

A Cross-Layer Interworking of DVB-T and WLAN for Mobile IPTV Service Delivery

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Abstract—The convergence between existing and emerging technologies such as broadband, mobile, and broadcast networks is considered as a promising opportunity for existing network operators to increase their services' audiences. An important research effort is undertaken by main industry actors to face the multidimensional (transport technology, session signaling, and QoS provisioning) heterogeneity in the Next Generation Network (NGN). The per-session service adaptation is crucial to accommodate terminal and network heterogeneity experienced by different mobile users. In this paper, we investigate an intermediate architecture based on interconnection of DVB-T and 802.11 WLAN networks. The 802.11 WLAN is considered today as the “*de-facto*” wireless access network for the last miles connection. Both its per-service cost-effectiveness and wide acceptance by users make it an ideal candidate to bridge existing broadcast TV services toward the end-users.

We propose a seamless cross-layer inter-working between broadcasting network (i.e. DVB-T) and 802.11 WLAN network for adaptive and interactive mobile TV service delivery. We use a specific Adaptation Gateway (AG) at each Access Network to perform media adaptations. AG features a joint channel and video bit-rate awareness to tackle link degradations (signal strength quality, packet loss, etc.). Performance evaluations using an experimental test-bed are conducted and they show that our proposed cross-layer adaptation gateway reduces considerably packet losses and enhances the perceived quality of the TV service. By independently adapting each individual service, it is possible to address network and terminal heterogeneity and to offer interactive and personalized service to the end users.

Index Terms—Cross-layer adaptation, DVB-T, IPTV, Rate Control Algorithm (RCA), 802.11-based networks.

I. INTRODUCTION

SEVERAL communication architectures have been developed for delivering broadcast multimedia services to large number of handheld mobile devices. Typical examples are: DVB-T/H, cellular mobile networks (3G) with MBMS, and wireless broadband local and metropolitan networks (WiFi/WiMax). Most of existing broadcast network infrastructures cannot (or not in an affordable manner) accommodate both service interactivity and service adaptation to a large set of heterogeneous mobile users. To reduce the per-service cost, these networks provision a push (one-to-many) services

to cover a large number of users at the same time, and thus increase the profitability of their underlying resources.

Many modern multimedia communication applications such as mobile television (TV), pay-per view video, advanced TV-blogging, video telephony, streaming and gaming become now reality for wireless mobile networks. The tremendous success of such demanding services would mostly depend on the ability of service providers to surpass technical challenges related to providing visual content adaptation, delivering QoS-based and interactive services, and ensuring architecture scalability.

Thanks to its low deployment cost and to its inherent per-session control, 802.11 WLANs can overcome the main drawback in the service delivery chain. The IEEE 802.11 WLAN standard is being accepted and deployed in many different environments such as companies, universities, government institutions, and public places (airports, train stations, etc.). Its adoption is favored by the promises of the forthcoming 802.11n specifications topping 540 Mbit/s (raw throughput) with the help of MIMO technology. This development appears as a serious alternative for wired Ethernet, paving the way towards QoS-enabled added-value services provisioning. QoS provisioning for multimedia services is actually the main challenge to overcome in order to render WLANs affordable in most of Network Operator's commercial offers, and ultimately more attractive for particular use [2]. We believe that a combined DVB-T and WLAN architecture is today a viable alternative for service providers to provision QoS-enabled TV (A/V: Audio/Visual) services for heterogeneous wireless terminals. A large-scale 802.11-based WLAN deployment in a public place and city is envisioned to cover population with Internet connection.

In this paper, we propose an interworking of broadcast network (DVB-T) and 802.11 WLAN to deliver a new class of interactive and adaptive services using an Adaptation Gateway (AG) at each Wireless Access Point (AP).

The AG plays the role of service and content provider in the sense that it centralizes the adaptation of individual TV session. Many features are provided at AG level including (1) browsing existing video services (2) subscribing to an existing quality of service offer by contracting a Service Level Agreement, and (3) guarantying a QoS level by cross-layer (joint link-layer / application-level) service adaptation to deal with network and signal quality degradation.

The cross-layer adaptation is coordinated to produce the best viewer-perceived quality. Based on link quality measurements, we first figure out the best transmission rate, at the physical layer that best suits the channel conditions experienced by the end-user. Based on a pre-established knowledge of the maximum achievable throughput for a given physical layer rate, we

Manuscript received July 16, 2006; revised November 13, 2006. This work was supported by the Sixth EU Framework Program for Research and Development 2006-2008, IST-ENTHRONE phase 2 Integrated Project under Contract 507637.

