A Framework for Cross Layer Adaptation for Multimedia Transmission over Wired and Wireless Networks

Christos Bouras
Research Academic Computer Technology Institute and Computer Engineering and Informatics Department, University of Patras
Rion, Greece

Apostolos Gkamas
Research Academic Computer Technology Institute
Rion, Greece

Georgios Kioumourtzis
Computer Engineering and Informatics Department,
University of Patras
Rion, Greece

Abstract - *This paper proposes a framework for cross-layer adaptation for multimedia transmission over wired and wireless networks. The proposed framework consists of four entities: The sender, the proxy, which is located at the edge of the wired network, the access point, which co-located with the proxy and finally the wired and wireless receivers. The main concept is based on a "holistic approach" in which all layers participate to the adaptation process and make its own contributions. In addition, the proposed framework can support both wired and wireless receivers in one platform and cross layer information are used in order to support wireless users whereas TCP friendly estimations are used in order to support wired users.*

Keywords: *Cross layer adaptation, Multimedia transmission, Wireless networks*

1 Introduction

Multimedia data transmission experience a number of constrains that result to low Quality of Service (QoS) that is offered to the end user. These constrains have mainly to do with the nature of multimedia applications, which are characterized by three main properties: the demand for high data transmission rate (bandwidth-consuming applications), the sensitiveness to packet delays (latency and jitter) and the tolerance to packet losses (packet-loss tolerant applications), when compared to other kind of applications. The above factors have led both the research community and the industry to develop and propose a number of new protocols and optimization techniques targeting at mitigating delay and packet loss ratio during the transmission of multimedia data. Most of these efforts are based on the classic layered approach in which the various layers try to optimize its performance by adapting its behavior to constantly varying network parameters and provide its best services to upper layers. This paper proposes a framework for cross-layer adaptation for multimedia transmission over wired and wireless networks. The main concept is based on a "holistic approach" in which all layers participate to the adaptation process and make its own contributions. In addition, the proposed framework can support both wired and wireless receivers in one platform in which cross layer information are used to support wireless users and TCP friendly estimations are used to support the wired ones. We strongly believe that any cross-layer adaptation effort is incomplete when only some of the layers take part in the adaptation process because all layers belong to the same system. However, further research will guide us to the most appropriate adaptation scheme as many factors, which are involved have to be investigated and evaluated through deeper studies and evaluation. The rest of this paper is organized as follows. Next section describes the related work. Our proposed framework is presented in section 3. Section 4 discusses the Algorithmic aspects. Conclusions and future work are discussed in section 5.

2 Related work

Over the last years a number of new protocols have been developed for multimedia applications in the whole OSI layer's scale. The family of MPEG protocols, which have been standardized, have been designed for encoding and compression of multimedia data. The MPEG-4 [1] protocol with the latest enhancement of the FGS (Fine Granularity Scalability) [2] provides adaptive video coding by taking into account the available bandwidth and is expected to be used by many multimedia applications. The RTP and RTCP protocols [17], which operate on the transport layer usually on top of the UDP protocol, have been especially designed for multimedia data transmission. Apart from the above developments in the protocol and the architectural fields there have been a number of proposals for improving QoS in multimedia applications through cross layer adaptation strategies. In [3] the need of a cross-layer optimization is examined and an adaptation framework is proposed amongst the application (APP), the Medium Access Control (MAC) and the Physical (PHY) layers. In the same publication a number of different methodologies for cross-layer adaptation are proposed, named "top-down" approach, "bottom-up", "application centric" and "MAC centric". In [4] the issue of cross-layer design in wireless networks is addressed. The focus is on the way that higher layers share knowledge of the PHY and MAC layers conditions in order to provide efficient methods to allocate network resources over the Internet. In [5] a joined APP and MAC adaptation is proposed with the use of MPEG-4 and the latest Fine Granularity Scalability (FGS) extension. Signalling issues between the layers for cross-layer optimization over wireless networks are examined in [6]. Although this proposal avoids heavy ICMP messages for out-bound signalling between the layers that is proposed in [7], it introduces very high complexity. In [8] a joined adaptation scheme of the APP, MAC and PHY layers is presented. Packet transmission is made by a novel-scheduling algorithm at the MAC layer whose function is based on the user and application priority levels. Finally, reference [9] outlines the need for new cross-layer architecture to address known problems of mobility, packet losses and delay that are observed in wireless networks. The main idea of a cross-layer manager is discussed in which all layers send notification messages to the manager who is responsible for intra layer co-ordination. This reference endorses our concept for a "holistic" participation of all layers in a cross-layer design. The subject of transmission of TCP friendly flows over networks has engaged researchers all over the world [21], [22], [23]. Various adaptation schemes deploy an analytical model of TCP [21] in order to estimate a TCP friendly bandwidth share. With the use of this model, the average bandwidth share (r_{tcp}) of a TCP connection is determined (in bytes/sec) with the following equation:

$$r_{tcp} = \frac{P}{t_{RTT}\sqrt{\frac{2Dl}{3}} + t_{out}\min(1, 3\sqrt{\frac{3Dl}{8}})l(1+32l^2)} \quad (1)$$

Where P is packet size in bytes, l is the packet loss rate, t_{out} is the TCP retransmission timeout, t_{RTT} is the Round Trip Time (RTT) of the TCP connection and D the number of acknowledged TCP packets by each acknowledgment packet. The proposed framework is using the above-described analytical model of TCP, in order to estimate TCP friendly bandwidth shares. For the following of this paper we assume that $D = 1$ (each acknowledgment packet acknowledges one TCP packet) and $t_{out} = 4t_{RTT}$ (the TCP retransmission timeout is set to be four time the RTT).

3 Proposed framework

3.1 General description

In this section we describe the proposed framework for cross-layer adaptation. Figure 1 presents the usage scenario of the proposed framework. The proposed framework consists of four entities: The sender which represents the multimedia server, the proxy which is located at the edge of the wired network, the AP which co-located with the proxy and can be integrated with the proxy and finally the wired and wireless receivers. This proposed framework separates the wired from the wireless part of the network by introducing a new entity named "proxy" between the sender and the receiver. The sender transmits multimedia data to the wired receivers and the proxy using the wired part of the network and the proxy is responsible to transmit the multimedia data received by the sender to the wireless receivers. The transmission in the wired part can benefit either by QoS features (IntServ / DiffServ [20]) of the network (if the network domains involved in the multimedia transmission offer QoS) or by congestion control mechanisms that are especially designed for multimedia data transmission in wired networks [16].

