Cross-Layer Architecture for Adaptive Video Multicast Streaming Over Multirate Wireless LANs

José Villalón, Pedro Cuenca, Luis Orozco-Barbosa, Yongho Seok, and Thierry Turletti

***Abstract*— Multicast video streaming over multirate wireless LANs imposes strong demands on video codecs and the underlying network. It is not sufficient that only the video codec or only the underlying protocols adapt to changes in the wireless link quality. Research efforts should be applied in both and in a synchronized way. Cross layer design is a new paradigm that addresses this challenge by optimizing communication network architectures across traditional layer boundaries. In this paper we present a cross-layer architecture for adaptive video multicast streaming over multirate wireless LANs where layer-specific information is passed in both directions, top-down and bottom-up. We jointly consider three layers of the protocol stack: the application, data link and physical layers. We analyze the performance of the proposed architecture and extensively evaluate it via simulations. Our results show that the real-time video quality of the overall system can be greatly improved by cross-layer signaling.**

***Index Terms*— Cross-layer, rate adaptation, video adaptation, wireless multicast transmission.**

I. Introduction

ONE of the major applications driving wireless LAN services is video streaming, which is based on the ability to simultaneously multicast the same video contents to a group of users, thus reducing the bandwidth requirement. However, multicast video streaming over wireless networks is a challenging task due to the characteristics of the video data and wireless channels. In wireless environments, the channel conditions change rapidly over time due to noise, interference, multipath and mobility of the mobile hosts.

IEEE 802.11 *wireless LANs* (WLANs) are one of the fastest growing network technologies in the wireless communications field. The IEEE 802.11a/b *Media Access Control* (MAC) protocol provides a multirate-capable physical-layer [1], [2]. Multicast video streaming over multirate wireless LANs imposes strong demands on video codecs and the underlying network. With respect to the underlying protocols, since the multirate enhancements are implemented into the physical (PHY) layer, the MAC mechanisms should be adapted in order to fully exploit them.

There are two main issues to be addressed to effectively deploy reliable and scalable multicast services over IEEE 802.11 wireless LANs. First, in the IEEE 802.11 standards, multicasting is specified as a simple broadcasting mechanism that does not make use of *Acknowledgement* (ACK) frames. The absence of a feedback mechanism has a strong impact on the reliability of the service provided to the user. Second, according to the IEEE 802.11a/b standards, all frames with multicast and broadcast *Receiver Address* (RA) should be transmitted at one of the rates included in the basic rate set. However, the standards do not provide the guidelines on the best rate to be used, and more important how to adapt the rate accordingly to the wireless channel operating conditions vary along the time. Some proposals for unicast communications have been reported in the literature to adapt the PHY data rate according to the channel conditions, such as the ARF (*Auto Rate Fallback*) [3] or RBAR (*Receiver Based Auto Rate*) mechanisms [4]. However, these mechanisms rely on the estimation of individual channel states, and therefore they can not be directly applied to the multicast service. The difficulty comes from the fact that the channel conditions between the *Access Point* (AP) and each one of the *Mobile Terminals* (MTs) in the multicast group may differ and in the absence of feedback, the AP does not have any means to get to know the operating conditions of each individual MT.

In this paper, we first introduce a novel *Auto Rate Selection* mechanism for *Multicast* in multirate wireless LAN, from now on referred as the ARSM mechanism. Basically, the ARSM mechanism dynamically selects the multicast data rate based on the channel conditions perceived by the MTs. The main idea behind our proposal is to identify the AP to MT channels exhibiting the worst conditions, expressed in terms of the SNR ratio. The channel rate is then set accordingly, i.e., by using the highest possible channel rate enabling the provisioning of reliable multicast services. Even though such mechanism has proved to be effective as reported in one of our previous works [5], such setting heavily penalizes those MTs exhibiting better channel conditions. There is then not only a need of a multicast mechanism capable of dynamically adapting to the quick changes of the channel, but of a multicast service also capable of serving an heterogeneous population of MTs characterized by the quality of the AP to MT channel.

In [5], we have presented a preliminary evaluation of ARSM which proved effective in adapting the channel rate taking into account the varying channel conditions. In this paper, we go a step further by enhancing the ARSM scheme to

Manuscript received May 15, 2006; revised December 17, 2006. This work was supported by the Ministry of Science and Education of Spain/Castilla-La Mancha under Projects CSD2006-46, TIN2006-15516-C04-02, PAI06-0106, by the Korean Government Grant, MOEHRD (KRF- 2005-214-D00340) and by the French Ministry of Research ANR Project "DIVINE."

J. Villalón, P. Cuenca, and L. Orozco-Barbosa are with the Instituto de Investigación en Informática de Albacete, 02071 Albacete, Spain (e-mail: josemvillalon@dsi.uclm.es; pcuenca@dsi.uclm.es; lorozco@dsi.uclm.es).

Y. Seok is with LG Electronics Institute of Technology, 16 Woomyeon-Dong, Seocho-Gu, Seoul, Korea (e-mail: yhseok@lge.com).

T. Turletti is with INRIA Projet Planéte, 2004 route des Lucioles, 06902 Sophia Antipolis, France (e-mail: thierry.turletti@sophia.inria.fr).

Digital Object Identifier 10.1109/JSAC.2007.0705xx.

address the aforementioned issues. We focus our attention to its use in multicast video services. For such setup, we further consider the use of hierarchical video coders in order to adapt the video transmission to the channel conditions. In such devices, the video is encoded into two layers: the base and the enhancement layers. In particular, we propose the use of hierarchical video coders based on the new H.264 standard. The main idea is that the MTs exhibiting the worst channel conditions will receive only the base layer of the encoded H.264 video, while the MTs exhibiting better channel conditions will receive both, the base and enhancement layers. We evaluate the effectiveness of such approach via simulation.

The dependencies shown above between application, MAC and physical layers are then solved following the principles of cross-layer protocol engineering [6]–[8]. The multicast protocol introduced herein is able to adapt the PHY data rate by making use of the physical channel conditions, normally not available to the MAC layer. In the ARSM scheme, the PHY and MAC layers collaborate in the auto rate selection process for multicast services. We then extend this approach to the application layer, enabling the provisioning of a multicast hierarchical video service over wireless networks.

The remainder of this paper is organized as follows. We start in Section II by providing some background on the issues to be addressed on the design of multicast services to be deployed in a multirate wireless LAN and on the design of hierarchical video codecs. The proposed ARSM mechanism adapted to support hierarchical video services is described in Section III. Section IV presents our simulation results. In Section V, we summarize and outline our future work plans.

II. Background

A. Multicast for Multirate Wireless LANs

Most research efforts on multicasting in IEEE 802.11 WLANs have focused on improving the service reliability by integrating ARQ mechanisms into the protocol architecture. In [9], the *Leader-Based Protocol* (LBP) ARQ mechanism has been introduced to provide the multicast service with some level of reliability. To address the ACK implosion problem, LBP assigns the role of group leader to the multicast receiver exhibiting the worst signal quality in the group. The group leader holds the responsibility to acknowledge the multicast packets on behalf of all the multicast group members, whereas other MTs may issue *Negative Acknowledgement* (NACK) frames when they detect errors in the transmission process.

The *802.11MX* reliable multicast scheme described in [10] uses an ARQ mechanism supplemented by a busy tone signal. When an MT associated to a multicast group receives a corrupted packet, it sends a NACK tone instead of actually transmitting a NACK frame. Upon detecting the NACK tone, the sender will retransmit the data packet. Since the 802.11MX mechanism does not need a leader to operate, it performs better than the LBP protocol in terms of both data throughput and reliability. However, this mechanism is very costly since it requires a signaling channel to send the NACK frames and busy tones. Moreover, both LBP and 802.11MX schemes do not adapt the multicast PHY rate to the state of receivers.

Very recently, the RAM scheme has been proposed in [11] for reliable multicast delivery. Similar to the LBP and 802.11MX schemes, the transmitter has first to send a RTS frame to indicate the beginning of a multicast transmission. However, in RAM the RTS frame is used by all the multicast receivers to measure the *Receiver Signal Strength* (RSS). Then, each multicast receiver has to send a variable length dummy CTS frame whose length depends on the selected PHY transmission mode. Finally, the transmitter senses the channel to measure the collision duration and can adapt the PHY rate transmission of the multicast data frame accordingly. This smart solution is more practical than 802.11 MX since it does not require a signaling channel but still requires the use of RTS/CTS mechanism and targets reliable transmission applications.

In [12], *SNR-based Auto Rate for Multicast* (SARM) is proposed for multimedia streaming applications. In SARM, multicast receivers measure the SNR of periodically broadcast beacon frames and transmit back this information to the AP. To minimize feedback collision, the backoff time to send this feedback increases linearly with the received SNR value. Then, the AP selects the lowest received SNR to adapt the PHY rate transmission. The main problem with this approach is that the transmission mode cannot be adapted for each multicast frame. The multicast PHY rate of SARM is adapted at each beacon intervals. SARM does not make use of any error recovery mechanism, such as, data retransmission.

Note that at the exception of RAM and SARM, the mechanisms just described above only focus on solving the reliability of the multicast service in WLANs. Only RAM and SARM adapt the PHY transmission rate of the multicast data frames. In this paper, we define an architecture by integrating the following facilities: 1) the optimal channel rate adaptation of the multicast service in IEEE 802.11 WLANs, 2) a more reliable transmission of the multicast data, 3) the limitation on the overhead required by the signaling mechanism, and 4) the support of heterogeneity of receivers by using different multicast groups and hierarchical video coding. The definition of the proposed cross layer architecture is based on the multirate capabilities present in the PHY layer of IEEE 802.11 WLANs.