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Digital Object Identifier 10.1109/TBC.2006.889111

adapt the video content rate by “transrating” the TV stream received from the DVB-T network and which is feeding the WLAN network.

The remainder of this paper is organized as follows: Section II presents an overview of the DVB and 802.11 WLAN. Section III presents the Rate Control Algorithm (RCA) used in the 802.11 WLAN. In Section III, we present some related works in the Cross-Layer paradigm. In Section IV, we detail our proposal cross-layer interworking. Section V presents the results of the conducted experimentations. Finally, we conclude and give some perspectives for future work in Section VI.

II. OVERVIEW OF DVB AND 802.11-BASED NETWORKS

Communication technology is evolving towards a world of pervasive (wireless/mobile) multimedia communication that is seamlessly and ubiquitously accessible at anytime, anywhere from any device. Central to this paradigm is the Internet Protocol (IP) which can integrate broadcast, cellular and IP-based networks into the Mobile Internet. The following two subsections present DVB and 802.11-based networks with their strength and limitation for delivery A/V services.

A. DVB Networks

The DVB project [3] is an industry-led consortium that regroups broadcasters, manufacturers, network operators, software developers, regulatory bodies and others in over 35 countries committed for designing global standards for the global delivery of digital television and data services. The DVB project defines different standards for transmission of digital television: the DVB-S (satellite transmission), DVB-C (cable transmission), and DVB-T (terrestrial transmission). DVB-T is the youngest of the three core DVB systems. Based on COFDM (Coded Orthogonal Frequency Divisional Multiplexing) and QPSK, 16 QAM and 64 QAM modulations, it is the most sophisticated and flexible digital terrestrial transmission system available today. The TV stream in the DVB is based on MPEG-2 audio/video compression and use the MPEG-2 transmission standards (MPEG-2 Transport Stream) to be deliver a broad range of push services in metropolitan areas.

Recently, the DVB project has defined the DVB-H (handheld) standards which adapt DVB-T to the specific requirements of handheld devices: power consumption efficiency, resilience against interference, and fading in mobile environment.

Current research efforts in the DVB project are oriented towards IPTV to define and develop appropriate standards for the delivery of DVB services over IP broadband networks. The goal being to extend the audience of TV services beyond the traditional coverage of broadcast networks, and thus get access to users attached to existing, and largely deployed, access networks such as WLANs.

In fact, the advent of high-speed bidirectional consumer broadband networks increases demands to provide DVB services over an IP network. To reach this goal, the DVB project should addresses different issues: How to interconnect the DVB and IP networks, which protocol to access and deliver the TV stream, and how to provide QoS in the IP network. Moreover, the DVB services conceived for unidirectional architecture (one-to-many) can and should be augmented with

interactivity to take advantage of bidirectional communication in IP networks.

B. 802.11-Based Network

With the recent standard 802.11a/g, the transmission rate of 802.11-based networks can reach up to 54 Mbit/s. This rate will be enhanced up to 540 Mbit/s with the forthcoming 802.11n specifications thanks to MIMO technology. However this rate highly depends of many different factors: number of connections, the distance with access point (AP), the interference levels, and the mobility pattern. The challenge is still to design effective scheduling mechanisms at MAC level to optimally deal with packets transmission over multiple interfaces in MIMO environment.

The 802.11 standards cover both the Physical Layer and the Medium Access Control (MAC) layer. They were conceived for the best effort traffic. Until today, no mature specifications (industry-endorsed ones) were considered for real time traffic such TV, VoD, VoIP, etc. Providing some mechanism to ensure the quality of service in the 802.11 standards will allow the QoS continuity in Access network. Recently, 802.11e specification was proposed to augment the 802.11 standards with some QoS mechanisms. 802.11e is devoted to (1) allow the access point to manage resources allocation (data rate and latency) (2) introduce different class of services in MAC layer (3) improve wireless bandwidth efficiency and packet overheads.

However, this specification is still not adopted by the industry. This is mainly due to the lack of QoS infrastructure and interoperability testing between products. In fact, the most popular standards that are implemented in commercial products are 802.11b/g/a and recently 802.11n. All these standards are compatible and interoperable. The different commercial offers that exist actually implement the CSMA/CA in MAC layer and use different modulation techniques and channel coding in the physical layer. The combinations of these modulations and channel coding produce different transmission bit-rates ranging from 1 Mbit/s up to 54 Mbit/s. It is important to notice that higher transmission rates come with lower resiliency against high Bit Error Rates (BERs) due to their sensitiveness to channel conditions: fading, attenuation, and interference. In contrast, lower data rates are more resilient and may sustain an important BER.

