INTERNATIONAL JOURNAL OF NETWORK MANAGEMENT
Int. J. Network Mgmt (2011)
Published online in Wiley Online Library (wileyonlinelibrary.com) DOI: 10.1002/nem.766

Power management adaptation techniques for video transmission over TFRC

Christos Bouras[1,2,*], Apostolos Gkamas[1,2], Vaggelis Kapoulas[1,2], Vassilis Papapanagiotou[1,2], Kostas Stamos[1,2,3] and Giannis Zaoudis[1,2]

[1] *Research Academic Computer Technology Institute, Patras, Greece*
[2] *Computer Engineering and Informatics Department, University of Patras, Greece*
[3] *Technological Educational Institute of Patras, Patras, Greece*

SUMMARY

In this paper, we describe power management adaptation techniques for wireless video transmission using the TCP Friendly Rate Control (TFRC) protocol that take into account feedback about the received video quality and try to intelligently adapt transmitting power accordingly. The purpose of the mechanisms is to utilize TFRC feedback and thus achieve a beneficial balance between power consumption and the received video quality. There are two power adaptation mechanisms proposed, each one with its own advantages. They both offer significant improvements when used in terms of both power consumption and received video quality. We use simulation in order to compare and evaluate our approach. Copyright © 2011 John Wiley & Sons, Ltd.

Received 23 December 2009; Revised 12 August 2010; Accepted 26 August 2010

1. INTRODUCTION

Networking complexity has led to the modularization of network architecture in layers. Traditional approaches focus on wired networks and try to separately optimize each network layer such as the physical, medium access, routing and transport layers. This approach reduces complexity and makes issues more manageable and architectures more flexible and upgradeable, but it may lead to suboptimal designs. Under this layered approach, communication occurs between two adjacent layers without taking into consideration the specific characteristics of multimedia applications. Although this layered approach has been the fundamental factor for the growth of the wired networks and the World Wide Web, it seems to pose constraints when attempting to adapt protocol behavior to multimedia application characteristics and to wireless network conditions. Therefore, a careful cross-layer approach, where selected communication and interaction between layers is allowed, can have performance advantages without negating the successful layer separation that has guided network design so far. A theoretical discussion of the cross-layer problem framework can be found in van der Schaar and Sai Shankar [1].

Wireless transmission differs in an important way from wired communication, in that the notion of the link is not as fixed and can vary depending on the movement of the communicating nodes, intermediate interference and the transmission characteristics of the communicating nodes, most notably their transmission power. While increased power generally correlates with a stronger signal and therefore improved transmission characteristics, in many wireless scenarios reduced power consumption is desired. This trade-off has been explored by various researchers studying Transmission

*Correspondence to: Christos Bouras, Computer Engineering and Informatics Department, University of Patras, 265 04 Rio Patras, Greece.
E-mail: bouras@cti.gr

Control Protocol (TCP) modifications [2–4] trying to combine reduced power consumption with increased data throughput. Wireless standards such as IEEE 802.11 specify power-saving mechanisms (PSM) [5], although studies have shown that PSM and other similar mechanisms carry a significant performance penalty in terms of throughput [6–8].

An important issue for the efficiency of wireless networks is to accurately determine the cause of packet losses. Packet losses in wired networks occur mainly due to congestion in the path between the sender and the receiver, while in wireless networks packet losses occur mainly due to corrupted packets as a result of the low signal-to-noise ratio (SNR), multi-path signal fading and interference from neighboring transmissions. A second difference between wired and wireless networks is the ‘mobility factor’. Mobility in wireless networks introduces a number of additional barriers in multimedia data transmission. Channel fading and handover time are the most important factors that cause packet losses as they introduce additional delays when the mobile user changes its location from one access point (AP) to another.

1.1. Multimedia transmission and oscillating rates

Over recent years a number of new protocols have been developed for multimedia applications in the whole Open Systems Interconnection (OSI) layer’s scale. The Moving Picture Experts Group (MPEG) protocol family includes the encoding and compression of multimedia data. The MPEG-4 protocol with fine granularity scalability (FGS), advanced video coding (AVC) and scalable video coding (SVC) enhancements provides adaptive video coding by taking into account the available bandwidth and is expected to be used by many multimedia applications. Moreover, congestion control and TCP friendliness pose additional design requirements as highly fluctuating (‘shark teeth’-like) transmission rates may be too difficult to be followed by audio–video (AV) encoders and decoders. TCP congestion control produces high fluctuations in the transmission rate which are not suitable for the current AV codecs which expect predictive and stable bandwidth allocation. Therefore, the development of protocols such as TCP Friendly Rate Control (TFRC) can be seen as a step to improve multimedia transmission. TFRC aims to achieve User Datagram Protocol (UDP) throughput efficiency, without stifling other network TCP flows. One way to cope with transient fluctuations of the transmission rate is with the use of buffers at the clients. However, an initial data pre-fetch in a buffer of more than 8 s before the player starts playing the stream is not easily accepted by the end-user. Moreover, in real-time video applications and conversational media large pre-fetch buffers are not acceptable. For multimedia applications smooth and steady transmission rates and short delay are more important attributes than guaranteed and on-order delivery of data packets.

1.2. The TFRC protocol

According to its specification, TFRC [9] is a congestion control mechanism for unicast flows operating in a best-effort Internet environment. It aims to be reasonably fair when competing for bandwidth with TCP flows, but at the same time achieving a much lower variation of throughput over time compared with TCP, thus making it more suitable for applications such as telephony or streaming media where a relatively smooth sending rate is important. For example, most video algorithms such as MPEG2 utilize the three major frame types (I-frames, P-frames, B-frames). The video bit rate tends to vary according to the complexity of the frame data; for example, an I-frame would be more complex compared to a P-frame as it results in more bits after compression. The same also applies to scene changes and high-motion scenes in a video sequence as they tend to incur a higher prediction error, which results in a lower compression efficiency. Thus a typical video bit rate will have occasional ‘pulses’. A smoothed transmission rate will reduce these ‘pulses’ and ends up affecting the video quality. Thus TFRC aims to prevent oscillatory behavior in environments with a low degree of statistical multiplexing. For this purpose, it is useful to modify the sender’s transmission rate to provide congestion avoidance behavior by reducing the transmission rate as the queuing delay (and hence round-trip time (RTT)) increases. To do this the sender maintains an estimate of the long-term RTT and modifies its sending rate depending on how the most

DOI: 10.1002/nem

recent sample of the RTT differs from this value. The long-term sample is *R_sqmean*, and is set as follows:

```
if no feedback has been received before
      R_sqmean = sqrt(R_sample);
   else
      R_sqmean = q2*R_sqmean + (1-q2)*sqrt(R_sample);
```

Thus *R_sqmean* gives the exponentially weighted moving average of the square root of the RTT samples. The constant *q2* should be set similarly to *q*, and a default value of 0.9 is recommended by the protocol's specification [10].

The sender obtains the base transmit rate, *X*, from the throughput function. It then calculates a modified instantaneous transmit rate *X_inst*, as follows:

```
X_inst = X * R_sqmean / sqrt(R_sample);
```

When *sqrt(R_sample)* is greater than *R_sqmean* then the queue is typically increasing and so the transmit rate needs to be decreased for stable operation.

Another important issue is the protocol's transmission rate. TFRC computes its maximum transmission rate as the number of packets per second that a TCP application would receive in similar conditions while breaking up its data into 1480-byte chunks. A TFRC application that is using large packets will experience roughly the same transmission rate in bits per second as a TCP application. However, a TFRC application using small packets will experience a lower transmission rate, in bits per second, than a TCP application. The reasoning for this is that bottlenecks can be the bits-per-second capacity of links, and also the packets-per-second capacity of routers.

