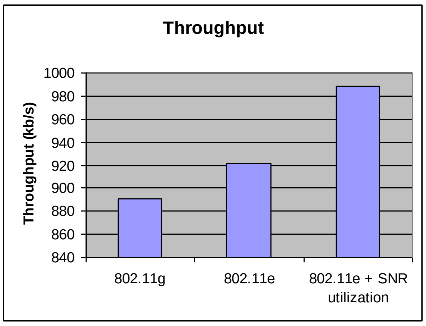

Figure 11. Average throughput

Fig. 11 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 improvement of the packet delivery ration.

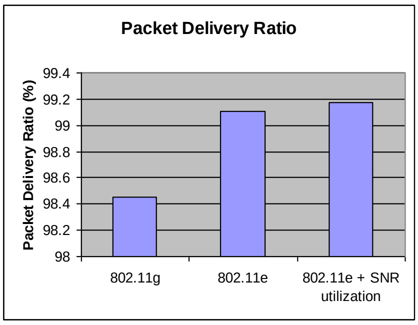

Figure 12. Packet delivery ratio

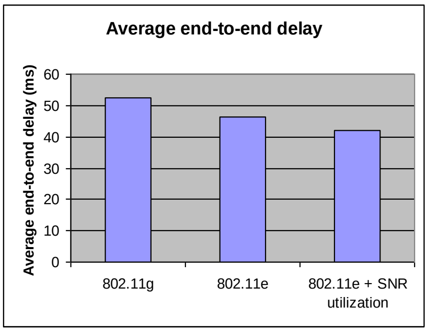

Figure 13. Average end-to-end delay

Finally, Fig. 12 depicts 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. As this metric is very important for delay sensitive applications like the video streaming, we need to mention that the above improvement in average end-to-end delay significantly improves the end user experience. In addition the improvement of average end-to-end delay has an important positive effect to real time delay sensitive applications like videoconference.

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 mobile ad hoc network. This improvement can lead to a noticeable quality improvement of the received video, as subjectively judged by some viewers. This judgment verifies that the improvement can also be perceived by the users.

In summary, the simulation results indicate that the use of the cross-layer design can lead to significant improvements of video transmission in MANETs. These improvements can make the difference in MANETs between an interrupted, low-quality video transmission and a usable video transmission service without perceived annoyances for the users. The results show that the proposed cross-layer design can address the limitations of MANETs regarding video streaming due to dynamic mobile topologies. In addition the proposed mechanism has an important positive effect both for real time (e.g videoconference) and non-real time delay sensitive applications (e.g. video streaming).

VII. Conclusions

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

The simulation results showed relative improvements, which disclose the possibilities of cross-layer design in MANETs. It is also important to notice that our evaluation for video transmission included several metrics which combined both network-centric and video quality metrics (PSNR).

The selection of the routing protocol in use disclosed that OLSR presented the lowest performance in terms of packet delivery ratio and jitter delay. The proactive behavior of a routing protocol cannot necessarily guarantee low jitter delay values although proactive protocols have always in its routing tables a possible path to any destination. Therefore, OLSR cannot be a proper choice for delay-sensitive applications.

VIII. Future Work

It is interesting to examine and evaluate the scalability of the proposed design among different node populations and mobile scenario models.

What is also left for future work is the implementation of an adaptive estimation of the appropriate SNR threshold based on the network attributes. We can also use the SNR measurements in order to locate the real cause of packet losses (network disruption or congestion). This is expected to have a positive impact on the performance of TFRC and it can further improve its rate adaptation mechanisms and video transmission in MANETs.

Furthermore, we plan to investigate the (combined) use of other cross-layer designs and mechanisms in order to come up with a balanced set of improvements that 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.

REFERENCES

[1] IEEE 802. 11 WG, Part 11: “Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications Amendment 8: Medium Access Control (MAC) Quality of Service Enhancements”.

[2] M. Handley, S. Floyd, J. Padhye and J. Widmer “TCP Friendly Rate Control (TFRC): Protocol Specification”, RFC 3448, September 2008.