The wireless network driver’s interface provides a mean to control the transmission bit rate manually by statically switching the rate or by enabling an algorithm that changes the rate dynamically according to Channel States Information (CSI). Different Control Rate Algorithms (CRAs) are implemented by different card vendors and only few of them are largely disseminated. The following section gives more details on RCAs and describes certain major algorithms commonly implemented and employed in 802.11 wireless interfaces.

III. RATE CONTROL ALGORITHMS IN 802.11 WLAN

The 802.11’s physical layer provides a detailed specification concerning modulation and channel coding techniques used in 802.11. A particular coding transforms a data stream into a sequence of symbols which are modulated in the amplitude, phase, or frequency of an analog signal. Different modulation techniques and channel coding have been proposed in the

Bit-rate	802.11 standards	DSSS / OFDM	Modulation	Bits / symbol	Coding Rate	Mega symbols per second
1	b	DSSS	BPSK	1	1/11	11
2	b	DSSS	QPSK	2	1/11	11
5.5	b	DSSS	CCK	1	4/8	11
11	b	DSSS	CCK	2	4/8	11
6	a/g	OFDM	BPSK	1	1/2	12
9	a/g	OFDM	BPSK	1	3/4	12
12	a/g	OFDM	QPSK	2	1/2	12
18	a/g	OFDM	QPSK	2	3/4	12
24	a/g	OFDM	QAM-16	4	1/2	12
36	a/g	OFDM	QAM-16	4	3/4	12
48	a/g	OFDM	QAM-64	6	2/3	12
54	a/g	OFDM	QAM-64	6	3/4	12

DSSS: Direct Sequence Spread Spectrum.

OFDM: Orthogonal Frequency Division Multiplexing.

BPSK: Binary Phase Shift Keying.

QPSK: Quadrature Phase Shift Keying.

CCK: Complementary Code Keying.

QAM: Quadrature Amplitude Modulation.

Fig. 1. Theoretical 802.11 bit-rates and modulation.

802.11 standard to improve a transmission rate. Fig. 1 summarizes the transmission rate (physical layer rate) used by different 802.11 cards for different combination between modulations and channel coding.

The complex modulations that provide a higher bit-rate need a higher Signal to Noise Ratio (SNR) to ensure appropriate symbols decoding. This means that they are more sensitive to noise and their operation range is short in term of distance. The opposite case is completely valid, and the less complex modulations that produce lower rate and are more resilient to noise. Consequently, their operation range is more extended. It is better to use a complex modulation when the Station (STA) is close to the Access Point (AP) and simple one when the STA moves away from the AP.

The 802.11 physical layer specifications introduce multi-rate capabilities which allow the physical layer to provide different modulations, and one can just choose one of them. The standard defines also at the MAC-level the possibility to tune appropriately the rate transmission at the wireless network interface card (WNIC). However, neither methods nor techniques are defined to adapt the rate dynamically (during runtime). Indeed, the wireless card constructors implement their own proprietary Rate Control Algorithm (RCA) to change the rate dynamically according to channel state information represented by the SNR value. The RCA can be implemented as part of the driver software or directly in the hardware as part of the WNIC chipset.

Recently, the research community was more concerned to investigate these algorithms with through studies and analysis of existing RCAs. We briefly summarize some of them in what follows.

The authors in [4] identified three classes of RCAs: Statistics-based RCAs, SNR-based RCAs, and Hybrid RCAs. The Statistics-based RCAs maintain statistics information about the transmission conditions and the achievable throughput to adapt the rate consequently. In this class, three types of adaptation can be distinguished: (1) throughput-based (2) Frame Error Rate (FER) based (3) retry-based. The first one sends a small fraction of traffic at lower and higher rate during a certain time interval,

and then chooses the rate that achieves a higher throughput. The second type is based on FER that can be easily inferred thanks to the acknowledgements used by 802.11 MAC-layer; in fact, each packet is explicitly acknowledged by the receiver, if the sender doesn't receive it, the frame is considered as lost. The FER-based RCA uses the FER thresholds to change the rate. In the last type (retry-based), the RCA is based on tracking the number of transmission retries in consideration. If no ACK is received during a timeout, the frame is considered as lost. The number of retransmission attempts can be a-priori fixed. The retry based RCA decreases the bit-rate after a number of unsuccessful retransmissions, while it increases the rate after a number of successful transmissions without retry. The Auto Rate Fallback (ARF) presented in [5] is among the most widespread RCA that uses a retry-based technique. Overall, statistics-based RCAs provide a good performance in long-term, but they fail to cope with highly changing link conditions.