B. Use of Hierarchical Video Coding

Hierarchical video coding has been proposed in the literature in order to improve the robustness of video communications applications [13], [14]. It has also been proposed as a smart solution to handle the heterogeneity of receivers in multimedia multicast transmission over the wired Internet, such as in RLM-based schemes [15]–[17] and SARC [18]. Basically, in a hierarchical encoding scheme, the most relevant elements of the video sequence are included in a *base layer*, while less relevant pieces of information are put into a second level, also denominated *enhancement layer*. According to their relevance, the base layer usually receives a high-priority treatment while the other layer is delegated to a second plane. The base layer provides by itself a minimum acceptable quality video image. Various implementation issues must be considered when designing hierarchical video codecs, such as, the amount of overhead required to implement it, the definition of the breakpoint used when splitting the encoded video bitstream into the base and enhancement layers and the ability of assigning different priorities to the underlying network mechanisms [19].

One of the main advantages of hierarchical coding is that this technique can be applied to all encoding schemes, such as, H.261, H.263, MPEG-1, MPEG-2, MPEG-4, H.264, among others. In this paper, we will consider the most recent H.264 video encoding standard, also known as MPEG-4 AVC [20], which is highly efficient by offering perceptually equivalent video quality at about 1/3 to 1/2 of the bitrates offered by the MPEG-2 format. In a nutshell, H.264 consists of two conceptually different layers. First, the *Video Content Layer* (VCL) contains the specification of the core video compression engines that achieve basic functions such as motion compensation, transform coding and entropy coding. This layer is transport-unaware, and its highest data structure is the video slice, a collection of the coded *Macroblocks* (MBs) in scan order. Second, the *Network Abstraction Layer* (NAL) is responsible for the encapsulation of the coded slices into transport entities of the underlying protocols. Additionally, this standard introduces a set of error resilience techniques such as data-partitioning that is an effective application-level framing technique that divides the compressed data into separate units of different importance. Data-partitioning creates more than one bit string (partition) per slice, and allocates all symbols of a slice into an individual partition with a close semantic relationship. In H.264, the following three different partitions types are used:

- **Partition A**, containing header information such as MB types, quantization parameters and motion vectors. This information is key to the system operation since the other two partitions fully rely on it.
- **Partition B** (intra partition), carrying intra coded block pattern and intra coefficients that can stop further drift error.
- **Partition C** (inter partition), containing inter coded block pattern and inter coefficients. This information is the least relevant one.

Usually, partitions A and B can provide by themselves a minimum acceptable video quality in situations in which the partition C is completely lost. Note that the H.264 video coding standard does not specify any hierarchical coding scheme. In this paper, rather than using hierarchical coding, we favor more interaction between, on one hand, the H.264's VCL layer that divides the original stream through data-partitioning and, on the other hand, the MAC layer that treats video streams differently. In particular, the application layer passes its traffic information (the priority of the stream) with their QoS requirements to the MAC layer, which maps these partitions to different traffic categories to improve the perceived video quality. In our case, the base layer will contain the partitions A and B (and other important control information at the beginning of the sequence such as picture size, display window, MB allocation map, ...), and the enhancement layer will contain the partition C.

III. Auto Rate Selection for Multicast (ARSM)

The ultimate goals of the ARSM mechanism to be introduced herein is: 1) to enable the deployment of a more reliable and efficient multicast protocol than the one defined by the IEEE 802.11 standard; and 2) to integrate the multicast service into the cross-layer architecture proposed for multirate wireless networks. By efficient, we mean that the overhead required by the ARSM to operate should be kept to minimum levels. In the following sections, we introduce the various mechanisms making part of ARSM.

2	2	6	6	1	4
Frame Control	Duration	Dest Address	Source Address	SNRleader	FCS

(a) Multicast Probe Frame

2	2	6	1	4
Frame Control	Duration	Dest Address	SNRmp	FCS

(b) Multicast Response Frame

Fig. 1. Special multicast control frames of MCPO.

A. Design of ARSM

ARSM is an adaptive mechanism in which the AP selects the PHY data rate to be used for the multicast service. The PHY data rate to be used is determined by taking into account the channel conditions perceived by each and every MT belonging to a given multicast group. Under the proposed scheme, the AP starts by multicasting a control frame, namely the *Multicast Probe* (MP) frame, to the multicast group members. Upon receiving the MP frame, each multicast member estimates the SNR of the channel, i.e., the quality of the wireless medium. Based on the SNR, each MT will determine the point in time for replying to the AP. According to the proposed mechanism, an MT having sensed the lowest SNR will be the one in charge of first replying to the AP, by issuing a *Multicast Response* (MR) frame. Upon detecting the transmission of the reply and in the absence of errors, all the other group members should normally refrain from replying to the AP. The AP assigns the role of group leader to this MT. The group leader holds the responsibility of acknowledging the multicast packets on behalf of all the multicast group members, whereas other MTs may issue NACK frames when they detect errors in the transmission process, in that case the AP retransmit the frame. The AP will select its PHY data rate using the SNR value contained in the received ACKs coming from the leader.

Fig. 1(a) shows the format of the MP frame. The duration field of the MP frame is initially set to $CW_m \times SlotTime$, where CW_m is the length of the contention window, expressed in slots, during which the group members may attempt to transmit the MR frame back to the AP. The destination address field of the MP frame represents the address of the multicast group being addressed by the AP, and the $\mathrm{SNR}_{\mathrm{leader}}$ field is set to the SNR received in the most recently ACK received by the AP.

After having sent the MP frame, the AP will wait for a period whose length is given by the *Short Inter Frame Space* (SIFS). At the time of sending the MP frame, the AP starts a timer, namely the *MP_timer*, initially set to CW_m slots. The timer is then decremented by one slot whenever the channel has been sensed idle for a period of time equal to one time slot ($SlotTime$). On the contrary, whenever the AP detects activity in the channel by means of the *Clear Channel Assessment* (CCA) mechanism, it immediately freezes the *MP_timer*.

When an MT receives the MP frame, it checks whether it is a member of this multicast group. If it is not, it sets the NAV parameter to $CW_m \times SlotTime$ by using the duration field included in the MP frame. Fig. 1(b) depicts the format of the MR frame. In the MR frame, the $\mathrm{SNR}_{\mathrm{mp}}$ field contains the SNR value of the previously received MP frame. When an MT replies to the AP with an MR frame, the MT uses a *BackoffTimer* in order to reduce the collision probability of the MR frames. This backoff timer is set based on the channel quality of the MTs. The MTs exhibiting the worst SNR choose the lowest backoff timer enabling them to transmit the MR frame earlier than all other MTs. In the absence of collision, all the other MTs to the one having successfully replied detect the transmission of the MR frame and refrain from transmitting. In this way, ARSM avoids the MP frame implosion problem.

According to the received information, and the value of the *MP_timer*, the AP could get to know the outcome of its query by three different means: *Explicit Feedback*, *Implicit Feedback* and *No Feedback*.

- **Explicit Feedback:** the AP receives the MR frame from an MT within the multicast group. In this case, the AP determines the SNR value of the MT with worse channel quality than the one of the current group leader. Then, it transmits the multicast data frames accordingly.
- **Implicit Feedback:** the AP receives a corrupted MR frame and the MP timer of the AP has not expired. This condition occurs when several MTs reply to the MP frame simultaneously and the MR frames have collided. In this case, the AP can predict the SNR value ($\overline{SNR_{mp}}$) of the MTs having sensed the worst channel quality. Through the current *MP_timer* of the AP, the AP identifies the lowest backoff timer among all the MTs in the multicast group. It must be mentioned that the MT with the lowest backoff timer first replies to the AP using an MR frame. The AP should already know the value of the backoff timer chosen by the MT to send the MR frame.
- **No Feedback:** The AP does not receive an MR frame and the *MP_timer* of the AP expires. This condition occurs when either all the MTs in this group have left or that the MP frame has been corrupted during its transmission. In this case, the AP will retransmit the MP frame after waiting for a period of time defined by the DCF backoff mechanism. The number of retransmission attempts for a given MP frame is limited to 4. When the number of retransmission attempts is reached, the AP assumes that there are no more MTs in the multicast group.

From the description above, it should be clear that the AP can determine the SNR as well as the identity of the MT to become the group leader in the first case: Explicit Feedback. However, if the MR frame collides (Implicit Feedback), the AP is unable to identify the MT to become the new leader. In this case, the AP will have to send a second MP frame before sending the following multicast data frame. The new MP frame to be sent out will set the $\mathrm{SNR}_{\mathrm{leader}}$ field set to a negative value. When the MTs in the multicast group receive the MP frame with the $\mathrm{SNR}_{\mathrm{leader}}$ field equal to a negative value, only those MTs having previously replied to the AP send a new MR frame. In this second round, these MTs do not use the backoff timer based on the SNR of the received signal, but a random value between $[0, CW_m - 1]$. This different backoff timer is used to further reduce the probability of collision of the MR frames.

Through the *Multicast Channel Probe Operation* (MCPO) described above, the AP can estimate the SNR value of the group leader. In order to reduce the amount of processing to be carried out by the MTs, we propose a *Dynamic Multicast Data Transmission* procedure by making use of several multicast PHY data rates. Under this scheme, the AP can be found in one of two different states depending on the feedback received.