[3] C. Perkins, E. Belding-Royer, “Ad hoc On-Demand Distance Vector (AODV) Routing”, RFC 3561, July 2003.

[4] G. Adam, C. Bouras, A. Gkamas, V. Kapoulas and G. Kioumourtzis, “A cross-layer design for video transmission with TFRC in MANETS”, *International Conference on Data Communication Networking* - DCNET 2012, Rome, Italy, 24-27 July 2012, pp. 5-12.

[5] T. Vazão, M. Freire, I. Chong, J. Yan, X. Zheng, J. Liu and J. Li, “Wireless Loss Detection for TCP Friendly Rate Control Algorithm in Wireless Networks”, *Information Networking. Towards Ubiquitous Networking and Services, Lecture Notes in Computer Science, Springer Berlin* / Heidelberg, 2008, pp. 113-122.

[6] C. Fu and S. Liew, "TCP Veno: TCP enhancement for transmission over wireless access networks", *IEEE Journal on Selected Areas in Communications*, vol. 21, no. 2, Feb 2003, pp. 216- 228.

[7] K. Xu, Y. Tian and N. Ansari, “Improving TCP performance in integrated wireless communications networks”, *Computer Networks*, Volume 47, Issue 2, Wireless Internet, ISSN 1389-1286, DOI: 10. 1016/j. comnet. 2004. 07. 006., 4 February 2005, pp. 219-237.

[8] P. Sreekumari and S. Chung, “TCP NCE: A unified solution for non-congestion events to improve the performance of TCP over wireless networks”, *EURASIP Journal on Wireless Communications and Networking* 2011, Published: 29 June 2011.

[9] S. Henna, “A Throughput Analysis of TCP Variants in Mobile Wireless Networks”, *In Proceedings of the 2009 Third International Conference on Next Generation Mobile Applications, Services and Technologies (NGMAST '09). IEEE Computer Society, Washington, DC, USA*, 2009, pp. 279-284.

[10] K. Chen and K. Nahrstedt, “Limitations of Equation-Based Congestion Control in Mobile Ad Hoc Networks”. *In Proceedings of the 24th International Conference on Distributed Computing Systems Workshops* - W7: EC (ICDCSW'04) - Volume 7 (ICDCSW '04), Vol. 7. IEEE Computer Society, Washington, DC, USA, 2004, pp. 756-761.

[11] M. Li, S. Lee, E. Agu, M. Claypool and R. Kinicki, “Performance Enhancement of TFRC in Wireless Ad Hoc Networks”, *In Proceedings of the 10th International Conference on Distributed Multimedia Systems (DMS), Hotel Sofitel, San Francisco, California, USA*, September 8 - 10, 2004.

[12] V. Timcenko, M. Stojanovic and S. Rakas, “MANET routing protocols vs. mobility models: performance analysis and comparison”, *Proceedings of the 9th WSEAS international conference on Applied informatics and communications*, 2009, p. 271-276.

[13] D. Johnson, Y. Hu and D. Maltz, “The Dynamic Source Routing Protocol (DSR) for Mobile Ad Hoc Networks for IPv4”, RFC 4728, February 2007.

[14] C. Perkins and P. Bhagwat, “Highly Dynamic Destination-Sequenced Distance-Vector Routing (DSDV) for Mobile Computers, ” *Computer Communications Review*, October 1994, pp. 234-244.

[15] B. Divecha, A. Abraham, C. Grosan and S. Sanya, “Impact of Node Mobility on MANET Routing Protocols Models”, *Journal of Digital Information Management*, 2007.

[16] M. Igartua and V. Frías, “Self-configured multipath routing using path lifetime for video-streaming services over Ad Hoc networks”, *Computer Communications* 2010, Volume 33, Issue 15, 2010, pp. 1879-1891.