The second class identified is SNR-based RCA. Since the SNR influences the modulation technique which is directly related to the rate, it is interesting to adapt the rate directly according to the SNR perceived by receivers. However, in the 802.11 standard there are no fields in the frame header (RTS/CTS/ACK/DATA) that may allow the sender to obtain the SNR perceived by the receiver. RCAs based on SNR level can have unstable states: the rate can change continually due to the rapid change in channel conditions.

The RBAR [6] takes advantage of this technique by introducing a modification in the CTS, RTS, and data frames header to add rate information which is calculated at the receiver side based on the perceived SNR.

Finally, the last class which represents a combination between the two above-mentioned techniques tries to take advantages of each of them while minimizing their shortcomings.

The Madwifi project [7] which provides a Linux-based device driver for Atheros Wireless addresses the RCAs problem. The Madwifi driver implements three different RCAs: Onoe [7], AMRR [8], and Sample [9]. We notice that all these RCAs are statistics-based techniques with different level of responsiveness/aggressiveness.

The Onoe uses a credit which is incremented and decremented each one (1) second. The credit is incremented if 10% of transmitted frames need a retransmission. When the credit reaches the value of ten (10), the rate is enhanced and the credit gets reset. Otherwise the credit is decremented and the rate gets move to the lower rate. With this operation model, Onoe is very conservative and don't overreact in face of transient channel state fluctuations.

The second RCA proposed by Madwifi is the Adaptive Multi Rate Retry algorithm (AMRR) that puts in place a Binary Exponential Backoff (BEB implemented in AARF) in the driver. The AMRR employs a simple heuristic to adapt the length of the sampling period used to change the value of transmission rate and transmissions count parameters.

The last RCA implemented in Madwifi driver is the Sample Rate. It is based on the transmission time estimation. The algorithm transmits frames at one rate with keeping information about transmission conditions. If it experiences four successive failures it decreases the rate until it finds one that is able to

transmit a frame with an acceptable reliability. If there is no failure, Sample sends one packet for each group of ten packets at rate that it chooses randomly. This rate should not experience four successive failures. Then, it estimates the transmission time of this frame and compare it to average transmission time of precedent frames. If it is better, then it switches to a higher rate. Note that the average transmission time accounts for the successive backoff procedures upon successive transmission failures.

It is evident that all RCAs provide some rate adaptation. However, one issue that is not addressed on different RCAs presented before is the impact of this rate adaptation on application-level achieved throughput. For TCP traffic, it is obvious that the congestion control algorithm implemented in TCP detects the packet losses and then adapts the congestion window accordingly. Admittedly, there is a bad interference between RCAs mechanisms and the congestion control mechanisms inherent to TCP protocol variants. Still, it is not clear how UDP traffic (especially A/V traffic) is affected by abrupt through variation.

When the distance between the STA and the access point (AP) is important, the RCA implemented at WNIC decreases the transmission rate to lower level up to 1 or 2 Mbit/s for 802.11b. If the video throughput is transmitted with these rates, it monopolizes all the bandwidth and penalizes others streams active in the network. The rate fixed by RCA in the MAC layer is therefore important information which should be shared with the upper layers. It is indeed more efficient to coordinate the video encoding rate adaptation and link layer rate adaptation. This cooperation is known as cross-layer interaction. Next section introduces the cross-layer paradigm and presents some existing cross-layer adaptation works.

IV. CROSS-LAYER RELATED WORKS

Cross-layer design has been particularly used in wireless communication as an important paradigm to optimize the scarce wireless bandwidth. This design investigates approaches where different OSI layers may cooperate to improve the ability of applications to ensure certain objectives such as QoS guarantees, power saving, or user's preferences, etc. Recent years have witnessed many proposals of various solutions addressing the issues of cross-layer optimization. These solutions are completely independent, use different parameters at different layers, and even different communication systems between the layers.