- While the AP successfully delivers multicast data frame, the MCPO is deactivated. In this state, the AP will adapt its PHY data rate using the SNR value contained in the received ACK coming from the group leader.
- If the AP shows a failure of N_{th} consecutive multicast transmissions (detected via NACKs), it initiates the MCPO.

B. Throughput Analysis of ARSM

The main goal of the analysis presented herein is to determine the PHY rate in order to maximize the throughput performance in a multicast scenario with the ARSM mechanism. Towards this end, we make the following assumptions to simplify our analysis; (1) there is only one AP sender and a group of multicast receivers from which an MT group leader has been previously selected, therefore we neglect the MCPO frames, (2) the AP generates L-byte long data frame and its queue is never empty, i.e., it is never idle for a period longer than the required by the DCF medium access rules to initiate the following transmission, (3) the L-byte long data frame is not fragmented, (4) the retry limit is N_{max}, (5) we neglect the air propagation delays, (6) the ACK frame is transmitted at the lowest PHY mode as defined by the IEEE 802.11 standards, and (7) we assume an IEEE 802.11b PHY specification.[1]

Under the aforementioned assumptions, a transmission cycle is composed of the following phases that are repeated over time: (1) DIFS deferral phase; (2) Backoff/Contention; (3) Data transmission phase; (4) SIFS deferral phase; and (5) ACK transmission phase. The time length for items (1) and (4) are defined by the standards. We need then to derive items (2), (3) and (5). We assume that an L-byte long frame has to be transmitted using PHY mode m, where, for the IEEE 802.11b, m=1, 2, 3, and 4 for 1, 2, 5.5, and 11 Mbps, respectively.

For estimating the length of the Backoff/Contention phase, in accordance with the IEEE 802.11 standard, and due to the fact that the AP is unable to detect collisions under a multicast transmission, the CW is always set to CW_{min}. Under these conditions, the average backoff interval $\overline{T}_{backoff}$ in μsec, can be simply expressed by:

$$\overline{T}_{backoff} = \frac{CW_{\min}}{2} \cdot aSlotTime \tag{1}$$

[1]This analysis could be easily extended to other PHY specifications, such as 802.11 a/g.

The time length of the Data and ACK transmission phases, taking into account the PLCP *Protocol Data Unit* format described in the standard [1], are given by the following expressions

$$T_{data}^{m} = tPCLP_{preamble} + tPLCP_{header} + \frac{(MACoverhead_{data} + L) \cdot 8}{tx\_rate\,(m)} \quad (2)$$

$$T_{ack}^{m} = tPCLP_{preamble} + tPLCP_{header} + \frac{MACoverhead_{ack} \cdot 8}{tx\_rate\,(m=1)} \quad (3)$$

where $MACoverhead_{data}$ and $MACoverhead_{ack}$ are the overhead bytes corresponding to a Data and ACK MAC *Protocol Data Unit*, respectively (i.e. MAC header plus FCS fields), the transmission rate $tx\_rate(m)$ for PHY mode m is given by $tx\_rate(m)$=1, 2, 5.5, 11 Mbps for m=1, 2, 3, 4, respectively.

Each successful data transmission duration is equal to the data frame transmission time, plus the ACK transmission time, plus one SIFS. However, if the data transmission fails, the station has to wait for an ACK timeout period (for ARSM is defined in the same way that in the IEEE 802.11 standard, i.e., by adding up the time lengths of a SIFS period, the ACK transmission time and a slot time), the backoff period and the frame retransmission time. Therefore, the average transmission time for a single frame is given by (4) (see next page), where $P[n = i]$ is the probability of successful transmission at i transmission attempts and is given by:

$$P\,[n=i] = [1 - P_{success}^{m}\,(L)]^{i-1} \cdot P_{success}^{m}\,(L) \quad (5)$$

where $P_{success}^{m}\,(L)$ is the probability of a successful transmission and it can be calculated by:

$$P_{success}^{m}\,(L) = \left[1 - P_{e\_data}^{m}\,(L)\right] \cdot \left[1 - P_{e\_ack}^{m}\right] \quad (6)$$

where $P_{e\_data}^{m}\,(L)$ and $P_{e\_ack}^{m}$ are the error probabilities for an L-byte long data frame and ACK frame, respectively. Note that an ACK frame is much shorter than a data frame. Therefore, the error probability of the ACK frame is very low compared to the error probability of the data frame, and hence we can approximate the probability of a successful transmission as:

$$P_{success}^{m}\,(L) \approx 1 - P_{e\_data}^{m}\,(L) \quad (7)$$

Now we can derive the probability of error for an L-bytes long data frame transmitted at PHY mode m in the following way:

$$P_{e\_data}^{m}\,(L) = 1 - \left[1 - P_{e}^{m=1}\left(PLCP_{preamble/header}\right)\right] \cdot \left[1 - P_{e}^{m}\,(MACoverhead_{data} + L)\right] \quad (8)$$

where $P_{e}^{m=1}\left(PLCP_{preamble/header}\right)$ is the probability of error of the PLCP preamble/header transmitted using PHY mode 1, and $P_{e}^{m}\,(MACoverhead_{data} + L)$ is the probability of error of the MPDU including the MAC overhead. Furthermore, in [21] an upper bound has been derived on the packet error probability, under the assumption of using binary convolutional coding and hard-decision *Viterbi* decoding with independent errors at the channel input. For an L-byte long packet to be transmitted using PHY mode this bound is:

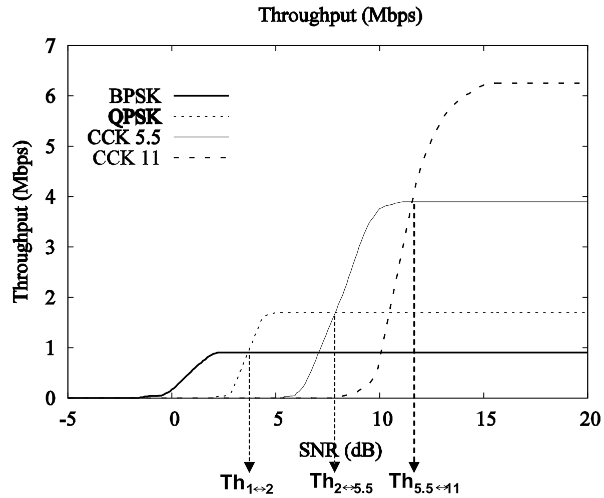


Fig. 2. IEEE 802.11b throughput vs. SNR for ARSM mechanism.

$$P_{e}^{m}\,(L) \leq 1 - [1 - P_{b}^{m}]^{8 \cdot L} \quad (9)$$

i.e. the $P_{e}^{m}\,(L)$ can be expressed in terms of BER, where P_{b}^{m} can be estimated for each PHY mode m using BER vs SNR curves for the IEEE 802.11b PHY modes. These curves could be derived theoretically [22], however for the purpose of this paper and to achieve a solution close to the reality, we have used empirical curves provided by *Intersil* for its chip called HFA3861B [23] measured in an *Additive White Gaussian Noise Channel* (AWGN) environment. These curves have been used to estimate the BER in both the analysis and the simulation results shown in the following sections.

Finally, to compute the throughput we need to consider the average time space between two successful frame transmissions, which will be given by $aDIFSTime + \overline{T}_{backoff}$. The average throughput can then be derived by:

$$\overline{Throughput} = \frac{8 \cdot L}{\overline{T}_{frame} + aDIFSTime + \overline{T}_{backoff}} \quad (10)$$

Using the IEEE 802.11b PHY layer values specified in [2] in the equations derived above and for a frame size of L=1000 bytes, we obtain the optimal throughput as a function of the SNR, for every PHY mode, as shown in Fig. 2.

From Fig. 2, it is clear that in order to determine the best reliable transmission rate at a given time for all multicast group members, the ARSM mechanism needs to know in advance: (1) the values for the thresholds $\mathrm{Th}_{1\leftrightarrow 2}$, $\mathrm{Th}_{2\leftrightarrow 5.5}$ and $\mathrm{Th}_{5.5\leftrightarrow 11}$ (see Fig. 2) and (2) the SNR feedback from one of the MT exhibiting the worst channel conditions. The thresholds (item 1 above) can be derived from the previous analysis. For the best estimation possible of the value for item (2) is crucial the election of the MT assuming the role of group leader. This latter task is done by means of the MCPO mechanism of ARSM. According to the design of MCPO, one of the MT having detected the lowest SNR interval will be the one in charge of first replying to the AP, by issuing an MR frame.

$$\overline{T}_{frame} = \sum_{i=2}^{N_{\max}} P\left[n=i\right] \cdot \sum_{j=1}^{i-1} \left[\begin{array}{c} aSIFSTime + T_{ack}^{m} \\ +aSlotTime + \overline{T}_{backoff} + T_{data}^{m}\left(L\right) \end{array} \right] + T_{data}^{m}\left(L\right) + aSIFSTime + T_{ack}^{m} \tag{4}$$