[17] M. Matin and N. Naaji, “Performance Analysis with Enhanced Distributed Channel Access (EDCA) in IEEE 802. 11e for Real Time Video Streaming (MPEG-4) in Multi-hop MANET”, *Journal of Communication and Computer*. Vol. 7, no. 4, Apr il 2010, pp. 24-29.

[18] C. Calafate, M. Malumbres, J. Oliver, J. Cano and P. Manzoni, “QoS Support in MANETs: a Modular Architecture Based on the IEEE 802. 11e Technology, ” *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 19, no. 5, May 2009, pp. 678-692.

[19] X. Pallot and L. Miller, "Implementing message priority policies over an 802. 11 based mobile ad hoc network", *IEEE Military Communications Conf.* (MILCOM), 2001, pp. 860 - 864.

[20] Y. Andreopoulos, N. Mastronarde and M. van der Schaar, “Cross-Layer Optimized Video Streaming Over Wireless Multihop Mesh Networks”, Selected Areas in Communications, *IEEE Journal on Selected Areas in Communications, IEEE Journal on*, Vol. 24, No. 11, 2006, pp. 2104-2115.

[21] N. Qadri, M. Altaf, M. Fleury, M. Ghanbari and H. Sammak, "Robust Video Streaming over an Urban VANET", *IEEE International Conference on Wireless and Mobile Computing, Networking and Communications*, 2009, pp. 429-434.

[22] J. Zhang, K. Tan, J. Zhao, H. Wu and Y. Zhang, "A Practical SNR-Guided Rate Adaptation, " INFOCOM 2008. *The 27th Conference on Computer Communications. IEEE*, vol., no., 13-18 April 2008, pp. 2083-2091.

[23] S. Wiethölter and C. Hoene, "Design and Verification of an IEEE 802. 11e EDCF Simulation Model in ns-2. 26", *Technical Report TKN-03-019, Telecommunication Networks Group, Technische Universität Berlin*, November 2003.

[24] C. Chiang, H. Wu, W. Liu, and M. Gerla, “Routing in Clustered Multihop, Mobile Wireless Networks with Fading Channel”, *in Proceedings of IEEE Singapore International Conference on Networks (SICON)*, April 1997, pp. 197-211.

[25] T. Clausen and P. Jacquet, “Optimized Link State Routing Protocol (OLSR)”, RFC 3626, October 2003.

[26] V. Park, and M. Corson, “Temporally -ordered routing algorithm (TORA) version 1 functional specification”, *IETF Draft: draft-ietf-manet-tora-spec*-04. txt, July 2001.

[27] L. Wang and S. Olariu, "A Two-Zone Hybrid Routing Protocol for Mobile Ad Hoc Networks, " *IEEE Transactions on Parallel and Distributed Systems*, vol. 15, no. 12, Dec. 2004, pp. 1105-1116.

[28] B. Carp and H. Kung, "GPSR: Greedy Perimeter Stateless Routing for Wireless Networks", *Proceedings of the sixth annual ACM/IEEE International Conference on Mobile Computing and Networking*, 2000.
[29] The Network Simulator - ns-2: http://www. isi. edu/nsnam/ns/ (accessed online 24/10/2013)
[30] H. Schulzrinne, S. Casner, R. Rrederick, and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", *IETF RFC* 3550, July 2003.
[31] R. Misra and C. Mandal, "Performance comparison of AODV/DSR on-demand routing protocols for ad hoc networks in constrained situation", *ICPWC 2005. 2005 IEEE International Conference on Personal Wireless Communications*, 2005, 23-25 Jan. 2005, pp. 86 – 89.
[32] S. Das, C. Perkins and E. Royer, "Performance comparison of two on-demand routing protocols for ad hoc networks", *INFOCOM 2000. Nineteenth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE* 3 - 12 vol. 1, 26 March 2000 - 30 March 2000.
[33] A. Lie and J. Klaue, "Evalvid-RA: Trace Driven Simulation of Rate Adaptive MPEG-4 VBR Video", *Multimedia Systems, Springer Berlin* / Heidelberg, Volume 14, Number 1, 2008, pp. 33-50.
[34] S. Tomar, Converting video formats with FFmpeg, *Linux Journal*, v. 2006 n. 146, June 2006, p. 10.
[35] S. Shakkottai, T. S. Rappaport; P. C. Karlsson, "Cross-layer design for wireless networks, " *Communications Magazine*, IEEE, vol. 41, no. 10, pp. 74, 80, Oct 2003