In [10], a general block diagram of end-to-end QoS supporting video delivery in the network-centric cross-layer solution is proposed. Authors consider QoS mapping across different layers as one of the key issues for their general block diagram. However, application-level QoS is not considered. In the same manner, works in [11] provide an overview of a cross-layer paradigm where both physical and MAC layer knowledge is shared with higher layers. In [12], different error control and adaptation mechanisms available in several layers are evaluated for robust video transmission. Based on this evaluation, a new adaptive cross-layer protection strategy is proposed to enhance the robustness and efficiency of scalable video transmission. In [13], authors propose two techniques for adaptive video streaming over wireless. One of these techniques

is based on cross-layer solution where the application layer may access to link layer's context for an effective packet loss retransmission strategy.

The work in [14] presents a new cross-layer content delivery architecture that is capable of receiving information from network and adaptively tune transport parameters (e.g., bit rates and other QoS mechanisms according to the underlying network conditions). This service-aware IP transport architecture is composed of a dynamic content-level audiovisual object classification model; a reliable application-level framing protocol with fine-grained TCP-friendly rate control and adaptive unequal error protection; and a service-level QoS mapping / packet-tagging algorithm for seamless IP differentiated service delivery. The presented results demonstrate that breaking OSI layers isolation paradigm, by injecting content-level semantic and service-level requirements within the transport protocols, leads to efficient support of multimedia services.

Recently, the research community tried to classify the existing cross-layer optimizations and to define a state of art on this new paradigm. In [15] the authors formalize the cross-layer problem, discuss its challenges, and present a classification of cross-layer solutions.

The paper illustrates the relationship between layers and the challenge to find the best configuration that optimizes different metrics at different layers. In the same manner authors in [16] present an investigation in cross-layer design literature and proposes definitions. It also discusses the basic types of cross-layer design from the existing solutions. Finally some open challenges and new opportunities for cross-layer design are presented.

The majority of the disseminated investigations target for wireless communication, and especially MAC and PHY layers interactions. Almost all cross-layer works barely consider higher level interactions such as user/terminal/application-level QoS translation into well-proportioned QoS mechanisms.

The cross-layer interaction presented in this paper exploits at the application layer the information available at the MAC layer (BERs, CSI, and Signal Strength) to adapt dynamically the A/V services. We investigate joint channel and video bit-rate adaptation in response to any degradation. This cross-layer interaction between MAC and Application layers allows for adaptive service delivery. The next section presents the architecture of our system and details the cross-layer adaptation enabled in this system.

V. A CROSS-LAYER INTERWORKING OF DVB-T STREAM AND 802.11 WLAN

Our work represents a contribution toward a network convergence. We aim to interconnect the broadcast and 802.11 WLAN networks in order to provide IPTV (A/V) services to the mobile users. The objective is to individually adapt each video session between an Adaptation Gateway (AG) and then end user's terminal. First, the AG takes into account user's terminal capabilities and user's preferences (profile) that are made available to the AG at session invocation through the RTSP channel. This information is used at the AG to tailor (transcoding, transrating, etc.) the IPTV stream based on users requirements. Second, there is a continuous per-session adaptation (transrating) that

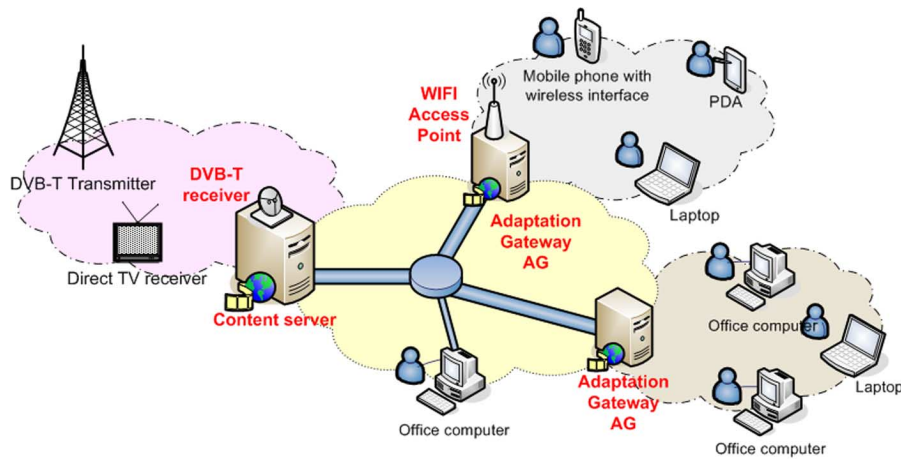


Fig. 2. Target architecture for DVB-T and 802.11-based WLAN interworking.

takes into account the cross-layer channel conditions experienced by the user of the session in question.