In the subsequent sections of this paper, we present mechanisms based on TFRC that still remain TCP-friendly and utilize TFRC's feedback reports to improve multimedia transmission performance.

The TFRC protocol presents a modern approach to transport layer protocols, which treat protocols as a set of building blocks: independent components, from which transport protocols are assembled. TFRC provides a sending rate within a factor of two of the sending rate a TCP flow would have under the same condition but with relatively more stable throughput, which is a desirable characteristic for a streaming service. TFRC is a receiver-based mechanism where the receiver performs some calculation of the congestion control indicators and reports them back to the server. It relies on the underlying transport protocol such as the DCCP [11] to provide means for the exchange of control information between the server and the client.

The algorithm used to calculate the next sending rate depends on whether the sender is still in the initial Slow Start phase or in the Congestion Avoidance phase. In the Slow Start phase, the sender approximately tries to double its sending rate every time a Receiver Report is received in order to reach the maximum throughput the channel can support, which can be detected by increasing RTT and losses. Once the first loss has been detected, the sender enters the Congestion Avoidance phase. The next sending rate *X* is now determined from the minimum between twice the previous receiving rate and the sending rate as calculated from the TCP throughput equation.

```
X = min ( TCP throughput, 2*receiving rate )
```

For the purposes of the cross-layer mechanisms detailed in the later sections of this article, it is important to describe in some detail the mechanism and structure of the feedback packets specified by the TFRC protocol.

The receiver periodically sends feedback messages to the sender. Feedback packets should normally be sent at least once per RTT, unless the sender is sending at a rate of less than one packet per RTT, in which case a feedback packet should be sent for every data packet received. A feedback packet should also be sent whenever a new loss event is detected without waiting for the end of an RTT, and whenever an out-of-order data packet is received that removes a loss event from the history. If the sender is transmitting at a high rate (many packets per RTT) there may be some advantages in sending periodic feedback messages more than once per RTT as this allows faster response to changing RTT measurements, and more resilience to feedback packet loss. However, there is little gain from sending a large number of feedback messages per RTT.

DOI: 10.1002/nem

Each feedback packet sent by the data receiver contains the following information:

- The timestamp of the last data packet received. We denote this by *t_recvdata*. If the last packet received at the receiver has sequence number *i*, then *t_recvdata* = *ts_i*. This timestamp is used by the sender to estimate the RTT, and is only needed if the sender does not save timestamps of transmitted data packets.
- The amount of time elapsed between the receipt of the last data packet at the receiver, and the generation of this feedback report. We denote this by *t_delay*.
- The rate at which the receiver estimates that data was received since the last feedback report was sent. We denote this by *X_recv*.
- The receiver's current estimate of the loss event rate, *p*.

Obtaining an accurate and stable measurement of the loss event rate is of primary importance for TFRC. Loss rate measurement is performed at the receiver, based on the detection of lost or marked packets from the sequence numbers of arriving packets.

1.3. Power adaptation over TFRC

The target of this paper is to describe power adaptation mechanisms, in wireless scenarios, in order to minimize or eliminate packet losses, since even a small packet loss rate can result in an important reduction of multimedia quality for the end-user and result in a bad end-user experience. In particular, the paper proposes mechanisms for cross-layer power management for video transmission over wireless 802.11 networks using the TFRC protocol. The rest of this paper is organized as follows. Section 2 gives an overview of related work in the area of cross-layer optimization. Section 3 describes the main idea of the paper and Section 4 gives the test bed setup. Experiments and their results are presented in Section 5, while Section 6 concludes the paper and discusses possible future work. Source code for our implementation and installation instructions can be found in CTI Research Unit 6 [12].

2. RELATED WORK

Many cross-layer design proposals can be found in the literature (Figure 1). It is worthwhile to present how the layers are coupled; in other words, what kind of architecture change has taken place in a

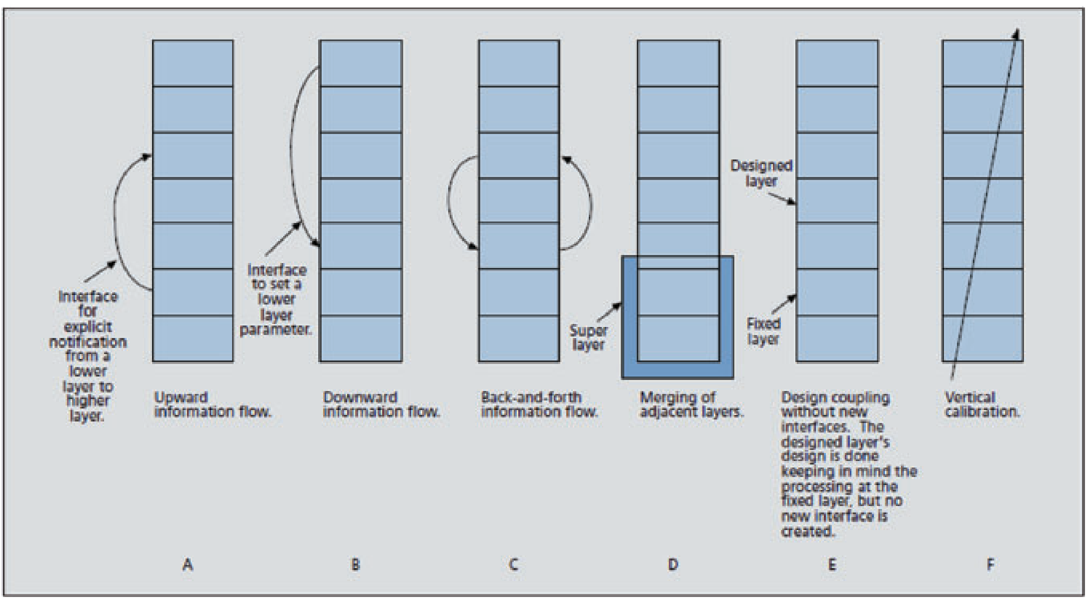


Figure 1. Illustrating the different kinds of cross-layer design proposals. The rectangular boxes represent the protocol layers [13].

DOI: 10.1002/nem

particular cross-layer design. We note that the layered architecture can be bypassed in the following basic ways according to Srivastava *et al*. [13]:

- *Creation of new interfaces*. Several cross-layer designs require the creation of new interfaces between layers. The new interfaces are used for information sharing between the layers at run time.
- *Merging of adjacent layers*. Another way to do cross-layer design is to design two or more adjacent layers together such that the service provided by the new superlayer is the union of the services provided by the constituent layers.
- *Design coupling without new interfaces*. Another category of cross-layer design involves coupling two or more layers at design time without creating any extra interfaces for information sharing at run time.
- *Vertical calibration across layers*. The final category in which cross-layer design proposals in the literature fit is what we call vertical calibration across layers. As the name suggests, this refers to adjusting parameters that span across layers.

The cross-layer design approach in this paper is categorized in the 'Creation of new interfaces' category for cross-layer proposals, which was introduced above. The reason that this cross-layer approach is used is that, as has been repeatedly argued, although layered architectures have served well for wired networks, they are not suitable for wireless networks. There are three main reasons for this: the unique problems created by wireless links, the possibility of opportunistic communication on wireless links, and the new modalities of communication offered by the wireless medium. Another example is, for instance, if the end-to-end TCP path contains a wireless link, errors on the wireless link can trick the TCP sender into making erroneous inferences about the congestion in the network, and as a result the performance deteriorates. Creating interfaces from the lower layers to the transport layer to enable explicit notifications alleviates such situations. For example, the explicit congestion notification (ECN) from the router to the transport layer at the TCP sender can explicitly tell the TCP sender if there is congestion in the network to enable it to differentiate between errors on the wireless link and network congestion [14].