George Adam was born in Athens, Greece in 1987. He graduated from the Computer Engineering and Informatics Department, University of Patras, Greece in 2010, and obtained his Masters Degree in Computer Science in 2013 from the same department. His major field of study was computer networks.

He is currently working for H&S Technology Solutions S.A. in Athens. His research interests include Information Retrieval, Data Mining and Mobile Ad Hoc Networks.

Christos Bouras was born in Kalamata, Greece in 1962. He graduated from the Computer Engineering and Informatics Department, University of Patras, Greece in 1985, and obtained his PhD in Computer Science in 1993 from the same department. His major field of study was computer networks.

He is Professor in the University of Patras, Department of Computer Engineering and Informatics. Also he is a scientific advisor of Research Unit 6 in Computer Technology Institute and Press "Diophantus", Patras, Greece. His research interests include Analysis of Performance of Networking and Computer Systems, Computer Networks and Protocols, Mobile and Wireless Communications, Telematics and New Services, QoS and Pricing for Networks and Services, e-learning, Networked Virtual Environments and WWW Issues.

Prof. Bouras has been a member of editorial board for international journals and PC member and referee in various international journals and conferences.

Apostolos S. Gkamas was born in Milea Metsovou, Greece at 04/09/1975. He obtained his Diploma (1998), Master Degree (2000) and Ph.D (2003) from the Computer Engineering and Informatics Department site of Patras University (Greece).

He is currently Assistant Professor in University Ecclesiastical Academy of Vella, Ioannina. His research interests include Computer Networks, Telematics, Multimedia transmission and Cross Layer Design. More particular he is engaged in transmission of multimedia data over networks and multicast congestion control. He has published more than 60 papers in international Journals and well-known refereed conferences. He is also co-author of three books (one with subject Multimedia and Computer Networks one with subject Special Network Issues and one with subject IPv6).

Dr. Apostolos S. Gkamas is member of Technical Chamber of Greece (TEE) and Pan-Hellenic Association of Qualified Computers Engineers and Information technology.

Vaggelis Kapoulas obtained his Computer Engineering diploma from the Computer Science and Engineering Department, School of Engineering, University of Patras, Greece in 1989,. He obtained his Ph.D. from the same department, in 2008.

He is currently an R&D engineer, and Head of Research Unit 6: Networks Telematics and New Services, Computer Technology Institute and Press "Diophantus", Patras, Greece. His research interests include Networks, Telematics and Algorithms for Distributed Systems. He has extended professional experience in the Analysis and Design of Networks, Applications of Networks and Advanced Network Services, Telematic Applications, Open & Distance Learning and Tele-working. He has participated in numerous (more than 50) R&D projects in several national and EU (co)funded programmes. He has published more than 10 articles in scientific journals and more than 50 research papers in various well-known refereed conferences.

Georgios Kioumourtzis obtained his Diploma from the Hellenic Army Military Academy and graduated as Signal Officer in 1986. In 1994, he extended his studies in the Telecommunications School for Signal Officers. He also received two Master Degrees in 2005, one in Computer Science and the second in Systems Engineering from the Naval Postgraduate School CA, USA. In 2010, he received his Ph.D from the Computer Engineering and Informatics Department of Patras University (Greece).

He is currently a research associate at the Center for Security Studies/Ministry of Public Order and Citizen Protection, Athens, Greece. His research interests include Computer Networks, Simulations, Multimedia transmission, Wireless Ad Hoc Networks and Cross Layer Design. He has published more than 25 research papers and journals in well-known refereed conferences.