The IPTV system consists in delivering digital television over the IP protocol to different subscribing users. With IP protocol, the TV stream can be augmented by many others functionalities: pay-per-view, interactive and scheduled program retransmission. Moreover, the TV stream can be customized to accommodate the access network, terminal's heterogeneity, and user preferences (screen format, movie's languages . . .).

However, allowing IPTV reception through WLAN environment is still a very challenging. The QoS provided by the broadcasting network should be preserved and maintained while the user is moving creating further channel interference.

Fig. 2 describes the target architecture. The DVB receiver connected to the content server provides an MPEG-2 TV program (up to 2–6 Mbps). The TV streams are distributed to the adaptation gateways (AG) by a high-speed connection without any degradation or adaptation. Besides, information and legends (found in DVB tables) describing the content are as well being transmitted to AGs in order to be presented to users. The AGs are deployed at the border of the network within the access network. A Wireless Access Point (AP), embedded in AG, provides a wireless connection to heterogeneous users (phone, PDA, laptop, PC, etc.). Different adaptation strategies can be implemented on the AG keeping an acceptable media perceived QoS for the users. The deployed gateways are both media and channel aware. They enable the adaptation of video bit-rate to the actual channel bit-rate which may change in response to any degradation (signal strength quality, packet loss, etc.) caused mainly by the users' mobility and other external factors such as interference. These gateways represent a bridge to adapt the stream to suit network/terminal characteristics.

The cross-layer interaction proposed in our architecture is a combined both top-down and bottom-up interactions. In this respect, the session/presentation (i.e. RTP protocol) has an access to MAC information layer leading to the bottom-up interaction and it has access to video system layer (video coding layer) leading to the top-down interaction. Our Adaptation Gateway (AG) deployed at wireless access network plays the role of service and content provider. The AG is media-aware by managing

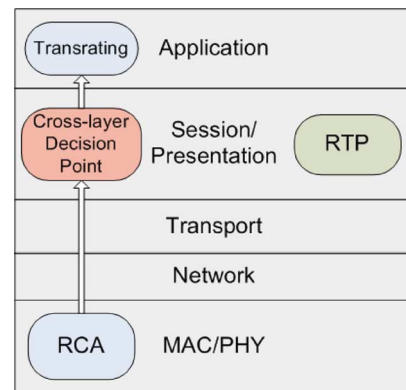


Fig. 3. Cross-layer decision point implemented at the AG.

the video bit-rate distributed to each user and channel-aware by accessing directly to the physical layer information (the signal strength, the rate fixed by the RCA, etc.).

As depicted in Fig. 3, the adaptation strategy implemented in the AG is completely controlled by the session/presentation layer. This layer implements a Cross-layer Decision Point (CDP) which takes relevant decision to enhance the A/V quality according to the surrounding information present at both lower and upper layers. In fact, CDP adapts the A/V throughput according to the link quality in term of actual rate used by the network access layer (MAC/Physical layers). Besides, the CDP determines the appropriate A/V bit-rate and it performs a transrating by changing video system properties to match the actual link quality condition and the instantaneous available bandwidth as closely as possible.

This adaptation is essential in the mobility context in which the PHY rate changes dynamically according to the RCA. When the STA is far from the AP, the RCA changes the signal modulation which results in a decrease in the physical transmission rate down to 1 or 2 Mbit/s. Obviously IPTV services (each of them having an average 2–4 Mbit/s) cannot be transmitted under this degradation. Indeed, the A/V packets are dropped in MAC queue before the transmission. These packet loss events affect considerably the user perceived video quality. Moreover, the

PHY Rate	UDP APP Rate	percentage
1Mbit/s	0,878Mbit/s	87%
2Mbit/s	1,725Mbit/s	86%
5,5Mbit/s	4,152Mbit/s	75%
6Mbit/s	4,765Mbit/s	79%
9Mbit/s	7,144Mbit/s	79%
11Mbit/s	7,514Mbit/s	68%
12Mbit/s	9,196Mbit/s	76%
18Mbit/s	14,67Mbit/s	81%
24Mbit/s	19,441Mbit/s	81%
36Mbit/s	28,166Mbit/s	78%
48Mbit/s	36,75Mbit/s	76%
54Mbit/s	40,814Mbit/s	75%

Fig. 4. Relationship between PHY and application rate experienced in 802.11-based network.

concurrent TCP streams which perform Windows-based adaptation according to the packet losses are seriously penalized with the UDP A/V stream monopolization.