Several researchers have focused on various issues of cross-layer optimization for wireless ad hoc networks, when there is no infrastructure assumed. Radunović [15] proposes a jointly optimal design of the three layers (physical, Medium Access Control (MAC), routing) for wireless ad hoc networks and studies several existing rate maximization performance metrics for wireless ad hoc networks in order to select appropriate performance metrics for the optimization. In [16] Yufeng and Zakhor propose an application adaptive scheme based on priority-based automatic repeat request (ARQ) together with a scheduling algorithm and forward error correction (FEC) coding combined with Radio Link Protocol (RLP) layer granularity. In van der Schaar and Sai Shankar [1] the need for cross-layer optimization is examined and an adaptation framework is proposed amongst the application (APP), MAC and physical (PHY) layers. In the same publication a number of different methodologies for cross-layer adaptation are proposed, named 'top-down', 'bottom-up', 'application centric' and 'MAC-centric' approaches.

In Yufeng and Zakhor [16] a joint cross-layer design for quality of service (QoS) content delivery is presented. The central concept of the proposed design is that of adaptation. A new QoS awareness scheduler with a power adaptation scheme is proposed and is applied at both uplink and downlink MAC layers to coordinate the behavior of the lower layers for resource efficiency. Lin *et al*. [17] summarize the recent developments in optimization-based approaches for resource allocation problems in wireless networks using a cross-layer approach. Li *et al*. [18] deal with 802.16 WiMAX (Worldwide Interoperability for Microwave Access) networks. The 802.16 standard provides four kinds of multimedia data services with QoS parameters but does not define any QoS scheduling algorithm. This paper presents an adaptive cross-layer scheduling algorithm for the IEEE 802.16 BWA (Broadband Wireless Access) system. The algorithm uses the adaptive modulation and coding (AMC) scheme at the physical layer according to the SNR on wireless fading channels. In addition, the cost function is defined for each kind of multimedia connection based on its service status, throughout the deadline in the MAC layer. The simulation results provided show that the scheduling algorithm achieved an optimum trade-off between throughput and fairness. In Warrier *et al*. [19], the

DOI: 10.1002/nem

gap between existing theoretical cross-layer optimization designs and practical approaches is examined.

Power management in wireless networks is surveyed in Klues [20] and techniques are classified according to the layer where they are applied (application, transport, network, data link, MAC or physical). Kim and Kim [21] propose a power management scheme for intra-frame-refreshed image sequences of the wireless video service in code division multiple-access (CDMA) systems, while Zamora *et al*. [22] introduces coordinated power management policies for video sensor networks. In Li *et al*. [23], transmission power is one of the parameters that were jointly optimized in order to minimize power consumption. A thorough survey of power awareness in mobile multimedia transmissions can be found in Zhang *et al*. [24]. It mainly focuses on the trade-off between power consumption for computation and communication usage. To the best of our knowledge, the cross-layer design presented in this paper is the first one taking into consideration parameters such as receivers' perceived video quality, while using TFRC in wireless video transmission.

3. POWER MANAGEMENT MECHANISMS

The target of the two mechanisms presented in this section is to minimize or eliminate packet losses, since even a small packet loss rate can result in important reduction of multimedia quality for the end-user and result in a bad end-user experience. Combined with TFRC's limited variation in transmission rates, we aim for improved media parameters such as peak signal-to-noise ratio (PSNR) and mean opinion score (MOS), which better represent the end-user experience. At the same time, we have to make sure that power consumption will be bounded and will only increase when this results in noticeably improved video quality. As shown in Figure 2, both mechanisms use the same cross-layer design. A new interface has been provided to TFRC in order to set the power transmission accordingly.

3.1. The MIMD mechanism

The first mechanism we propose is the multiplicative increase, multiplicative decrease (MIMD) mechanism [25]. Due to TFRC operation, the receiver sends periodic reports to its sender, where information such as the number of packet losses is included. The MIMD mechanism uses the TFRC

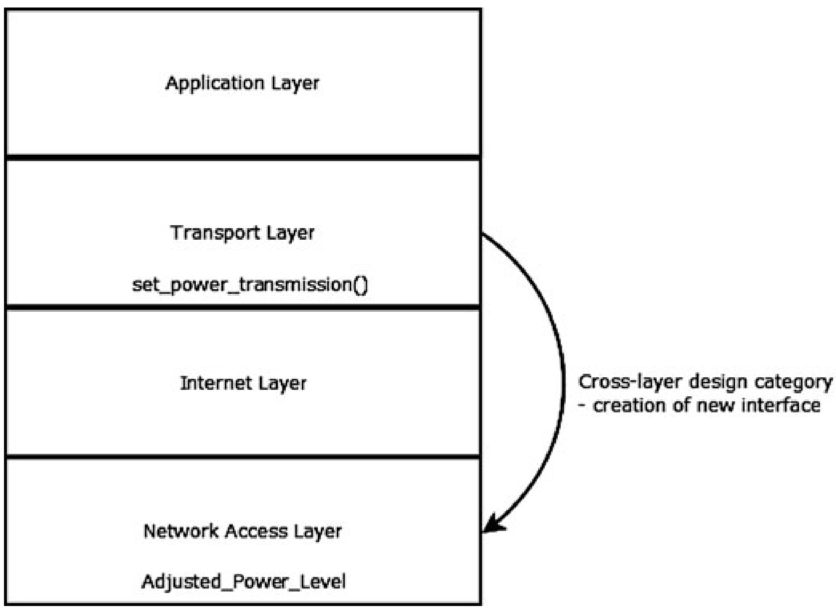


Figure 2. Proposed cross-layer design.

receiver's reports to the sender in order to calculate the packet loss rate percentage. The algorithm considers only a constant number of previous packet losses, so that it is more adaptive to the most recent conditions of the network and does not take into account the overall number of previous packet losses by the specific time or just the last packet loss in order to make sure that the power transmission will be increased only if the channel conditions continue to affect the transmission. In addition, if the packet loss rate increases above a preset threshold, then the power is also increased by a percentage; otherwise, if the packet loss falls way below the threshold, the power consumed is decreased for reasons of power efficiency. Moreover, the power consumed has a lower bound to prevent the base station from halting the transmission and an upper bound to prevent excessive consumption. This cross-layer mechanism uses information provided by the TFRC protocol, which is a transport layer protocol and needs to act upon the physical layer to adjust the transmission power. The parameters involved at each layer include the transmission power at the physical layer, and the packet loss information at the transport layer. The interaction of these parameters is explained in the pseudocode below. We should note here that the selection of both multiplicative increase and decrease has been made in order for power to be adjusted quickly and equally in both the increasing and the decreasing direction. Although it is well known that MIMD is not efficient for the purposes of controlling window size [26], our concern for the level of transmission power is not the interaction with other instances of the algorithm. Rather, the resource whose usage we are trying to manage more effectively is the available power of the transmitting node, and it is expected that a slower-increasing strategy (like Additive Increase Multiplicative Decrease [AIMD]) would negatively affect received video quality.