In order to comply with the standard specifications, the value for CW_m has to be set to the *Extended IFS* parameter, i.e. equal to 8. This setup will enable the smooth integration of our proposal into the architecture of IEEE 802.11 WLANs systems concurrently supporting unicast and multicast services. Under this same scenario, an MT belonging to the multicast group could send its reply by randomly selecting a slot from the interval $[0, CW_m - 1]$. However, we propose to divide the window into three sub-window in order to segregate the stations into three groups according to their channel conditions. We have then divided the backoff window [0,7] of the standard in the following three intervals [7,6], [5,3] and [2,0]. Since much likely the MCPO will be invoked to reduce the PHY mode being used, the main issue is to identify if the new leader will force the channel rate of the multicast group one or two modes down. For instance, the channel rate may be dropped from 11 to 5.5 or even down to 2 Mbps. We have then mapped the backoff window subintervals into the SNR levels sensed by the MTs having been sensed:

$$\textit{BackoffTimer} = \begin{cases} [0,2] & SNR_{mp} < Th_{\leftarrow 2} \\ [3,5] & Th_{\leftarrow 2} \leq SNR_{mp} < Th_{\leftarrow 1} \\ [6,7] & Th_{\leftarrow 1} \leq SNR_{mp} \end{cases} \tag{11}$$

where $Th_{\leftarrow i} \in \{\mathrm{Th}_{1\leftrightarrow 2}, \mathrm{Th}_{2\leftrightarrow 5.5}, \mathrm{Th}_{5.5\leftrightarrow 11}\}$ for i=1 and 2, are the thresholds of the channel rates corresponding to one or two modes down of the mode being currently used by the AP and having been previously fixed by the current value of the SNR of the multicast leader, $\mathrm{SNR}_{\mathrm{leader}}$. That is to say if $\mathrm{Th}_{5.5\leftrightarrow 11} \leq \mathrm{SNR}_{\mathrm{leader}}$ then $\mathrm{Th}_{\leftarrow 1}=\mathrm{Th}_{5.5\leftrightarrow 11}$ and $\mathrm{Th}_{\leftarrow 2}=\mathrm{Th}_{2\leftrightarrow 5.5}$. Otherwise if $\mathrm{Th}_{2\leftrightarrow 5.5} \leq \mathrm{SNR}_{\mathrm{leader}} < \mathrm{Th}_{5.5\leftrightarrow 11}$ then $\mathrm{Th}_{\leftarrow 1}=\mathrm{Th}_{2\leftrightarrow 5.5}$ and $\mathrm{Th}_{\leftarrow 2}= \mathrm{Th}_{1\leftrightarrow 2}$. A special case is when $\mathrm{Th}_{1\leftrightarrow 2} \leq \mathrm{SNR}_{\mathrm{leader}} < \mathrm{Th}_{2\leftrightarrow 5.5}$, in this case $\mathrm{Th}_{\leftarrow 1}=\mathrm{Th}_{1\leftrightarrow 2}$ and $\mathrm{Th}_{\leftarrow 2}=\mathrm{Th}_{\leftarrow 1}/2$. From this expression, one of the MTs classified as belonging to the group of MTs having sensed the worst SNR chooses the shortest backoff window. In this way, such MT will be able to place its reply before the stations having sensed much better channel conditions.

For the *Implicit Feedback* case, using equation 12, the AP can inversely estimate the SNR range with the shortest backoff window, where $\underline{BT_{mp}}$ is the current *MP_timer* value known by the AP and $\overline{SNR_{mp}}$ is the estimated worst SNR value.

$$\overline{SNR_{mp}} = \begin{cases} 0 & BT_{mp} \geq 6 \\ Th_{\leftarrow 2} & 6 > BT_{mp} \geq 3 \\ Th_{\leftarrow 1} & 3 > BT_{mp} \geq 1 \end{cases} \tag{12}$$

C. H-ARSM: Adaptation of ARSM to Hierarchical Video Communications

As described in the previous sub-section, the ARSM mechanism adapts the PHY data rate of the multicast group taking into account the receiver whose SNR lies in the lowest range. In this way, the mechanism guarantees that all members of the multicast group properly receive all packets while transmitting the packets at the highest possible PHY data rate. However, the use of this mechanism under certain scenarios, such as video services, heavy penalizes those users capable of receiving data at higher rates. Furthermore, as shown in [24], the performance of an IEEE 802.11 WLAN is severely affected when operating at such low PHY data rates. It is for this reason that we propose enhancing the ARSM to support services capable of accommodating users with different capabilities, in particular hierarchical video communications. We will refer to our proposal as hierarchical ARSM or simply by H-ARSM.

Under H-ARSM, the video is expected to be encoded into two layers, namely the base and enhancement layers. The packets containing the base layer of the video are sent to all the members of the multicast group following the rules as established by ARSM. In this way, the mechanism should guarantee a minimum video quality to all users. In the case of the enhancement layer, the operating mode is quite similar to the one used by the ARSM scheme, i.e., the AP has to first select a group leader for the enhancement layer, whose main mission is to acknowledge (ACK) the packets sent to the group. The other MTs may issue NACK frames when they detect errors in the transmission process, in that case the AP retransmit the frame. The main difference for enhancement layer comes from the fact that instead of selecting one of the members with the lowest SNR as the group leader, this new mechanism selects one of the members with the highest SNR. The rate of the enhancement layer is then selected by H-ARSM taking into account the rate channel at which the base layer is being transmitted, in the following way:

- Once that the ARSM has decided that the base layer should be transmitted at 1 or 2 Mbps, then H-ARSM decides that the enhancement layer should be transmitted at 5.5 Mbps if there is one (or more) MT capable of receiving the enhancement layer (MTs with $\mathrm{SNR} \geq \mathrm{Th}_{2\leftrightarrow 5.5}$). In this case, an MT belonging to enhancement layer should leave the enhancement layer multicast group whenever its SNR falls below $\mathrm{Th}_{2\leftrightarrow 5.5}$.
- If the ARSM has decided that the base-layer should be transmitted at 5.5 Mbps, then H-ARSM decides that the enhancement layer should be transmitted at 11 Mbps if there is one (or more) MT belonging to the enhancement layer (MTs with $\mathrm{SNR} \geq \mathrm{Th}_{5.5\leftrightarrow 11}$) multicast group. In this case, an MT belonging to the enhancement layer should leave this group whenever its $\mathrm{SNR} < \mathrm{Th}_{5.5\leftrightarrow 11}$.
- If ARSM has decided that the base layer should be transmitted at 11 Mbps, then H-ARSM decides that the enhancement layer should also be transmitted at 11 Mbps.

The format for the MP and MR frames used by this mechanism is the same that the one used by the original ARSM. However, in the case of the enhancement layer, the $\mathrm{SNR}_{\mathrm{leader}}$ field of the MP contains the maximum SNR (SNR_{max}) known by the AP (from the feedback of the leaders) measured during all time. Because we want to

$$BackoffTimer = \begin{cases} [0,1] & SNR_{mp} \geq Th_{enhancement} + FM1 \\ [2,4] & Th_{enhancement} + FM1 > SNR_{mp} \geq Th_{enhancement} + FM2 \\ [5,7] & Th_{enhancement} + FM2 > SNR_{mp} \geq Th_{enhancement} \\ CW_m & Th_{enhancement} > SNR_{mp} \end{cases} \quad (13)$$

know the member with the highest SNR possible, a member of the enhancement layer multicast group which receives an MP frame, will use the following rule to adapt its backoff timer for transmitting its MR frame (13), where $Th_{enhancement}$=$\mathrm{Th}_{2\leftrightarrow 5.5}$, $FM2$ =$\mathrm{Th}_{5.5\leftrightarrow 11}-\mathrm{Th}_{2\leftrightarrow 5.5}$ and $FM1$ =$\mathrm{Th}_{5.5\leftrightarrow 11}-\mathrm{Th}_{2\leftrightarrow 5.5}$ + $(SNR_{\max} - \mathrm{Th}_{5.5\leftrightarrow 11})/2$ when ARSM has decided that base layer is transmitted to 1 or 2 Mbps, or $Th_{enhancement}$=$\mathrm{Th}_{5.5\leftrightarrow 11}$, $FM1$=2 $\cdot$ $(SNR_{\max} - \mathrm{Th}_{5.5\leftrightarrow 11})/3$ and $FM2$ = $(SNR_{\max} - \mathrm{Th}_{5.5\leftrightarrow 11})/3$ when ARSM has decided that base layer is transmitted to 5.5 or 11 Mbps.

From this expression, one of the MTs classified with the best SNR chooses the lowest backoff timer and has a chance to transmit the MR frame earlier than all other MTs. We have selected four backoff intervals following the same reasoning as for (11), in order to better identify the MT exhibiting the best operating conditions. The *BackoffTimer* will choose a random value inside the selected interval.

IV. Performance Evaluation

In this section, we carry out a performance evaluation on the effectiveness of our cross-layer architecture proposed. Throughout our study, we have made use of the OPNET Modeler tool 11.5 [25], which already integrates the IEEE 802.11 simulator. We have integrated into it the LBP, RAM, ARSM and H-ARSM mechanisms.

A. Scenarios

Our performance evaluation has been structured in the following way: first we evaluate and compare the ARSM, RAM and LBP schemes when hierarchical video coding is not applied. This should allow us to show the benefits of using an auto rate selection multicast mechanism over a multirate wireless LAN. Then, we present the comparative performance evaluation of the RAM, H-ARSM and ARSM schemes when hierarchical video coding is applied in order to show the benefits of an adaptive video multicast streaming over multirate wireless LANs. To this end, we have conducted an exhaustive campaign of simulations.