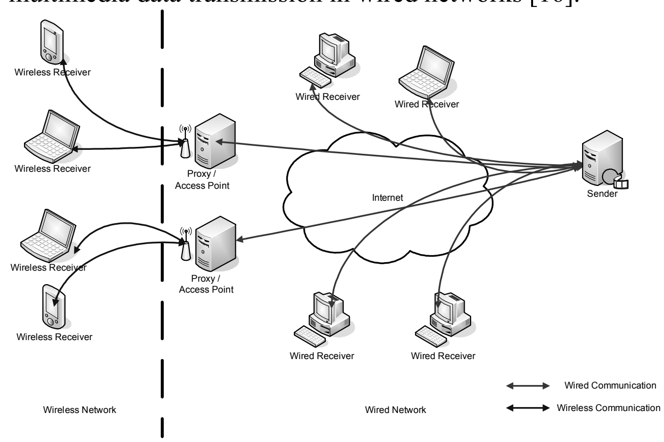

Fig. 1. Usage scenario of the proposed framework

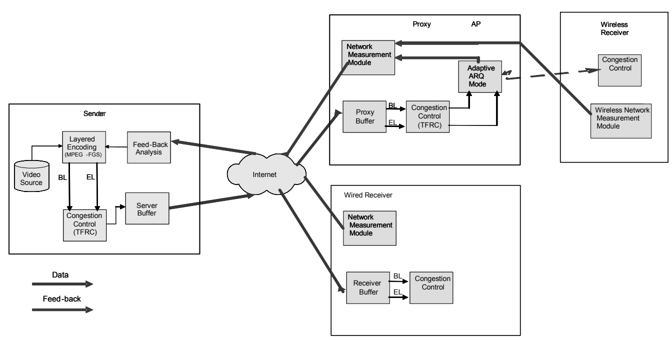

Fig. 2. Block diagram of the proposed framework

For the transmission of multimedia data in the wired network multicast is used. Proxy entity is responsible for collecting wireless users requirements and sends them to the sender in order to perform the cross layer adaptation. This approach is more efficient than any other mentioned in the literature in which cross layer adaptation is realized only in the wireless portion of the network for the following reasons: Firstly, the cause of packet losses is different in wired and wireless networks and therefore different error and congestion control mechanisms are needed accordingly. Secondly, in the wired portion we can apply existing adaptation and optimization mechanisms (extensively tested and evaluated during the last years), along with new protocols that have been especially designed for multimedia data transmission in wired networks. Thirdly, this architecture minimizes the distance in terms of the number of hops between the multimedia source and the wireless receiver. With the use of the proxy and depending of its buffer size the retransmission of lost RTP packets from the proxy instead of the sender can meet the time constrains of the application, as the distance between the proxy and the wireless receiver is only one hop. Figure 2 shows the block diagram of our proposed framework. In the next paragraphs we analyze each component of the proposed framework and present its functions.

3.2 Sender entity

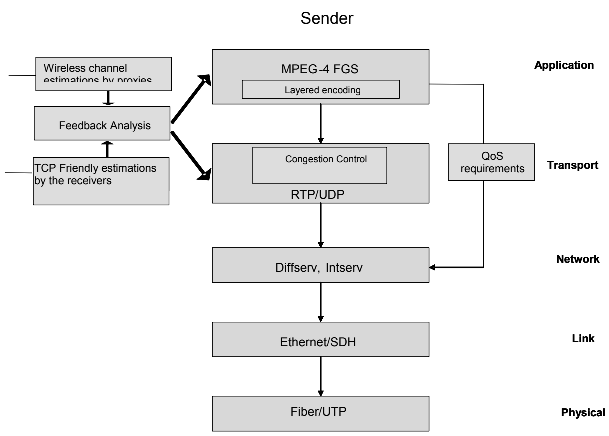

Fig. 3. Proposed Sender configuration

At the sender we distinguish four main functions: Feedback analysis, Layered encoding, Congestion control and taking advance of available (if they are available) network services (Diffserv or Inteserv). For layered encoding the MPEG-4 protocol is used. The proposed framework is using the MPEG-4 protocol with the latest FGS enhancement for video streaming between the sender, the receivers and the proxies for the following reasons: FGS encoding produces only two layers; the base layer (BL) in which the bit rate is equal to or lower to the minimum bandwidth and the enhancement layer (EL) that consumes the remaining available bandwidth. The sum of BL and EL bit rates must be equal or less to the available bandwidth. This scheme fits well in wireless networks due to bandwidth oscillations over time as we can transmit at least the BL when the available bandwidth reaches its lowest limit. With the use of RTCP adaptive feedback mechanism the wired receivers and proxies send their feedback to the sender in the form of RTCP receiver reports. We have added an application specific part (APP) to the RTCP receiver reports, which are sent by the wired receivers, in order to include the wired receivers' estimation about the TCP friendly bandwidth share $r^i_{r\_tcp}$, in the path between the wired receiver i and the sender. In addition, we have added an application specific part (APP) to the RTCP receiver reports, which are sent by the proxies, in order to include the proxies' j estimation about the available bandwidth r^j_w in the wireless network that are responsible for. The sender stores the last values of $r^i_{r\_tcp}$, and r^j_w from the all the wired receivers and proxies and this information is used for the adjustment of the transmission rate of the BL r_{BL} and EL r_{EL} according to the following procedure:

$$
\begin{gathered}
r_{BL} = \min(r^1_w,\ldots,r^j_w) \\
r_{EL} = \min(r^1_{r\_tcp},\ldots,r^i_{r\_tcp}) - r_{BL} \\
if\,(r_{EL} < 0) \\
\big(r_{BL} = r_{BL} + r_{EL} \\
r_{EL} = 0\big)
\end{gathered}
\qquad (2)
$$

With the above described procedure we set as transmission rate of the BL r_{BL} the minimum estimation of wireless bandwidth in the wireless part of the network and we set as transmission rate of EL r_{EL} the minimum value of TCP friendly estimation by wired receivers minus the transmission rate of the BL r_{BL}. In the rare event that the minimum value of TCP friendly estimation by the wired receivers is smaller than the minimum estimation of wireless bandwidth in the wireless part of the network, we set the transmission rate of the BL r_{BL} to the minimum value of TCP friendly estimation and we set the transmission rate of EL r_{EL} to zero. Sender entity executes the above-mentioned procedure when receives an RTCP receiver report either by a wired receiver or a proxy that contains the application specific part. With the above mention procedure the congestion control mechanism of the sender entity is implemented. Further simulation results and experiments will verify the validity and the performance of our proposal.