In the pseudocode below, *PL* stands for packet losses (as a percentage) and *TP* for transmission power, while $A > 1$ and $0 < B < 1$):

```
while (true) {
   retrieve last TFRC report
   set PL = Average of last N reports
   if (PL > Threshold_1) and (TP < Upper_Bound)
   then set TP = A * TP
   else if (PL < Threshold_2) and (TP > Lower_Bound)
then set TP = B * TP
}
```

After extensive experimentation (using ns2 simulations as detailed in the following evaluation sections with node movement speeds of up to 6 m/s) with the values *A*, *B* and the thresholds we concluded with the values $A = 1.05$, $B = 0.05$, *Threshold_1* = 0.1, *Threshold_2* = 0.075, which led to both good PSNR values and limited energy consumption. These values imply that increases take place more cautiously than decreases in power levels. For example, a doubling of the level of power will take around 15 consecutive feedback reports with low packet losses. Parameters *A* and *B* can therefore be considered as the deciding factors for the speed of power adjustment relative to packet losses feedback. It therefore follows that the above-mentioned values are efficient for the specific mobility speeds that are discussed in the experiments presented in this paper. In settings where mobile nodes are expected to move much faster, *A* and *B* values should be adjusted accordingly, and the opposite applies in settings where mobile nodes move infrequently. The values for *Upper_Bound* and *Lower_Bound* are discussed in the Experiments section.

The target of the proposed mechanism is to minimize or eliminate packet losses, since even a small packet loss rate can result in an important reduction of multimedia quality for the end-user and result in a bad end-user experience. Improvements in the above two areas will lead to improved media parameters such as PSNR and MOS, which better represent the end-user experience. At the same time, it has to make sure that power consumption will be bounded and will only increase when this results in noticeably improved video quality.

3.1. The binary mechanism

We call the second mechanism that we propose the binary mechanism. Similar to the MIMD approach presented above, the binary mechanism uses the TFRC receiver's reports to the sender in order to

DOI: 10.1002/nem

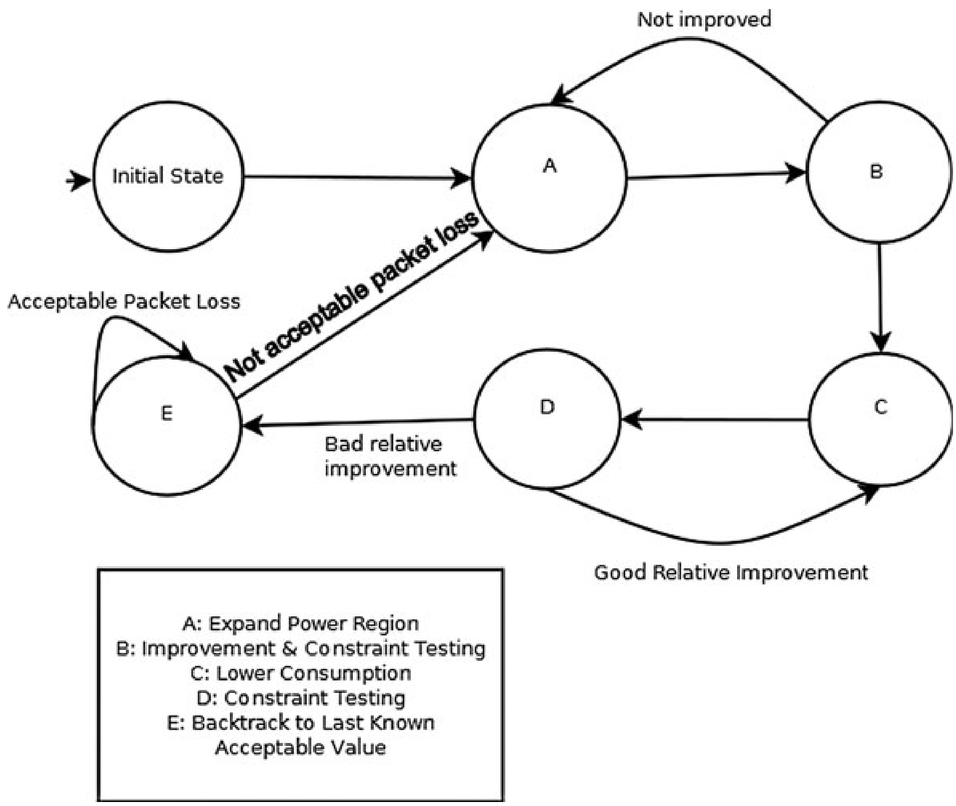


Figure 3. Finite state machine for the proposed mechanism for the sender.

calculate the packet loss rate percentage. The algorithm considers only a constant number of previous packet losses, so that it is more adaptive to the most recent conditions of the network. This cross-layer mechanism uses information provided by the TFRC protocol, which is a transport layer protocol and needs to act upon the physical layer to adjust the transmission power. The parameters involved at each layer include the transmission power at the physical layer and the packet loss information at the transport layer.

The essence of the mechanism is to provide a better and quicker convergence to efficient power levels in scenarios where the mobile nodes follow more random movement patterns (i.e., we do not expect to have many consecutive feedback reports that would push the MIMD mechanism in the same direction). We have to note that the objective of the binary mechanism is therefore to accommodate rapidly changing movement patterns, but not necessarily fast-moving nodes, which may be satisfied by suitable parameterization of the MIMD mechanism as described in the previous section. The 'binary' name comes from the dichotomic ('divide and conquer') nature of the mechanism, since it tries to divide the possible power-level ranges through their center, as will be shown below.

The finite automaton presented in Figure 3 is the mechanism used by the sender of the video via TFRC. Every time the sender receives a TFRC report from the receiver, it changes its state according to the state it is in and the new data. The mechanism after receiving the first report, if packet loss is not satisfactory, defines a region in which it will try to approximate the optimum power. The optimum power is that which produces a desired value of packet loss. After defining the region, the sender will increase its power to the maximum possible in that region and send the next TFRC packet with that power (state A). When the sender receives the next report, it tests whether there has been as significant improvement. If there has been an improvement and packet loss is below a predetermined threshold, the sender goes to state C or else repeats the actions of state A. In state C, the mechanism sets the power to the middle of the defined region and the sender goes to state D. In state D the algorithm tests whether the packet loss constraints are still satisfied and, if this is the case, it repeats state C. If this is not the

DOI: 10.1002/nem

case the algorithm goes to state E, where it goes back to the previous known acceptable power value. The mechanism stays at state E while the packet loss value is acceptable and, if not, it goes back to state A. Below is a summary of the states of the automaton:

```
INIT: initializations
A: Expand ″power region″ and apply region-maximum power, then goes to
state B
B: Improvement and constraint testing. If qualified, goes to state C,
else it goes to state A
C: Lowers consumption to the middle of the defined power region and
goes to state D
D: If all the constraints are satisfied, goes to state C, else goes to
state E
E: Backtracks to the last known acceptable power value and stays
there while packet loss is acceptable, else it goes to state A.
```

4. TEST BED SETUP

For our experiments we have used the Network Simulator 2 (ns-2.30) [27] as a basic tool for simulating multimedia data transmission over wireless networks.

In order to simulate MPEG-4 video transmission using ns-2, another software package is needed, namely Evalvid-RA [28,29]. Evalvid-RA supports rate-adaptive multimedia transfer based on trace file generation of an MPEG video file. A typical trace file provides information for frame number, frame type, size, fragmentation into segments and timing for each video frame. The multimedia transfer is simulated by using the generated trace file and not the actual binary multimedia content. The simulator keeps its own trace files, holding information on timing and throughput of packets at each node during simulation. Combining this information and the original video file Evalvid-RA can rebuild the video file as it would have been received on a real network. Additionally, by using the Evalvid-RA toolset the total noise introduced can be measured (in dB PSNR) and MOS can be calculated. An example implementation is illustrated in Haukass [30].