In our simulations, we model an IEEE 802.11b WLAN consisting of an AP, nine or eighteen multicast wireless MTs, and five unicast wireless MTs. We have varied the network size expressed in terms of the area covered by the AP and MTs. The network size has been initially set to a geographical area of 50m $\times$ 50m. We have then increased the network size in both dimensions by 10m $\times$ 10m to a maximum network size of 140m $\times$ 140m. The access point is located in the center of the BSS. The cell size is changed throughout the different scenarios under study. The multicast MTs move randomly within the BSS at a constant speed of 5 km/h, whereas the unicast MTs are static and placed close to the access point. We assume that the unicast packets are always transmitted at 11 Mbps. For the ARSM and H-ARSM schemes we have found that setting $Nth = 3$, is a good compromise to limit the number of MP frames to be sent and the time to react to a change on the network operating conditions. Similar findings have been reported in [26].

In our scenarios, we have assumed the use of two types of traffic flows: multicast traffic downlink flows and unicast traffic uplink flows. For the downlink traffic, the access point transmits a video stream to the multicast MTs. For the video streaming source, we have used traces generated from a variable bit-rate H.264 video encoder [27]. We generate two types of traces corresponding to the use or not of the hierarchical video coding scheme presented in Section II-B. We have used the sequence *Mobile Calendar* encoded on CIF format at a video frame rate of 25 frames/s. When the hierarchical coding is used, the base layer accounts approximately for 30% of the total video data. Throughout our experiments, we have confirmed that an acceptable video quality in the base layer can be obtained while the amount of data traffic pertaining to the enhancement layer is in the order of 70%. We have used two average video transmission rates (around 400 kbps and 700 kbps) using different quantization factors with a packet size equal to 1000 bytes (including RTP/UDP/IP headers). This video application is randomly activated within the interval $[1, 1.5]$ seconds from the start of the simulation. In order to limit the delay experienced by the video streaming application, the maximum time that a video packet may remain in the transmission buffer has been set to 2 seconds. Whenever a video packet exceeds these upper bounds, it is dropped. For the unicast traffic, we assume greedy sources. The unicast packet size is equal to 1000 bytes (including the RTP/UDP/IP headers). The unicast sources are also randomly activated within the interval $[1, 1.5]$ seconds from the start of the simulation. Throughout our study, we have simulated two minutes of operation of each particular scenario.

B. Metrics

For the purpose of our performance study, the seven metrics of interest are: *multicast throughput*, *unicast throughput*, *multicast packet loss rate*, *overhead*, *delay*, *jitter* and the *video quality perceived by the end user*.

The *multicast throughput* is given by the successfully received average data rate by all the multicast MTs. To be able to better evaluate the various schemes with respect to the optimum case, we plot the normalized throughput which is calculated with respect to the multicast downlink traffic generated by the AP.

The *unicast throughput* is given by the total throughput received by the AP from all the unicast MTs. This metric should allow us to estimate the bandwidth not being used (made available to the unicast sources) by each one of the multicast schemes under consideration. If the multicast traffic

TABLE I
VIDEO QUALITY SCALE

RATING	IMPAIRMENT	QUALITY
5	Imperceptible	Excellent
4	Perceptible, not annoying	Good
3	Slightly	Fair
2	Annoying	Poor
1	Very annoying	Bad

is transmitted at the higher PHY rate, the unicast MTs can receive the higher throughput. Recall that the goal of the multicast PHY rate adaptation mechanism is to choose the highest PHY rate guaranteeing transmission reliability.

The *multicast packet loss rate* is defined by the ratio between the packets not having been received by at least a member MT of the multicast group over the total number of packets transmitted to the network.

The *overhead* is defined as the ratio between the number of control bits ($P_{control}$) and the total number of bits having been transmitted ($P_{control}$ + $P_{MulticastData}$); it can be simply stated as:

$$Overhead\,(\%) = \frac{\sum P_{control}}{\sum P_{control} + \sum P_{multicastData}} \times 100 \tag{14}$$

where ($P_{control}$) will be dependent on the mechanism being used.

The *average delay* defines the time elapsed from the time instant when the data packet is inserted into the channel and the time when the packet actually reaches the MT. In the case of the video traffic, we also evaluate the *jitter* between two consecutive video frames. Both metrics, delay and jitter, are key in the evaluation of QoS provided to video services.

Finally, we will present a quantitative assessment of *video quality perceived* using the *Moving Pictures Quality Metric* (MPQM) [28]. The MPQM video quality metric has been proved to behave consistently with human judgments according to the quality scale that is often used for subjective testing in the engineering community (see Table I) [29], [30].

Our measurements started after a warm-up period (about three seconds) allowing us to collect the statistics under steady-state conditions. Each point in our plots is an average over thirty simulation runs, and the error bars indicate 95% confidence interval.

C. Results

Figs. 3 and 4 show the results for the comparative analysis between ARSM, RAM and LBP schemes for a multicast load of 400 kbps, for 9 and 18 multicast wireless MTs, respectively. The results depicted in Fig. 3(a) show that the ARSM, RAM and LBP (1Mbps) schemes are able to provide a reliable multicast for all network sizes. For other rates, the performance of LBP decreases as the network size is increased. This is expected since adapting the PHY data rate helps to compensate for the signal impairments due to the distance to be covered by the signal. The results clearly show the benefits of adapting the PHY data rates taking into account the channel conditions.

While Figs. 3(a) and 3(c) show that the LBP is unable to provide good support to the multicast service even at rates as low as 5.5 Mbps, the ARSM and RAM schemes, are capable of effectively transmitting all the multicast traffic.

For the case of the unicast traffic, Fig. 3(b) shows that ARSM outperforms the RAM and LBP at 1Mbps for all network sizes. Furthermore, in the case of small-sized network, ARSM is even able to deliver twice the load carried by the LBP operating at 1Mbps. The figure also shows that ARSM outperforms the RAM and LBP schemes when this latter is able to fully deliver the multicast traffic. This is due to the fact that the ARSM scheme introduces less overhead than the RAM (see Fig. 3(d)), and LBP use a fixed PHY data rate.

Fig. 3(d) shows the overhead for all schemes under study. As seen form the figure, ARSM outperforms the RAM scheme. By requiring the exchange of RTS/CTS control frames for every multicast data frame to be sent, the RAM scheme introduces as much as twice the overhead required by ARSM. On the contrary, the MCPO mechanism of ARSM is invoked when needed. Fig. 3(d) also shows that ARSM requires less overhead to properly operate than LBP, except for LBP at 1 Mbps. From the results, it is clear that at high PHY data rates, the AP is forced to retransmit a large number of frames due to the high error rate. The overhead then increases due to the ACK/NACK control frames sent to overcome this anomaly.

Fig. 3(e) shows the delay for the multicast traffic. Fig. 3(e) shows that ARSM and RAM schemes outperform the LBP for all network sizes. Once again, this is due to the inadequacy of the PHY data rate being used. LBP is then forced to retransmit most of the frames due to the error transmissions, increasing the delay. The same reasoning can be applied to explain the results obtained for the jitter (Fig. 3(f)). These results clearly show the benefits of adapting the PHY data rates taking into account the channel conditions.

Finally, Fig. 4 shows that the ARSM scheme is able to cope with large multicast group sizes.

Figs. 5 and 6 show the results for the ARSM, RAM and H-ARSM (ARSM Hierar) mechanisms for a multicast load of 700 kbps, for 9 and 18 multicast wireless MTs, respectively. From the results shown in Fig. 5(a), it is clear that the H-ARSM scheme is capable of sending all the traffic associated to the base layer (ARSM Hierar.HP) independently of the network size. However, the throughput of the enhancement layer (ARSM Hierar.LP) decreases as the network size is increased. The reduction experienced by the enhancement layer is due to the decrease of the MTs pertaining to the enhancement layer as the network size increases.

Fig. 5(b) shows the throughput obtained for the unicast traffic being sent when one of each of the three multicast schemes is also being used for the multicast service. From the results, it is clear that H-ARSM offers better results than the RAM and ARSM schemes. In the case of large networks, the throughput obtained by the unicast services is 50% higher (3 Mbps) when H-ARSM is preferred over the RAM and ARSM (2 Mbps). In this case, the base layer is sent at low rates while the enhancement layer is sent at a rate no lower than 5.5Mbps.

Fig. 5(c) shows the packet loss rate for the three schemes. In the case of H-ARSM, the only packets being lost belong