3.3 Wired receiver entity

Each wired receiver measures the characteristics of the path, which connects it with the sender and informs the sender with the use of RTCP receiver reports. The following parameters are measured: *Packet loss rate* (l_i)*:* The wired receiver calculates the packet loss rate of both layers (BL and EL) during the reception of packets with the use of RTP packets sequence numbers. *RTT estimations* (t^{e-i}_{RTT})*:* The wired receiver makes estimations for the RTT between it and the sender based on one-way delay measurements with the use of RTP packets timestamps. The wired receiver emulates the behaviour of a TCP agent with the use of the analytical model of TCP and estimates a TCP friendly bandwidth share $r^i_{r\_tcp}$ every RTT time using equation (1). If the wired receiver experiences packet losses it will use the following equation to estimate a TCP friendly bandwidth share (in bytes/sec):

$$r^{i}_{r\_tcp} = \frac{P}{t^{e-i}_{RTT}\sqrt{\frac{2l_i}{3}} + 4t^{e-i}_{RTT}\min(1, 3\sqrt{\frac{3l_i}{8}})l_i(1+32l_i^{\,2})} \quad (3)$$

If the wired receiver does not experience packet losses, in order to estimate a TCP friendly bandwidth share $r^{i}_{r\_tcp}$, the $r^{i}_{r\_tcp}$ must not be increased more than a packet / RTT. For this reason the wired receiver calculates the value of $r^{i}_{r\_tcp}$ with the following equation (in bytes/sec):

$$r^{i}_{r\_tcp} = r^{i}_{r\_tcp} + \frac{1}{t^{e-i}_{RTT}} P \quad (4)$$

Each time the wired receiver sends a receiver report to the sender it includes the average value of $r^{i}_{r\_tcp}$ since last receiver report.

3.4 Proxy and wireless receiver entities

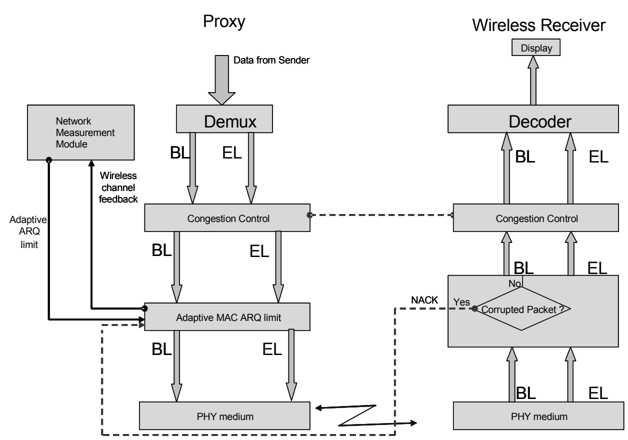

Fig. 4. Proxy and Receiver Block Diagram

Cross-layer adaptation in the wireless part of the proposed framework is more challenging than in the wired portion. The lack of standardized protocols that support QoS for multimedia transmission along with the unpredictable nature of the wireless medium offer an open area for extensive research. TCP or TCP like congestion control mechanism for throughput calculation it is a good candidate mechanism for multimedia congestion control transmission. However, recent research has shown that it does not perform well in wireless network [24]. The introduction of the proxy adds a level of abstraction to help facing in a more efficient and practical way the problems mentioned above. The proxy handles two important issues. The first one is the varying network conditions in the links between proxy- wireless receiver and proxy-sender. The second one is the classification between multimedia objects and other data objects (e.g. web pages, e-mails etc) in such way that the multimedia objects will be favoured by the proxy. Proxy and wireless receiver entities are designed "jointly" to clearer demonstrate the interactions between them. The proxy entity is based on the following modules: Adaptive congestion control, joint MAC-APP ARQ error control for the BL and the EL and finally the network measurement module. Figure 4 depicts the block diagram of the proxy and the wireless receiver configuration. The proxy receives the multimedia data (voice-video) from the sender and stores the BL and the EL into two different files. At this point the proxy has the ability either to transmit to wireless receiver only the BL, or to transmit the BL and the EL that require different transmission rates based on wireless link bandwidth estimation. More particularly the following procedure is used:

$$\begin{aligned} &if\,(r^{i}_{w} > r_{BL}) \rightarrow transmit - BL - and - EL \\ &if\,(r^{i}_{w} \leq r_{BL}) \rightarrow transmit - BL \end{aligned} \quad (5)$$

r^{i}_{w} is the proxy estimation for the available bandwidth of the wireless network. The joint APP-MAC error control defines the allowed number of retransmissions (Automatic Retransmission reQuest, ARQ) of a lost MAC frame. The ARQ retransmission limit is defined by the current channel conditions and the time constrains of the application. Finally, we have to mention that the proxy acts also as wired receiver and it performs all the actions described in the wired receiver section.

3.5 Signaling/Extensions to RTP/RTCP

The network measurement modules both at the proxies and wired receivers are responsible for the exploitation of the output of the above-described algorithms and also for forwarding the bandwidth information to the sender. This information is used for the MPEG-4 FGS coder in order to adjust its coding rate. Our main argument is to take advantage of the existing protocols and extend its functionalities. The operation of the proposed framework is based on the transmission of feedback packets with the use of RTP / RTCP reports. RTP provides an extension mechanism to allow individual implementations that require additional information to be carried in the RTP data packet header. The proposed framework uses the extension mechanism of RTP in order to add the following fields into the RTP header: t^{r-i}_{RTT} and wired receiver id: With this field the sender informs the wired receiver i about the effective RTT measurement between this receiver and the sender. In addition, RTCP gives the capability to the participants to include in the RTCP reports an application specific part (APP) intended for experimental use. The wired receivers add to their receiver reports an application specific part, which contains the average value of their estimations for TCP friendly bandwidth share $r^i_{r\_tcp}$, since the last receiver report. The proxies add to their receiver reports an application specific part, which contains the average value of their estimations for wireless network bandwidth r^i_w, since the last receiver report. Due to the fact that all the participants listen at least to the RTP / RTCP session of the BL layer, the above described extensions to RTP / RTCP are used only in the BL RTP / RTCP session.

4 Algorithmic aspects

In this section we define the algorithm for the estimation of the available bandwidth in the wireless part of the network. In addition we present the algorithms for packet loss estimation and RTT estimations. We then explain how joined APP-MAC adaptive error control can be realized.

4.1 Wireless link bandwidth estimation

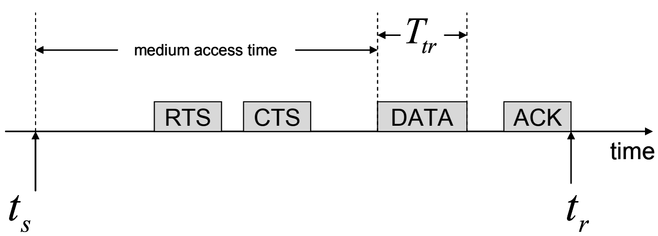

Fig. 5. IEEE 802.11 MAC Frame sequence transmission

The estimation of the available bandwidth has been extensively researched in wired networks and the main effort has been to discover the bottleneck in the path between the sender/receiver pair. In our proposed framework however, we need the available bandwidth of the wireless link between the proxy and the receiver. This wireless link is the last hop between the sender and the receiver and related algorithms for wired networks do not apply. Here the available bandwidth is the maximum achievable rate between proxy and receivers in the absence of any competing traffic. This maximum achievable rate is lower than the nominal channel transmission rate due to the presence of control packet (e.g. RTS/CTS mechanism), retransmitted MAC frames, etc. The nominal channel transmission rate has a predefined value depending on the underlying PHY MAC 802.11 protocol. Our proposed algorithm works as follows: The sending MAC frame is stamped with a time-stamp (t_s) indicating the sending time at the MAC layer. After the reception of the ACK sent by the receiver the algorithm calculates the interval between the sending time and the time when the ACK is received (t_r). In predefined periods ($t_{threshold}$), we calculate the available bandwidth as a fraction of the total bytes sent via the wireless link over the sampling time. Figure 5 shows the packet transmission sequence in an IEEE 802.11 wireless link. The available bandwidth is derived from the equation:

$$BW = \frac{S}{t_{sampling}} \quad (6)$$

Where S is the total size in bytes of the MAC frames that were sent over the sampling period $t_{sampling}$. Figure 6 presents the used algorithm.