Several modifications of the network simulators were needed in order to build a working instance of the proposed mechanism. Firstly, a module that implements the logic of the proposed mechanism was added in the simulator. Then, the module that implements the TFRC protocol was changed so that it provides information about packet losses to our mechanism. The mechanism calculates the power needed to improve PSNR and this information is then passed to the modified wireless physical layer module, which is able to increase or decrease power according to the mechanism.

In our experiments we used the network topology illustrated in Figure 4. The akiyo sample video found in Xiph.org Test Media [31] was used for video streaming for the purposes of our experiments.

The simulation environment consists of three parts and is depicted in Figure 4. During the pre-processing phase a raw video file, which is usually stored in YUV format, is encoded with the desired video encoder into 30 different encoded MPEG-4 video clips with quantizer scale values in the range 2–31. Quantizer scale 2 provides an encoded video with the highest quality. We use the ffmpeg [32] free video encoder for the creation of the video clips. For our simulations, all video clips have a

Figure 4. Topology in experiments.

DOI: 10.1002/nem

temporal resolution of 25 frames per second and GoP (Group of Pictures) pattern IBPBPBPBPBPB, with a size of 12 frames. The frame size of all clips is 352×288 pixels, which is known as the Common Intermediate Format (CIF). After all the video files are encoded they are then traced to produce 30 frame-size trace files. At the end of the pre-processing phase we thus have 30 m4v files with their associated frame size files.

Briefly, the video file was pre-processed and many video files were produced of different quality and resolution using the ffmpeg tool [29] and shell scripts included in the Evalvid-RA toolset. Trace files were then generated for all these files and by using these trace files the simulation took place. Ns-2 scripts were created to simulate video transmission over a wireless network over TFRC. After simulating the transfer of the video in several different resolutions, ns-2 trace files were obtained which were then used to reconstruct the video as it would have been sent over a real network.

The third part of the simulation environment consists of reconstruction of the transmitted video and measurement of the performance evaluation metrics. The reconstruction of the received video traces is implemented offline by comparing the transmitted and received traces with those of the original video sequence of all the transmitted simulcast streams. In this phase, several measurements and calculations can be done involving network and video metrics such as PSNR, MOS, jitter, throughput and delay. With the above-described procedure we are able to make extensive comparisons between algorithms and reach conclusions about the efficiency of each one.

5. PERFORMANCE EVALUATION EXPERIMENTS

In our ns-2 experiments, we transfer H.264 video over TFRC over wireless links, and in particular over a single hop in a wireless ad hoc network. In order to model various instances of network degradation, we have performed a series of experiments with various scenarios, with both stationary and mobile nodes. We then compare the achieved throughput in terms of PSNR, packet losses and power consumption. Objective PSNR measurements can be approximately matched to subjective MOS according to the standardized Table 1. The MOS scores reported below are derived from the automatic PSNR to MOS mapping according to Table 1.

In the MIMD mechanism, *Lower_Bound* ranged from 0.02 to 0.04 and *Upper_Bound* from 0.06 to 0.1. In Experiments 1 and 2 we ran a set of experiments with different *Lower_Bound* and *Upper_Bound* each time in the above range and increasing by 0.01 in each experiment. The results in Table 2 are from the average of these experiments.

- *Scenario 1*. Two nodes: both stationary

- *Scenario 2*. Two nodes: one stationary, one moving away

- *Scenario 3*. Two nodes: one stationary, one moving closer and then moving away

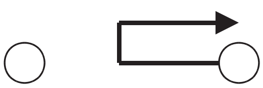


Table 1. ITU-R quality and impaired scale [10] and possible PSNR to MOS mapping [33].

PSNR [dB]	MOS	Impairment
>37	Excellent (5)	Imperceptible
31–37	Good (4)	Perceptible, but not annoying
25–31	Fair (3)	Slightly annoying
20–25	Poor (2)	Annoying
<20	Bad (1)	Very annoying

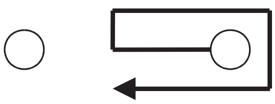
DOI: 10.1002/nem

Table 2. Scenario results.

Scenario	Normal	MIMD	Binary
	PSNR/power	PSNR/power	PSNR/power
1	669.2	813.1	790.1
2	666.4	769.4	782.5
3	662.2	759.8	798.8
4	676.2	798.9	814.8
5	671.8	800.3	789.7
6	666.4	769.4	782.5
7	669.2	813.1	790.1
8	919.3	902.3	968.4
Average	700.9	803.3	814.6
SD	88.66	45.06	63.02

- *Scenario 4*. Two nodes: one stationary, one moving closer

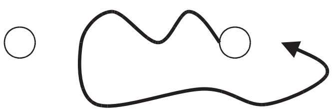
- *Scenario 5*. Two nodes, one stationary, one moving closer and then moving away and then moving closer again



- *Scenario 6*. Two nodes: one stationary, one moving away and then stops moving

- *Scenario 7*. Two nodes: one stationary, one moving closer and then stops moving

- *Scenario 8*. Two nodes: one stationary, one moving randomly



We repeat each scenario three times: one without any power management, one with the MIMD power management algorithm and one with the binary power management algorithm. Nodes moved with speeds up to 6 m/s in the applicable scenarios. We then compare the achieved throughput in terms of PSNR, packet losses and power consumption. Objective PSNR measurements can be approximately matched to subjective MOS according to the standardized Table 1. The MOS scores reported below are derived from the automatic PSNR to MOS mapping according to Table 1.

The MIMD method's performance varied according to the values of the thresholds chosen, while the binary method is insignificantly susceptible to threshold change. The binary method's performance, however, depends on the initial desired power that one wants to use.

5.1. Experiments

We ran the eight scenarios described above and took the ratio average PSNR over average power per experiment. For moving nodes the speeds ranged from 1 m/s to 6 m/s with steps of 1 m/s, and the values reported below correspond to the average of all experiments. The purpose is to maximize this ratio, as the larger its value the better the performance. Indeed, a large value means larger average PSNR or lower average power or both. The binary method clearly outperforms the MIMD method and the version without mechanism. In the summary Table 2 we have listed all scenarios and the aggregate score of each approach, where larger values indicate more favorable trade-off results.

DOI: 10.1002/nem

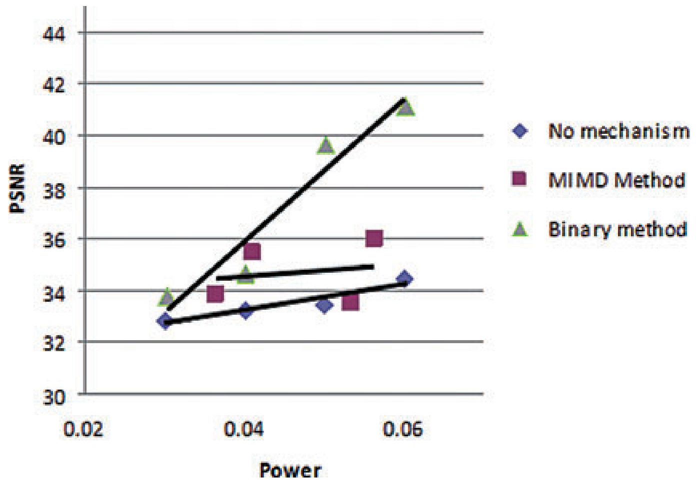


Figure 5. Scenario 1: two nodes, both stationary.