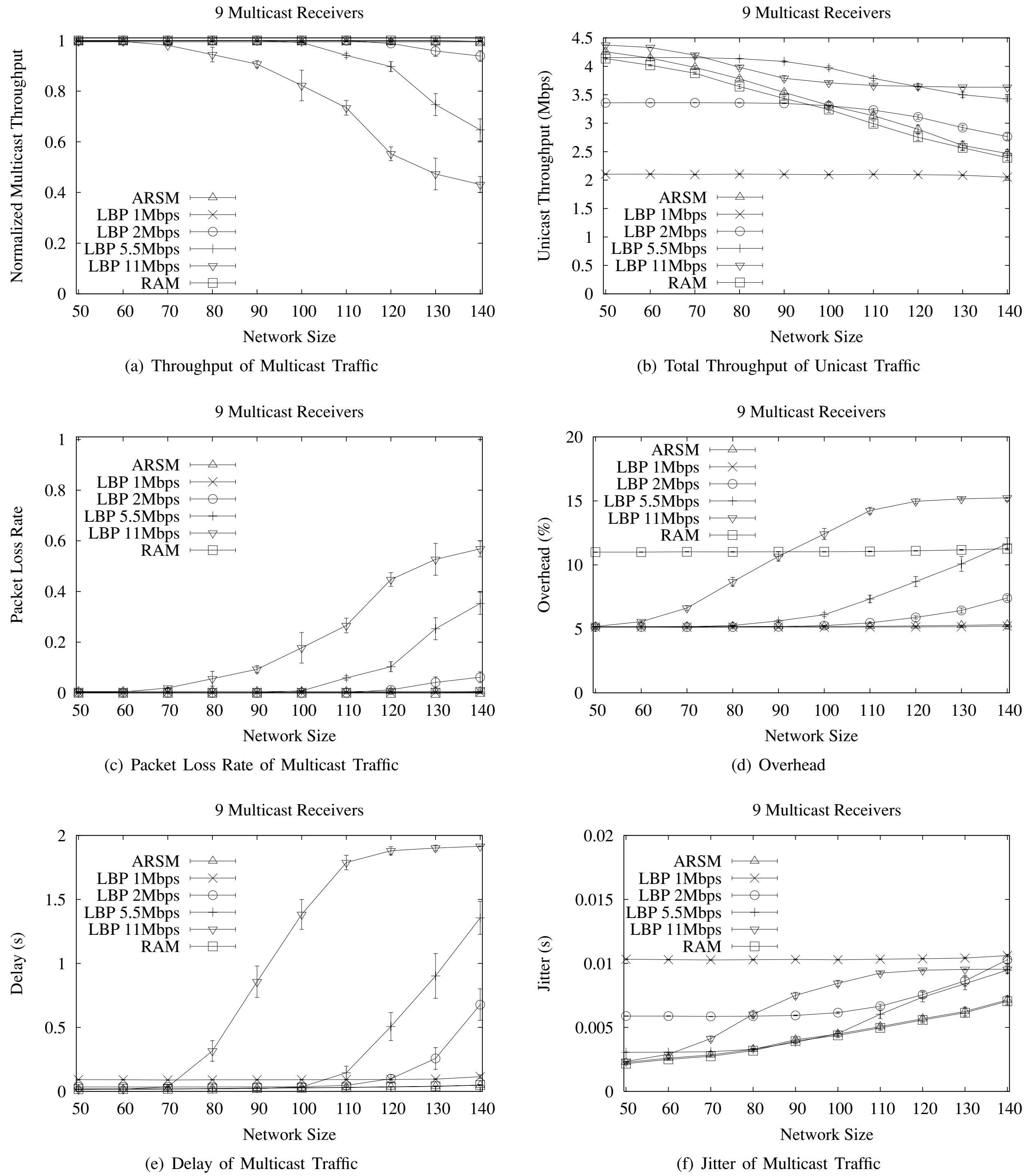
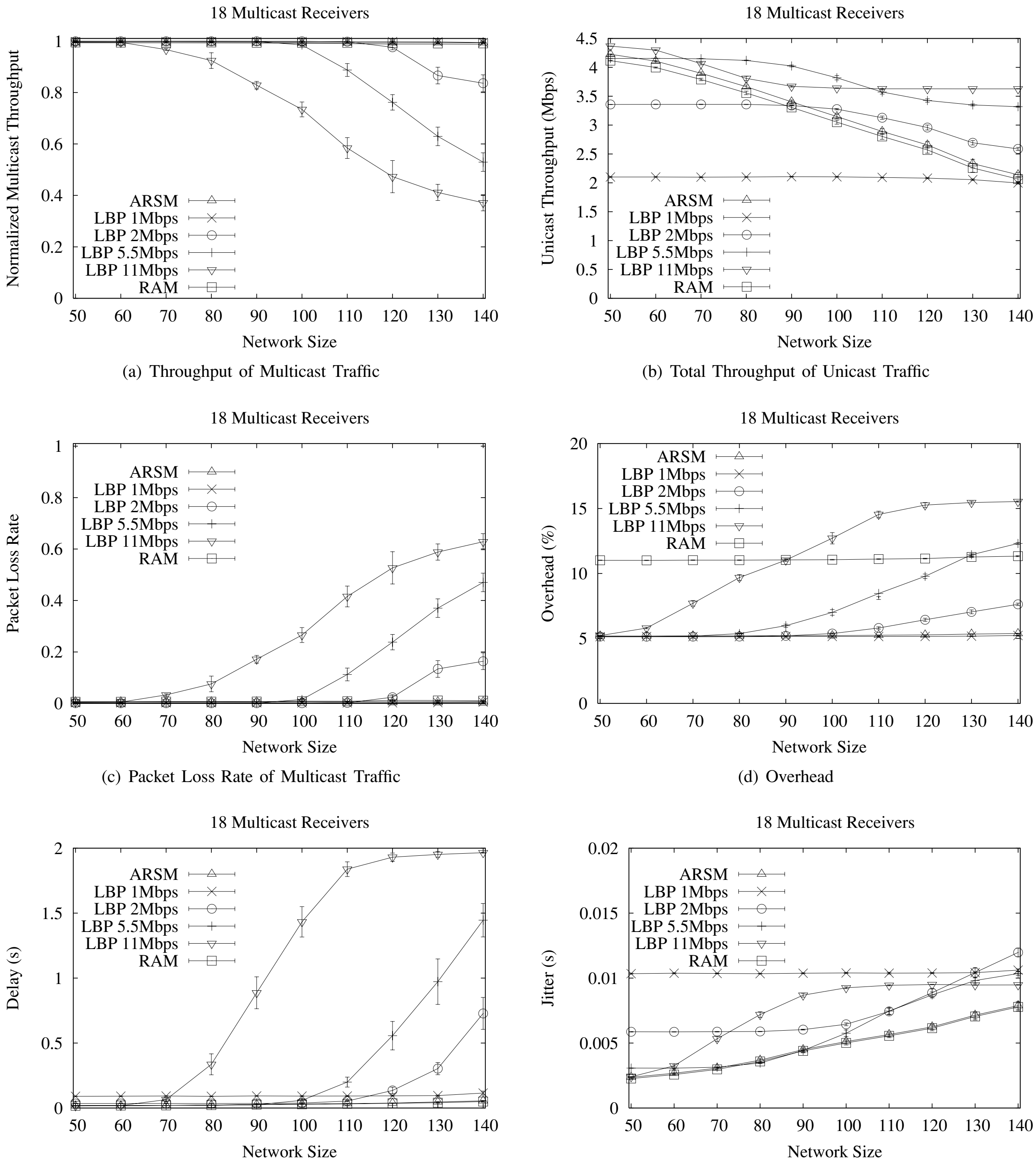


Fig. 3. Comparative Performance Evaluation (ARSM, RAM and LBP schemes). Multicast Group Size = 9.

to those carrying the enhancement layer and being addressed to the stations not belonging to the enhancement multicast group. The H-ARSM scheme guarantees the correct delivery of all the packets containing the base layer to all the members of the multicast group (see ARSM Hierar.HP). The scheme also guarantees the correct delivery of all the packets of the enhancement layer to all the members of the enhancement group. In the case of the ARSM and RAM schemes, all the packet losses are due to excessive delay by the packets while waiting to be transmitted. H-ARSM limits the packet losses conveying the enhancement layer to those stations exhibiting the worst channel conditions, i.e., those stations belonging only to the base layer multicast group. With RAM and ARSM, all the members of the multicast group are affected.

Fig. 5(d) shows once again that the H-ARSM scheme introduces lower overhead than the RAM scheme. This is expected since H-ARSM shares close similarities, in terms of the signaling procedure, to the ARSM mechanism. This translates into higher bandwidth available to the unicast services when H-ARSM is applied (see Fig. 5(b)). Fig. 5(e) shows the delay