$send(MAC\_frame)$
$t_s \leftarrow t_{system}$
$S_{sent} \leftarrow S_{MAC\_frame}$
$t_{sample} \leftarrow t_{sample} + t_{elapsed}$
$if\,(ACK\_received)\ then$
  $t_r \leftarrow t_{system}$
  $if\,(t_{sample} \leq t_{threshold})\ then$
    $t_{elapsed} \leftarrow t_r - t_s$
    $S_{total} \leftarrow S_{total} + S_{MAC}$
    $BW \leftarrow \frac{S_{total}}{t_{sample}}$
  $else$
    $PRINT\ BW$
    $t_{sample} \leftarrow null$
    $t_{elapsed} \leftarrow t_r - t_s$
    $S_{total} \leftarrow S_{MAC}$
$else$
  $t_{elapsed} \leftarrow t_{system} - t_s$

Fig. 6. Wireless bandwidth estimation algorithm

4.2 Packet loss rate estimation in wired network

Each wired receiver measures the packet loss rate based on the RTP packet sequence numbers in both layers (BL and EL). In order to prevent a single spurious packet loss having an excessive effect on the packet loss estimation, wired receivers smooth the values of packet loss rate using the following filter, which computes the weighted average of the m most recent loss rate values $l^m_{i,l}$ (the following filter has been presented and evaluated in [23] and provides a good estimation of the packet loss rate):

$$l_{i,l} = \frac{\sum_{j=0}^{m-1} w_j l_{i,l}^{m-j}}{\sum_{i=0}^{m-i} w_i}$$ for wired receiver i and layer j (BL or layer EL). (7)

Where $l_{i,l}$ is the smooth value of packet loss rate for layer l (BL or EL). The weights w_i are chosen so that very recent packet loss rates receive the same high weights, while the weights gradually decrease to 0 for older packet loss rate values. We use m=8 and the following values for the weights w_i: {1,1,1,1,0.8,0.6,0.4,0.2}. The wired receiver estimates the packet loss rate l_i, for both layers (BL and EL) that the receiver receives, with the following equation:

$$l_i = \frac{l_{i,BL} * r_{BL} + l_{i,EL} * r_{EL}}{r_{BL} + r_{EL}} \quad (8)$$

4.3 RTT estimations

When a wired receiver i receives a RTP packet from a sender layer (BL or EL), it uses the following algorithm in order to estimate the RTT between the sender and the wired receiver. Assuming that the sender and the wired receiver have synchronized clocks, the wired receiver can use the timestamp of the RTP packet ($T_{timestamp}$) and the local time when it receives that packet ($T_{receiver}$) to estimate the one-way time (T_{oneway}):

$$T_{oneway} = T_{receiver} - T_{timestamp} \quad (9)$$

If the path between the sender and the wired receiver were symmetric and the path had the same delay in both directions, the RTT between the sender and the wired receiver would be twice the T_{oneway}:

$$t_{RTT} = 2T_{oneway} \quad (10)$$

Until now, we have made two assumptions: (1) the sender and the wired receiver have synchronized clocks (2) the path between the sender and the wired receiver is symmetric. The above assumptions are not true for the Internet. Therefore, the wired receivers have to take the above assumptions into account in order to perform accurate RTT estimations (t_{RTT}^{e-l}). For this reason, we use a parameter a and we can write the equation (9) as:

$$t_{RTT}^{e-l} = (1+a)T_{oneway} \quad (11)$$

The parameter a is used to smooth the estimation of the RTT due to the potential unsynchronized clocks and the asymmetry of the path between the sender and the wired receiver. Furthermore, to avoid solely phenomenon that will affect the RTT estimations, the wired receivers pass the t_{RTT}^{e-l} values through a filter similar to one that they use for filtering the values of the packet loss rate. To estimate the value of parameter a, the wired receivers need an effective estimation of RTT, which can be acquired, with the use of RTCP reports: The RTCP receiver report contains two additional fields; the t_{LSR} (the timestamp of the most recent RTCP sender report from the sender) and the t_{DLSR} (the delay between the reception of the last sender report and the transmission of the receiver report). As a result the sender can make an effective RTT measurement for the path between it and a wired receiver by using the following equation (where A is the time when the sender receives the receiver report from the given wired receiver):