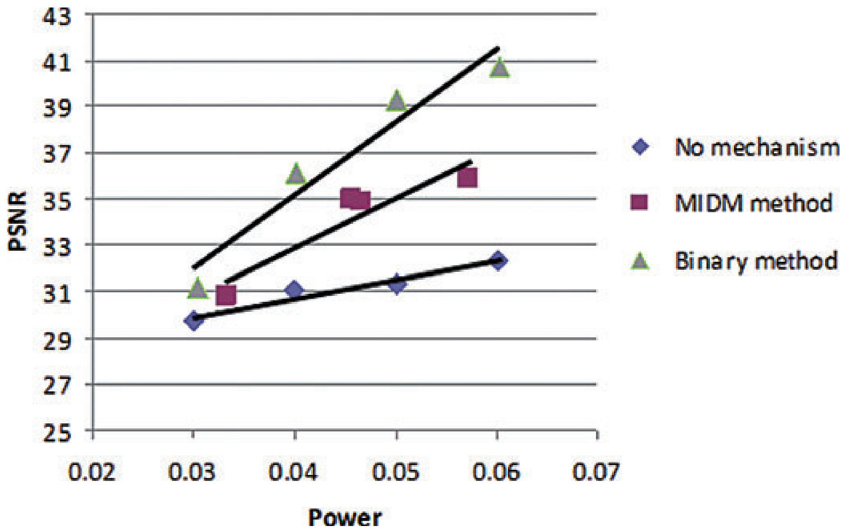


Figure 6. Scenario 2: two nodes, one stationary, one moving away.

We also present a detailed graph for each scenario, and provide trend lines in order to illuminate the behavior of each mechanism under different conditions. It is worthwhile noting that in many cases, as shown in Figures 5–12, the binary method achieves an Excellent Mean Opinion Score (see Table 1), whereas the other methods achieve at most a Good Mean Opinion Score.

In the first scenario both nodes are stationary, so power requirements do not vary. Nevertheless, power management mechanisms offer a better ratio of PSNR to transmission power. The proposed mechanism proves especially capable in taking advantage of the available transmission power. For a given amount of transmission power, the proposed mechanism significantly outperforms both the MIMD method and the original transmission approach, in terms of achieved video quality (as measured by the PSNR metric). For example, as shown in the graph, for an average available transmission power of about 0.03 all mechanisms achieve a PSNR value of about 34, which is considered good. When however, average available transmission power is doubled, the simple approach yields almost no benefit, the MIMD mechanism achieves a slightly higher PSNR value of 36, while the binary mechanism excels with a PSNR value of over 41, which qualifies as excellent.

The same observations apply also when one of the nodes is moving away. This time, the MIMD mechanism also displays a noticeable performance advantage over the simple approach. The

DOI: 10.1002/nem

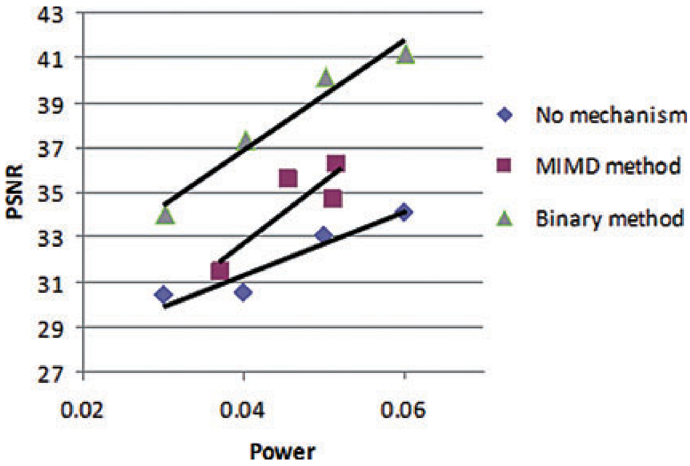


Figure 7. Scenario 3: two nodes, one stationary, one moving closer and then moving away.

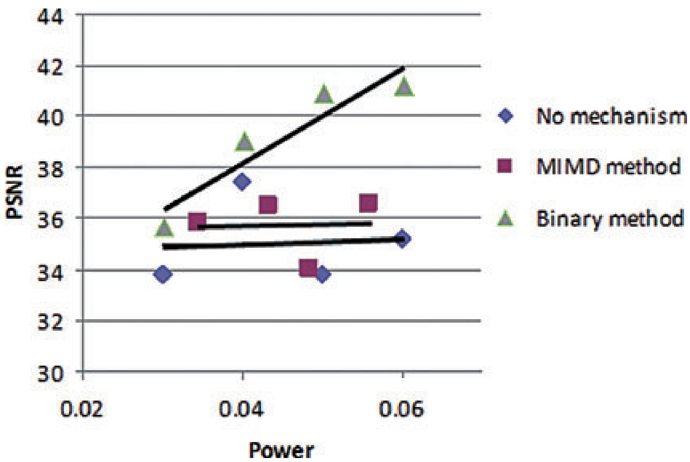


Figure 8. Scenario 4: two nodes, one stationary, one moving closer.

binary method converges faster and closer to an optimum value of power needed to decrease packet loss, and therefore achieves better PSNR values for the same average power. This scenario is one of the most beneficial for the proposed MIMD and binary mechanisms, because the movement is monotonous and they can easily find an excellent trade-off between energy consumption and video quality.

Because of the increased proximity of the nodes in Scenarios 3 and 4, the simple transmission approach is able to achieve better performance, without, however, being able to match either the MIMD or the binary power management approach because of their adjustment of power according to packet loss. It is interesting to note that in this scenario the binary mechanism clearly outperforms the other implementations even when average transmission power is low, since the variability in the movement of one of the nodes better suits a quickly adapting algorithm.

When a node is moving closer it is natural to achieve a better PSNR value in all methods. By also using rapid adjustment of power even better results occur. Also, the binary method achieves an Excellent Mean Opinion Score for power over 0.04. In this scenario, the performance gain of the MIMD mechanism over the simple one is reduced because, as the moving node increases its proximity, less transmission power is required and therefore a simple implementation can cope.

In scenarios with more complicated movement patterns, the basic conclusion seems to be the same: the proposed binary approach demonstrates a significant performance lead, which often results in the

DOI: 10.1002/nem

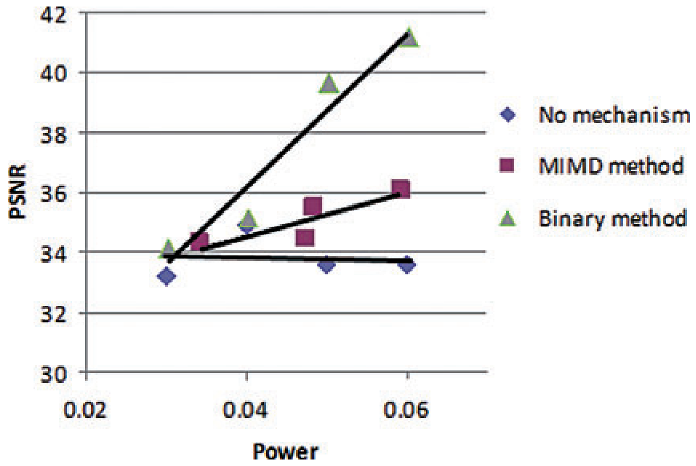


Figure 9. Scenario 5: two nodes, one stationary, one moving closer and then moving away and then moving closer again.