It is important to notice that there is a difference between the physical layer rate and the application layer rate. This can be explained by packet data unit overhead added at each encapsulation layer. Fig. 4 illustrates the application rate average achieved by UDP traffic for each configuration of the physical rate. These experiments were performed in a close environment as we will explain it in the next section.

The adaptation policy implemented at the CDP recovers the actual rate used by the RCA before each video frame transmission. The estimated application rate according to the PHY rate is fairly shared between concurrent streams. In fact, the AG maintains a list of current IPTV services and other traffic delivered over the same wireless 802.11's interface. In other words, in order to figure out the best transrating for a given IPTV stream, the CDP takes into account other streams (and their respective bit rates) being transmitted in the same network.

By transrating the IPTV stream in real time according to the physical rate variation, the video bit-rate may decrease to avoid packet losses and may increase to better exploit the rate provided by the physical layer. More the video bit-rate decreases, more the video quality decreases. But the perceived video quality and consistency is preserved compared to quality degradation caused by random packet losses that usually lead to video freezing do to decoding dependencies between different packets.

VI. PERFORMANCE EVALUATION

The experimental topology proposed in this paper is based on two main components: the content server's which receives live-TV services from a DVB-T network and the Adaptation Gateway (AG) that performs a cross-layer interworking between the DVB and 802.11 networks. The DVB-T stream is received using a FRECOM DVB-T USB receiver [17]. The content server transmits the DVB Transport Stream (TS) Packet-Data Units (PDUs) over UDP to all AGs active in the network. If the underlying network is based on Ethernet (such as in our case),

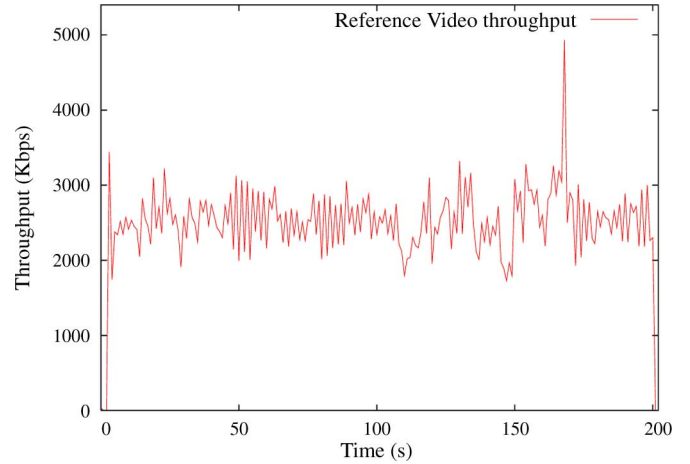


Fig. 5. Reference video throughput.

seven (7) PDUs (188 byte-long) are packed on to a single UDP packet. In our implementation, both the client and the AG are based on the open source project VideoLan (VLC) [18]. The video adaptation (i.e. transrating) is carried out using a VLC built-in FFMPEG [19] API.

The AG is composed of an Access Point (AP) that provides wireless network connection to end-users' equipments. The AP is based on a three-band (a/b/g) wireless card with an Atheros chipset. To adjust the link-layer transmission rate, we use the Madwifi Open Source Project.

Based on RCAs provided in Madwifi driver (Section III), an extension of VLC at the AG is proposed to enhance TV services delivery with channel awareness capabilities, and ultimately perform a joint (coordinated) channel and content adaptation. The AG demon is implemented in way to control both transrating libraries (VLC) and link-layer mechanisms (Madwifi) in order to allow for a higher mutual awareness between video content adaptation strategies and link transmission rate switching algorithms (i.e. RCA). In this manner, for each single streaming session the AG uses its current link-layer rate to figure out the optimal video throughput to be used.

In our experiment, we used an MPEG-2 sequence captured using DVB-T. This video reference sequence of 200 seconds is of 720*576-pixels size with a frame rate of 25 frames/seconds. The instantaneous reference video throughput is represented in Fig. 5.

Fig. 6 depicts the test-bed environment. The experiment duration is about 120 seconds, during which the wireless mobile user moves from the Adaptation Gateway (i.e. location A in the figure) to location (B), and then comes back to location (A) as illustrated. This mobility is intended to degrade link quality and to introduce location-varying network conditions. The underlying objective is to cause link-layer transmission rate variation in order to evaluate the performance of our cross-layer adaptation.