$$t_{RTT}^{r-i} = A - t_{LSR} - t_{DLSR} \quad (12)$$

The sender estimates an effective RTT measurement for a wired receiver i every time it receives a receiver report from that wired receiver and includes this effective RTT measurement (with the id of the receiver) in the next RTP packet of BL streams. A wired receiver after receives an effective RTT measurement from the sender, it estimates an appropriate value for the parameter a by using the following equation:

$$a = \frac{t_{RTT}^{r-i}}{T_{oneway}} - 1 \quad 13)$$

The values of t_{RTT}^{e-l} give an estimation of the RTT based on measurement on each layer (BL and EL). The wired receiver is using for TCP friendly transmission rate estimation and the average value of t_{RTT}^{e-l} for BL and EL layers:

$$t_{RTT}^{e-i} = \frac{t_{RTT}^{e-BL} + t_{RTT}^{e-EL}}{2} \quad (14)$$

4.4 Adaptive error control mechanism

Automatic Repeat request (ARQ) error control in wireless networks has been proven to be an efficient mechanism to recover from channel fluctuations. However, a predefined ARQ limit in multimedia data transmission without taking into account the nature of the underlying data may further degrade QoS. There are cases in which an untimely retransmission of a lost data block does not improve QoS (e.g. real time video). Furthermore, an untimely retransmission of a lost multimedia data block may prevent newly data to be transmitted at the receiver side. Our proposed scheme is based on two principles: a) the data block that has to be retransmitted does not exceed the application time constrains and b) the retransmission ARQ limit must be adjusted to the "value" of the data block. Thus BL and EL data frames are treated differently by the ARQ error control mechanism. The above principles are presented in the algorithm of figure 7 where t_{dl}, is the display deadline of the frame, and $ARQ\_\lim$ is a predefined value, which is different for BL and EL frames. Current research [5] showed that the best performance could be obtained when the ARQ limit is set to 8. In our future work we will evaluate the proposed ARQ algorithm and the optimal value of the retransmission limit.

01 $if\,(NACK\_received)\ then$
02 $\quad set(t_{dl})$
03 $\quad get(RTT)$
04 $\quad if\,(RTT < t_{dl})\ then$
05 $\quad\quad if\,(ARQ\_\lim > 0)\ then$
06 $\quad\quad\quad send(MAC\_frame)$
07 $\quad\quad\quad ARQ\_\lim \leftarrow ARQ\_\lim\ - 1$
08 $\quad else(discard)$

Fig. 7. ARQ algorithm

5 Conclusion/future work

This paper proposes a framework for cross-layer adaptation for multimedia transmission over wired and wireless networks. The main concept is based on a "holistic approach" in which all layers participate to the adaptation process and make its own contributions. In addition, the proposed framework can support both wired and wireless receivers in one platform in which cross layer information are used to support wireless users while TCP friendly estimations are used to support wired users. Our future work includes the evaluation of the proposed framework through a number of simulations. Main target of the simulations will be the examination of the proposed framework behaviour to a heterogeneous group of wired and wireless receivers.

6 References

[1] MPEG - 4: ISO / IEC JTC1 / SC29 / WG11 N4668.

[2] W. Li, "Overview of Fine Granularity Scalability in MPEG-4 Video Standard," IEEE Trans on Circuits and Systems for Video Technology, Vol. 11, No. 3, March, 2001, pp.301--317.

[3] M. van der Schaar, D. Sai Shankar N, "Cross-layer wireless multimedia transmission: Challenges, principles and new paradigms, IEEE wireless Communications, August 2005

[4] Shakkottai, S. Rappaport, T.S. Karlsson, P.C. "Cross-layer design for wireless Networks", IEEE Communication magazine, Volume: 41, Issue: 10, Oct 2003.