(a) Throughput of Multicast Traffic

(b) Total Throughput of Unicast Traffic

(c) Packet Loss Rate of Multicast Traffic

(d) Overhead

(e) Delay of Multicast Traffic

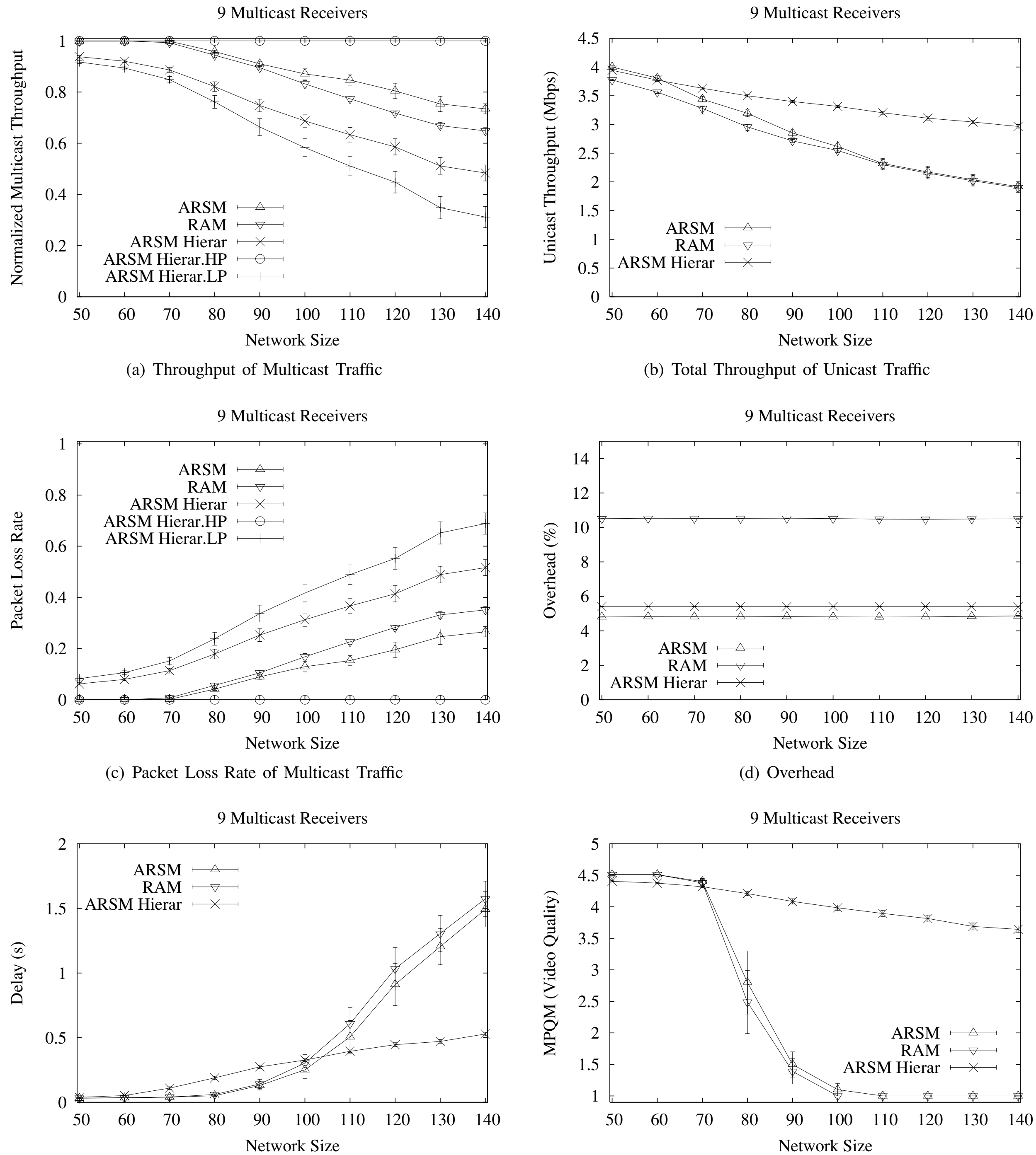
(f) Jitter of Multicast Traffic

Fig. 4. Comparative performance evaluation (ARSM, RAM, and LBP schemes). Multicast group size=18.

for the multicast traffic. The H-ARSM scheme outperforms the ARSM and RAM schemes. This is due to the excessive delay encountered by the packets while waiting to be transmitted when using the ARSM and RAM schemes.

Fig. 5(f) shows the quality of the video being received. In the case of the ARSM and RAM schemes, the video quality is rated around 4.4 (corresponding a quantization factor equal to 17) as long as no packet losses are reported. However, the video quality drastically falls as the packet loss rate increases (network sizes beyond 70 m) as a result of missing the packet delivery deadlines. In the case of the H-ARSM, a minimum video quality is always guaranteed since the base layer is always delivered.

The results depicted in Fig. 5(f) also show that the video quality decreases as the network size increases. This results from the increasing number of stations switching exclusively to the base layer multicast group. The fact of using a hierarchical transmission scheme clearly helps out; the multicast members are able to receive the packets according to the most up-to-date channel conditions.



(a) Throughput of Multicast Traffic

(b) Total Throughput of Unicast Traffic

(c) Packet Loss Rate of Multicast Traffic

(d) Overhead

(e) Delay of Multicast Traffic

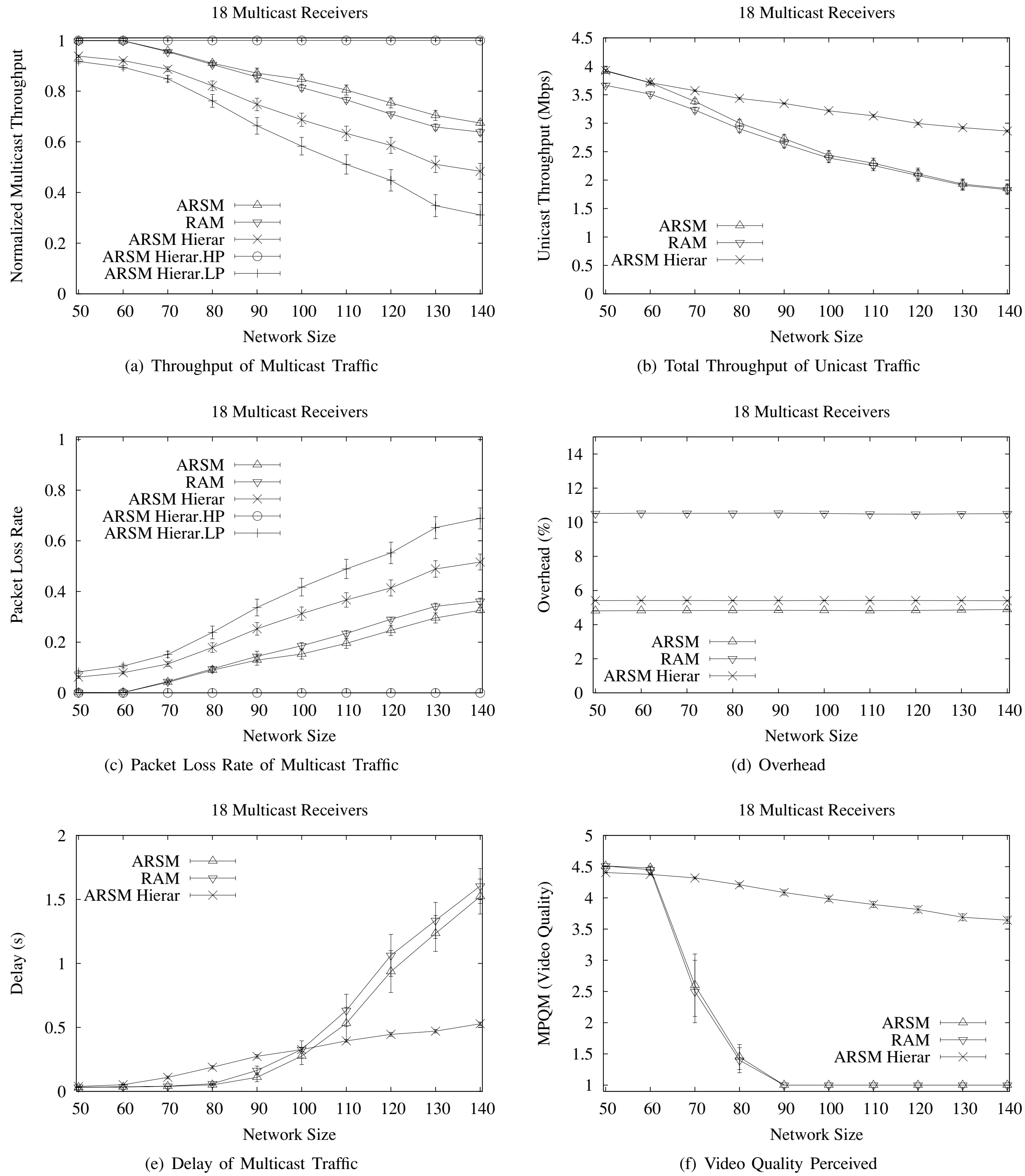
(f) Video Quality Perceived

Fig. 5. H-ARSM performance evaluation (multicast load=700kbps, 9 multicast receivers).

Fig. 6 shows that the H-ARSM can still provide a better service for a larger number of multicast wireless MTs. Similar to the previous scenario, the H-ARSM scheme exhibits better results than the ARSM and RAM schemes. On the contrary, ARSM and RAM exhibit worst results than those obtained in the previous scenario. In particular, they are unable to cope with larger group sizes. This is mainly due to the higher probability that one of the member of the multicast group will require to receive the data at a lower rate. Since the AP is required to send to all the members according to the rate setup taking into account the worst channel conditions, all the members of the multicast group are penalized. Fig. 6(f) clearly shows that the quality of the video signal degrades starting at network sizes of 60 m and falling to minimum levels for network sizes of 90 m.

V. Conclusions

We have proposed a cross-layer architecture for adaptive video multicast streaming over multirate wireless LANs where



(a) Throughput of Multicast Traffic

(b) Total Throughput of Unicast Traffic

(c) Packet Loss Rate of Multicast Traffic

(d) Overhead

(e) Delay of Multicast Traffic

(f) Video Quality Perceived

Fig. 6. H-ARSM performance evaluation (multicast load=700kbps, 18 multicast receivers).

layer-specific information is passed in both directions, top-down and bottom-up. This architecture requires knowing the operating conditions of the channel as perceived by the multicast members. The PHY data rate to be used for the multicast traffic is determined based on the feedback received by the group leader. We have also paid particular attention to limit the overhead introduced by the multicast rate adaptation mechanism. We have then proposed to use an H.264 hierarchical video coder in order to adapt the video transmission to the channel conditions perceived by the MTs. Therefore, we have considered jointly three layers of the protocol stack: the application, data link and physical layers. An extensive campaign of simulations has been carried out in order to analyze the impact of various key parameters over the performance of the proposed architecture, specifically the network size and the size of multicast receivers. Our results show that the real-time video streaming quality of the total system can be drastically improved by cross-layer signalling. Our cross layer architecture allows graceful video degradation while maximizing the available resources for background (unicast) traffic.

Our future plans will focus on extending the H-ARSM scheme to work with more than two layers for multirate IEEE 802.11a/g WLANs.

ACKNOWLEDGMENT

We thank Dr. C. J. Van den Branden and Dr. O. Verscheure for providing us with the video quality metric (MPQM) to evaluate the quality of the video sequences.