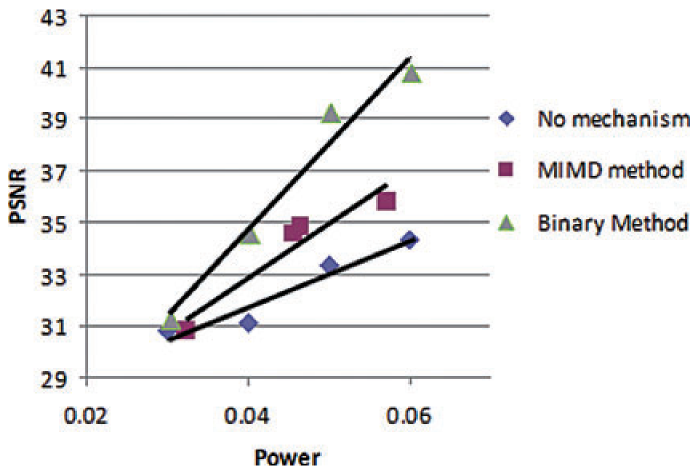


Figure 10. Scenario 6: two nodes, one stationary, one moving away and then stops moving.

received video quality to be excellent. The MIMD method provides intermediate benefits, while the original transmission approach without active power management lacks behind both in terms of video quality and consumed power.

In cases where the nodes stop after moving, the MIMD and binary methods adjust themselves to be as power saving as possible without making a reduction to the quality of video image transmitted. In fact, for the same PSNR values the MIMD and binary methods consume less energy than when using no mechanism and, when the binary method uses power 0.05 and over, achieves excellent results.

When one node moves randomly the results show that all mechanisms tend to display similar behavior for power values up to 0.04. Above this value the binary method again gains a significant advantage and achieves excellent results. The fact that in this scenario the performance gains are not as pronounced as in previous scenarios can be attributed to the fact that the adaptive methods need some time to adjust (they adjust every time they receive a TFRC report). Random motions tend to quickly change the assumptions upon which adaptive behavior is based. Therefore, the adaptive methods (MIMD, binary) tend to perform best in situations where there are movement patterns and changes in movement direction occur slower than the RTT of a TFRC report.

DOI: 10.1002/nem

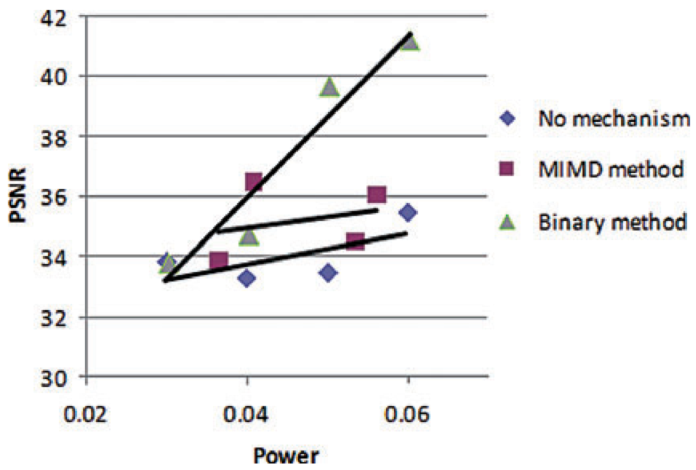


Figure 11. Scenario 7: two nodes, one stationary, one moving closer and then stops moving.

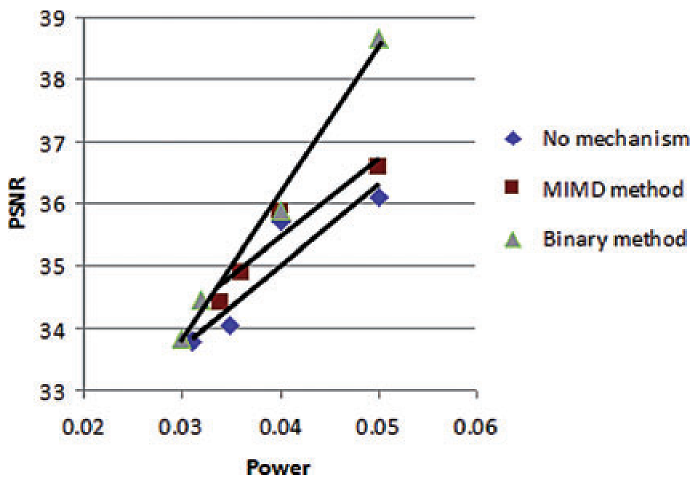


Figure 12. Scenario 8: two nodes, one stationary, one moving randomly.

6. CONCLUSIONS

In this paper we have proposed two variations of cross-layer mechanisms for power management in wireless TFRC transmission, which significantly improve both the objective quality of the transmitted video and make more optimal usage of available power compared to the traditional static approach. The MIMD power management approach has many performance benefits over the static approach, albeit significantly smaller than the binary method. Additionally, the MIMD mechanism is stateless and very easy to implement and therefore ideal for integration in low-capability systems. The binary method uses a more complicated algorithm which improves both the quality of transmitted video and power consumption significantly in comparison with the MIMD and no mechanism methods. The complexity cost of the binary mechanism is relatively small, as our implementation in the ns2 simulator has shown.

7. FUTURE WORK

The proposed cross-layer mechanism could be further improved in a wide range of ways. Firstly, the method to find the optimal power needed for transmission in order to minimize packet loss could be expanded to take into account the PSNR metric along with packet loss and adjust the transmission rate,

DOI: 10.1002/nem

power and video transmission quality in order to optimize the perceived video quality. An important aspect of the mechanisms is to examine what should be the most efficient approach in scaling them to multiple streams and multiple receivers, where various methods may be followed, depending on how the notion of fairness is handled in a wireless environment with multiple streams and whether average or worst-case performance is more important. Another aspect for future investigation is to compare the power consumption related to the algorithms' complexity and the resulting computational overhead. Furthermore, by using the capabilities of H.264 one can change video quality dynamically so that there can be adaptation of the transmission rate according to the available bandwidth. Moreover, the adaptation mechanism could become more elaborate and use machine learning techniques to adjust the transmission power.