[5] M. van der Schaar, S. Krishnamachari, S. Choi "Adaptive Cross-Layer Protection Strategies for Robust Scalable Video Transmission Over 802.11 WLANs", IEEE Journal on selected areas in communications, vol 21, no.10 December 2003.

[6] Qi Wang; Abu-Rgheff, M.A., "Cross-Layer Signalling for Next-Generation. Wireless Systems", IEEE WCNC 2003, vol. 2, Mar. 2003, pp. 1084–89.

[7] P. Sudame and B. R. Badrinath, "On providing support for protocol adaptation in mobile wireless networks," Mobile Networks and Applications, Vo. 6, No. 1,pp. 43-55, January-February 2001.

[8] Jie Chen, Tiejun Lv and Haitao Zheng. "Cross-layer design for QoS wireless communications", Proceedings of the 2004 International Symposium on Circuits and Systems, 2004. ISCAS '04, Volume: 2, On page(s): II- 217-20 Vol.2, May 2004.

[9] Carneiro, G. Ruela, J. Ricardo, M, "Cross-layer design in 4G wireless terminals", IEEE Wireless Communications, Volume: 11, Issue: 2, page(s): 7- 13, Apr 2004.

[10] S. Choudhury and Jerry D. Gibson, "Joint PHY/MAC Based Link Adaptation for Wireless LANs with Multipath Fading", IEEE Wireless Communications and Networking Conference, 2006, Vol2, On pages 757-762, Apr 2006.

[11] C. Verikouris, L.Alonso, T. Giamalis, "Cross-layer optimization for wireless systems: A European research key challenge", Global Communications Newsletter, July 2005.

[12] Jie Chen Lv, T. Haitao Zheng, "Joint cross-layer design for wireless QoS content delivery", IEEE International Conference on Communications, Volume: 7, page(s): 4243- 4247, June 2004.

[13] H. M. Radha, M. van der Schaar and Y. Chen, "The MPEG-4 Fine-Grained Scalable video coding method for multimedia streaming over IP", IEEE Transactions on Multimedia, Vol. 3, No.1, March 2001.

[14] T. Ahmed, A. Mehaoua, R. Boutba and Y. Iraqi, "Adaptive packet video streaming over IP networks: A cross-layer approach", IEEE Journal on selected areas in Communications, Vol. 23, No. 2, February 2005.

[15] T.Goff, J. Moronski, D. S. Phatak, and V. Gupta, "Freeze-TCP: A true end-to-end TCP enhancement mechanism for mobile environments," in Proc. 19th Annual Joint Conference of the IEEE Computer and Communications Societies (INFOCOM' 99), vol. 3, pp. 1537-1545, Tel Aviv, Israel, March 2000.

[16] RFC 3448, M. Handley, S. Floyd, J. Padhye, J. Widmer, "TCP Friendly Rate Control (TFRC)", Network Working Group, January 2003

[17] RFC 3550, RTP: A Transport Protocol for Real-Time Applications, H. Schulzrinne, S. Casner, R. Frederick, V. Jacobson, July 2003

[18] RFC 2326 Real Time Streaming Protocol (RTSP). H. Schulzrinne, A. Rao, R.Lanphier. April 1998.

[19] RFC 3261 SIP: Session Initiation Protocol. J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, E. Schooler. June 2002.

[20] RFC 2475, "An Architecture for Differentiated Services", S. Blake, D. Black, M. Carlson, E. Davies, Z. Wang, W. Weiss, December 1998

[21] J. Pandhye, J. Kurose, D. Towsley, R. Koodli, "A model based TCP-friendly rate control protocol", Proc. International Workshop on Network and Operating System Support for Digital Audio and Video (NOSSDAV), Basking Ridge, NJ, June 1999.

[22] D. Sisalem, A. Wolisz, "MLDA: A TCP-friendly congestion control framework for heterogeneous multicast environments", in Eighth International Workshop on Quality of Service (IWQoS 2000), Pittsburgh, PA, June 2000.

[23] L. Vicisiano, L. Rizzo, J. Crowcroft, "TCP - like congestion control for layered multicast data transfer", in IEEE INFOCOM, March 1998, pp. 996 - 1003.

[24] A. Alexiou, D Antonellis, C. Bouras, "Adaptive and reliable video transmission over UMTS for enhanced performance", International journal of Communication systems, Vol. 20, issue 1, pp. 65-81, 2007