REFERENCES

[1] LAN MAN Standards Committee of the IEEE Computer Society, ANSI/IEEE Std 802.11, "Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications: High-speed Physical Layer in the 5GHz Band," IEEE 802.11 Standard, 1999.

[2] LAN MAN Standards Committee of the IEEE Computer Society, ANSI/IEEE Std 802.11, "Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications: High-speed Physical Layer Extension in the 2.4 GHz Band," IEEE 802.11 Standard, 1999.

[3] A. Kamerman and L. Monteban, "WaveLAN II: A high-performance wireless LAN for the unlicensed band," *Bell Labs Technical Journal*, pp. 118–133, 1997.

[4] G. Holland, N. Vaidya, and P. Bahl, "A rate-adaptive MAC protocol for multi-hop wireless networks," in *Proc. ACM MOBICOM*, July 2001.

[5] J. Villalón, Y. Seok, T. Turletti, P. Cuenca, and L. Orozco-Barbosa, "ARSM: Auto rate selection multicast mechanism for multirate wireless LANs," in *Proc. IFIP PWC*, Sep. 2006.

[6] V. Srivastava and M. Motani, "Cross-layer design: A survey and the road ahead," *IEEE Commun. Mag.,* vol. 43, no. 12, pp. 112-119. Dec. 2005.

[7] A. Ksentini, M. Naimi, and A. Gueroui, "Toward an improvement of H.264 video transmission over IEEE 802.11e through a cross-layer architecture," *IEEE Commun. Mag.,* vol. 44-1, pp. 107–114, Jan. 2006.

[8] I. Haratcherev, J. Taal, K. Langendoen, R. Lagendijk, and H. Sips, "Optimized video streaming over 802.11 by cross-layer signaling," *IEEE Commun. Mag.,* vol. 44-1, pp. 115–121, Jan. 2006.

[9] J. Kuri and S. K. Kasera, "Reliable multicast in multi-access wireless LANs," *ACM Wireless Networks*, vol. 7-4,

[10] S. K. S. Gupta, V. Shankar, and S. Lalwani, "Reliable multicast MAC protocol for wireless LANs," in *Proc. IEEE ICC*, May 2003.

[11] A. Basalamah, H. Sugimoto, and T. Sato, "Rate adaptive reliable multicast MAC protocol for WLANs," in *Proc. IEEE VTC*, May 2006.

[12] Y. Park, Y. Seok, N. Choi, Y. Choi, and J. M. Bonnin, "Rate-adaptive multimedia multicasting over IEEE 802.11 wireless LANs," in *Proc. IEEE CCNC*, Jan. 2006.

[13] P. Cuenca, L. Orozco-Barbosa, A. Garrido, and F. Quiles, "Loss-resilient ATM protocol architecture for MPEG-2 video communications," *IEEE J. Select. Areas Commun.,* vol. 18-6, pp. 1075–1086, June 2000.

[14] S. Worrall, A. H. Sadka, P. Sweeney, and A. M. Kondoz, "Prioritisation of data partitioned MPEG-4 video over mobile networks," *ETT-European Trans. Telecommunications*, vol. 12-3, May/June 2001.

[15] S. C. Brennan, N. K. Chilamkurti, and B. Soh, "Split-layer video multicast protocol: A new receiver-based rate-adaptation protocol," in *Proc. IEEE Network Computing Applications*, Apr. 2003.

[16] F. C. M. Martins and T. R. Gardos, "Efficient receiver-driven layered video multicast using H.263+ SNR scalability," in *Proc. ICIP*, Oct. 1998.

[17] S. McCanne, V. Jacobson, and M. Vetterli, "Receiver-driven layered multicast," in *Proc. ACM SIGCOMM*, Aug. 2005.

[18] J. Vieron, T. Turletti, K. Salamatian, and C. Guillemot, "Source and channel adaptive rate control for multicast layered video transmission based on a clustering algorithm," *EURASIP Journal Applied Signal Processing*, pp. 158-175, Feb. 2004.

[19] P. Cuenca, L. Orozco-Barbosa, F. Delicado, and A. Garrido, "Breakpoint tuning in DCT-based nonlinear layered video codecs," *EURASIP Journal Applied Signal Processing*, vol. 16, pp. 2555–2570, 2004.

[20] ITU-T RECOMMENDATION H.264, "Advanced video coding for generic audiovisual services," May 2003.

[21] M. B. Pursley and D. J. Taipale, "Error probabilities for spread-spectrum packet radio with convolutional codes and Viterbi decoding," *IEEE Trans. Commun.*, vol. 35, no. 1, pp. 1–12, Jan. 1987.

[22] D. Qiao, S.Choi, and K. G. Shin, "Goodput analysis and link adaptation for IEEE 802.11a wireless LANs," *IEEE Trans. Mobile Computing*, vol. 1, no. 4, pp. 278–292, Oct.–Dec. 2002.

[23] Intersil, "HFA3861B; Direct Sequence Spread Spectrum Baseband Processor," Jan. 2000.

[24] G. Berger-Sabbatel, F. Rousseau, M. Heusse, and A. Duda, "Performance anomaly of 802.11b," in *Proc. IEEE INFOCOM*, 2003.

[25] Opnet Technologies, Inc., OPNET Modeler 11.5 (c)1987-2006. [Online.] Available: http://www.opnet.com

[26] J. Pavon and S. Choi, "Link adaptation strategy for IEEE 802.11 WLAN via received signal strength measurement," in *Proc. ICC*, May 2003.

[27] Joint Video Team (JVT) of ISO/IEC MPEG and ITU-T VCEG, Reference Software to Committee Draft, JVT-F100 JM10.2. 2006. [Online.] Available: http://iphome.hhi.de/suehring/tml/

[28] C. J. Van den Branden and O. Verscheure, "Perceptual quality measure using a spatio-temporal model of human visual system," in *Proc. SPIE*, Jan. 1996, vol. 2668, pp. 450–461.

[29] P. Frossard and O. Verscheure, "AMISP: A complete contend-based MPEG-2 error-resilient scheme," *IEEE Trans. Circuits Syst. Video Technol.*, vol. 11, no. 9, pp. 989–998, Sep. 2001.

[30] O. Verscheure, P. Frossard, and M. Hamdi, "User-oriented QoS analysis in MPEG-2 video delivery," *Real-Time Imaging Journal*, no. 5, pp. 305–314, May 1999.

José Villalón received his M.Sc. degree in Computer Science from the University of Castilla-La Mancha, Albacete, Spain in 2003. He is currently a Ph.D. degree student in the Department of Computer Engineering at the University of Castilla-La Mancha. His research interests include high-performance networks, wireless networks, Quality of Service over IEEE 802.11, multicast transmission, video transmission and error-resilient protocol architectures.

Pedro Cuenca received his M.Sc. degree in Physics (award extraordinary) from the University of Valencia in 1994. He got his Ph.D. degree in Computer Engineering in 1999 from the Polytechnic University of Valencia, Spain. In 1995 he joined the Department de Computer Engineering at the University of Castilla-La Mancha where he is currently an Associate Professor and Vice-Dean of the Escuela Politécnica Superior de Albacete. He has also been a visiting researcher at The Nottingham Trent University, Nottingham (England) and at the Multimedia Communications Research Laboratory, University of Ottawa (Canada). His research topics are centered in the area of high-performance networks, wireless LAN, video compression, QoS video transmission and error-resilient protocol architectures. He is currently the chairman of the IFIP 6.8 Working Group.

Luis Orozco-Barbosa received the B.Sc. degree in electrical and computer engineering from Universidad Autonoma Metropolitana, Mexico, in 1979, the Diplome d'études Approfondies from École Nationale Supérieure d Informatique et de Mathématiques Appliquées (ENSIMAG), France, in 1984 and the Doctorat de l Université from Université Pierre et Marie Curie, France, in 1987, both in computer science. From 1991 to 2002, he was a Faculty member at the School of Information Technology and Engineering (SITE), University of Ottawa, Canada. In 2002, he joined the Department of Computer Engineering at Universidad de Castilla La Mancha (Spain). He has also been appointed Director of the Albacete Research Institute of Informatics, a National Centre of Excellence. He has conducted numerous research projects with the private sector and served as Technical Advisor for the Canadian International Development Agency (CIDA). His current research interests include Internet protocols, network planning, wireless communications, traffic modeling and performance evaluation. He is a member of the IEEE.

Yongho Seok received B.S. in computer engineering from Handong University, M.S. and Doctor of Engineering in computer engineering from Seoul National University, in 2000, 2002 and 2005, respectively. From October 2005 to September 2006, he was a post-doctoral researcher in PLANETE group, INRIA, France. He is a senior researcher in Personal Area Network Group, Information and Technology Lab., LG Electronics Institute of Technology. His working area in LG is IEEE 802.11 standard activity, such as IEEE 802.11s, IEEE 802.11v, DLP, AV streaming.

Thierry Turletti received the M.S. (1990) and the Ph.D. (1995) degrees in computer science both from the University of Nice-Sophia Antipolis, France. He has done his PhD studies in the RODEO group at INRIA. During the year 1995-96, he was a postdoctoral fellow in the Telemedia, Networks and Systems group at LCS, MIT. He is currently a research scientist in the Planète group at INRIA Sophia Antipolis. His research interests include multimedia applications and wireless networking. Dr. Turletti currently serves on the Editorial Board of *Wireless Communications and Mobile Computing* (WCMC), *Wireless Networks* (WINET) and *Advances in Multimedia* journals.