REFERENCES

1. van der Schaar M, Sai Shankar N. Cross-layer wireless multimedia transmission: challenges, principles and new paradigms. *IEEE Wireless Communications* 2005; **12**(4): 50–58.
2. Tsaoussidis V, Badr H. TCP-probing: towards an error control schema with energy and throughput performance gains. In *8th IEEE Conference on Network Protocols*, Japan, November 2000.
3. Zhang C, Tsaoussidis V. TCP real: improving real-time capabilities of TCP over heterogeneous networks. In *11th IEEE/ACM NOSSDAV*, New York, 2001.
4. Jones CE, Sivalingam KM, Agrawal P, Chen JC. A survey of energy efficient network protocols for wireless networks. *Wireless Networks* 2001; **7**(4): 343–358.
5. IEEE 802.11 PSM Standard. *Power Management for Wireless Networks*. Section 11.11.2.
6. Chen H, Huang C-W. Power management modeling and optimal policy for IEEE 802.11 WLAN systems. In *IEEE Vehicular Technology Conference*, 2004.
7. Anastasi G, Conti M, Gregori E, Passarella A. A performance study of power-saving polices for Wi-Fi hotspots. *Computer Networks* 2004; **45**(3): 295–318.
8. Simunic T. Power saving techniques for wireless LANs. In *Conference on Design, Automation and Test in Europe*, Vol. 3, 2005; 96–97.
9. Handley M, Floyd S, Pahdye J, Widmer J. TCP friendly rate control (TFRC): protocol specification. *RFC 3448*, January 2003.
10. ITU-R Recommendations BT.500-11. Methodology for the subjective assessment of the quality of television pictures, 2002.
11. Kohler E, Handley M, Floyd S. Datagram Congestion Control Protocol (DCCP). *RFC 4340*, March 2006.
12. CTI Research Unit 6. work simulations. Available: http://ru6.cti.gr/ru6/research_ns.php [12 September 2010].
13. Srivastava M. Cross-layer design: a survey and the road ahead. *IEEE Communications Magazine* 2005; **43**(12): 112–119.
14. Shakkottai S, Rappaport TS, Karlsson PC. Cross-layer design for wireless networks. *IEEE Communications Magazine* 2003; **41**(10): 74–80.
15. Radunović B. A cross-layer design of wireless ad-hoc networks. PhD thesis, Ecole Polytechnique Federale de Lausanne (EPFL), July 2005.
16. Shan Y, Zakhor A. Cross layer techniques for adaptive video streaming over wireless networks. In *IEEE International Conference on Multimedia and Expo, 2002 (ICME '02)*, Vol. 1, 2002; 277–280.
17. Lin X, Shroff NB, Srikant R. A tutorial on cross-layer optimization in wireless networks. *IEEE Journal on Selected Areas in Communications* 2006; **24**(8): 1452–1463.
18. Li X, Wu X, Li W, Wang X. An adaptive cross-layer scheduling algorithm for multimedia networks. In *4th International Conference on Intelligent Information Hiding and Multimedia Signal Processing, IIH-MSP 2008*, art. no. 4604006; 52–55.
19. Warrier A, Le L, Rhee I. Cross-layer optimization made practical. In *4th International Conference on Broadband Communications, Networks, Systems, BroadNets*, art. no. 4550507; 733–742.
20. Klues K. Power management in wireless networks. Report, for *Advanced Topics in Networking: Wireless and Mobile Networking*, Jain R. Washington University in St Louis, MO, 2006.
21. Kim I-M, and Kim H-M. An optimum power management scheme for wireless video service in CDMA systems. *IEEE Transactions on Wireless Communications* 2003; **2**(1): 81–91.
22. Zamora NH, Kao J-C, Marculescu R. Distributed power-management techniques for wireless network video systems. In *Design, Automation and Test in Europe Conference and Exhibition*, 16–20 April 2007; 1–7.
23. Li Z, Zhai F, Katsaggelos A. Joint video summarization and transmission adaptation for energy-efficient wireless video streaming. *EURASIP Journal on Advances in Signal Processing* 2008; **2008**: 657032.
24. Zhang J, Wu D, Ci S, Wang H, Katsaggelos A. Power-aware mobile multimedia: a survey. *Journal of Communications* 2009; **4**(9): 600–613.
25. Bouras C, Gkamas A, Kapoulas V, Papapanagiotou V, Stamos K, Zaoudis G. Video transmission over TFRC using cross-layer power management. In *17th International Conference on Software, Telecommunications and Computer Networks (SoftCOM 2009)*, Split–Hvar–Korcula, 24–26 September 2009.
26. Chiu D, Jain R. Analysis of the increase/decrease algorithms for congestion avoidance in computer networks. *Journal of Computer Networks* 1989; **17**(1): 1–14.

DOI: 10.1002/nem

27. Network Simulator ns-2. Available: http://www.isi.edu/nsnam/ns/ [12 September 2010].
28. Lie A, Klaue J. Evalvid-RA: trace driven simulation of rate adaptive MPEG-4 VBR video. *Multimedia Systems* 2008; **14**(1): 33–50.
29. EvalVid-RA. Available: http://www.item.ntnu.no/~arnelie/Evalvid-RA.htm [12 September 2010].
30. Haukass T. Rate adaptive video streaming over wireless networks explained and explored. MSc thesis, Department of Telematics, Norwegian University of Science and Technology, Trondheim, Norway.
31. Xiph.org Test Media. akiyo. Available: http://media.xiph.org/video/derf/ [12 September 2010].
32. FFmpeg. Available: http://www.ffmpeg.org/ [12 September 2010].
33. Klaue J, Rathke B, Wolisz A. EvalVid: a framework for video transmission and quality evaluation. In *Proceedings of the 13th International Conference on Modeling, Techniques and Tools for Computer Performance Evaluation*, Urbana, IL, 2003.

AUTHORS' BIOGRAPHIES

Christos Bouras obtained his Diploma and PhD from the Computer Science and Engineering Department of Patras University (Greece). He is currently a Professor in the above department. He is also a scientific advisor of Research Unit 6 in Research Academic Computer Technology Institute (CTI), Patras, Greece. His research interests include Analysis of Performance of Networking and Computer Systems, Computer Networks and Protocols, Telematics and New Services, QoS and Pricing for Networks and Services, e – learning, Networked Virtual Environments and WWW Issues. He has extended professional experience in Design and Analysis of Networks, Protocols, Telematics and New Services. He has published 300 papers in various well-known refereed conferences and journals. He is a co-author of 8 books in Greek. He has been a PC member and referee in various international journals and conferences.

Apostolos Gkamas obtained his Diploma, Master Degree and Ph.D from the Computer Engineering and Informatics Department of Patras University (Greece). He is currently Assistant Professor in Higher Eccelesiastic Academy Vellas of Ioannina, visiting lecturer in Hellenic Open University in the Informatics Curriculum and Visiting Assistant Professor in the Informatics Department of University of 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). He has participated in various R&D project (in both EU and national) such as IST, FP6, FP7, Intereg eLearning, PENED, EPEAEK, Information Society.

Vaggelis Kapoulas obtained his Computer Engineering diploma from the Computer Science and Engineering Department, School of Engineering, University of Patras, Greece. He obtained his Ph.D. from the same department. He is currently an R&D engineer, and Head of Research Unit 6: Networks Telematics and New Services, Research Academic Computer Technology Institute (RACTI), 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.

Vassilis Papapanagiotou entered the Computer Engineering and Informatics Department in 2003 and graduated in 2008, entering the graduate program of the same department 'Computer Science and Technology'. He was a member of Research Unit 6 from October 2006 to December 2009. His research interests include cross-layer architectures for wireless networks, network protocols, Layer 2 and Layer 3 Quality of Service, ad-hoc networks and network programming. He has extensive knowledge of C, C++, Java, PHP, Database Manipulation, Web Technologies, Wide Area Network management and Cisco devices' configuration. He has worked for the Greek Research Network (www.grnet.gr) developing modules for automatic management of QoS services and managing its networking equipment in the city of Patras. He has also participated in the GN3 Project for advancing the European GEANT3 network. Additionally, he is the co-author of 3 papers in international refereed conferences, 1 article in an international Journal and 2 chapters in technical books.

Kostas Stamos obtained his Diploma, Master Degree and Ph.D from the Computer Engineering and Informatics Department of Patras University (Greece). He has worked for the Networking Technologies Sector of CTI,

DOI: 10.1002/nem

Research Unit 6 of CTI and the Greek Research and Education Network (Grnet). He is also teaching at the Computer Engineering and Informatics Department at the University of Patras, and at the Technological Educational Institute of Patras. He has published 8 articles in Journals and 30 papers in well-known refereed conferences. He is also co-author of 2 technical books, several encyclopaedia articles and of a Global Grid Forum (GGF) standard document.

Giannis Zaoudis obtained his Diploma from the Computer Engineering and Informatics Department of Patras University (Greece). He is currently in the graduate program of the same department 'Computer Science and Technology'. He has worked for the Research Unit 6 of CTI and the Greek Research and Education Network (Grnet). He is also participating in the GN3 Project for advancing the European GEANT3 network. His research interests include cross-layer architectures for wireless networks, network protocols and ad-hoc networks. He has extensive knowledge of C, C++, Java, PHP, Database Manipulation, Web Technologies and Web development. He has published 1 article in Journal and 3 papers in well-known refereed conferences.

DOI: 10.1002/nem