The test-bed is achieved in closed environment (laboratory), and the mobile user may be separated by up to three walls when it is at its maximum distance from the AP. Note that we perform many experiments in which the receiver moves with the same mobility pattern. In order to appropriately receive and display A/V streams, we use a wireless laptop with 1.7 GHz processing

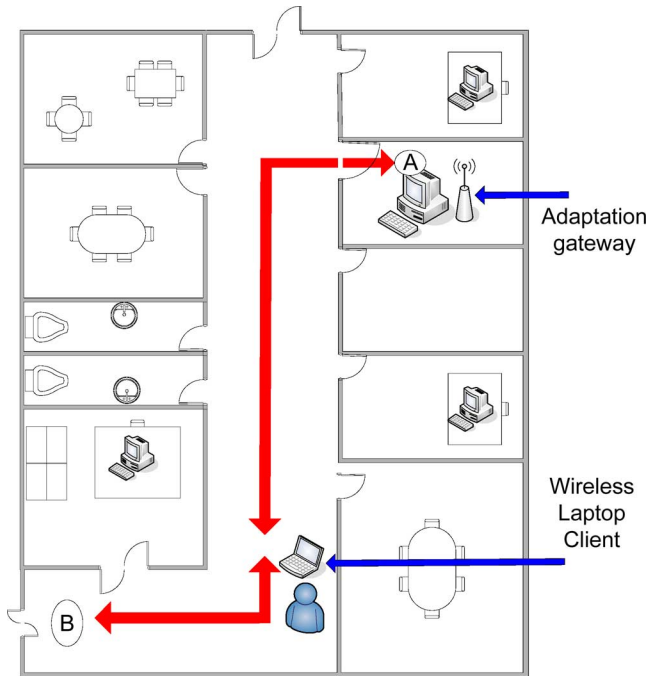


Fig. 6. Test-bed configuration topology.

capabilities and running a 0.8.4 version of VLC, and equipped with an Atheros-based WNIC.

To evaluate the performance gain that may be achieved by our system, we compare it with a conventional (without adaptation) streaming system. As mentioned, the Madwifi driver used by the atheros wireless chipset implements three distinct RCAs (Onoe, AMRR, and Sample) to control the rate at MAC layer. We can choose one particular RCA at the driver installation. Therefore, to control link layer rate, we used “Sample” RCA. Two scenarios are conducted:

- **Scenario 1:** reference sequence streaming with an original version of VLC without cross-layer adaptation.
- **Scenario 2:** reference sequence streaming with a modified version of VLC performing a cross-layer adaptation using the “Sample” RCA at the Madwifi driver.

In each scenario test, we recorded the video stream received by the client, the reported RTCP loss ratio, instantaneous video throughput during the test, and the link layer transmission rate variations pattern produced by the “Sample” RCA.

Fig. 7 and Fig. 8 show the packet loss ratio reported by the RTCP feedback and the video throughput estimated at the server, respectively. While Fig. 9 and Fig. 10 show the per-packet physical rate transmission in both scenarios 1 and 2. As result, in scenario 2, when using the cross-layer adaptation, the AG has lesser packets to send compared to scenario 1 without adaptation. As to be expected, these differences in the number of transmitted packets are matched with a difference in the video throughput, as illustrated in Fig. 8.

At the beginning of the experiment [0s, 30s], the client is close to the AG. At this point of time, the transmission signal strength is at its higher level and there are no MAC frame losses or frequent frame retransmissions. The “Sample” RCA uses the

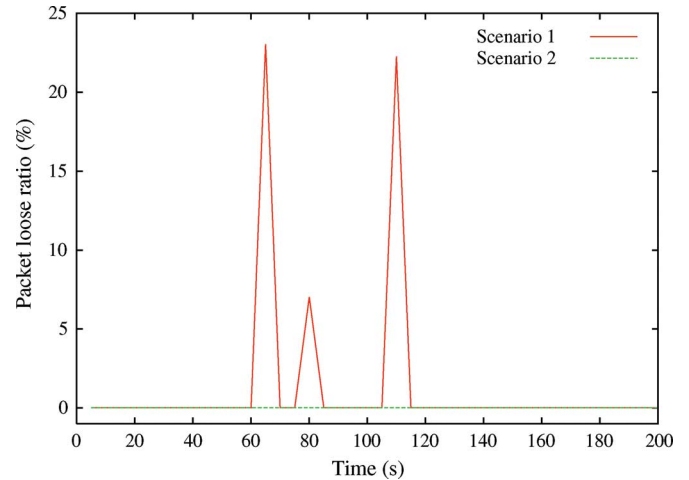


Fig. 7. Packet loss ratio comparison for scenarios 1 and 2.

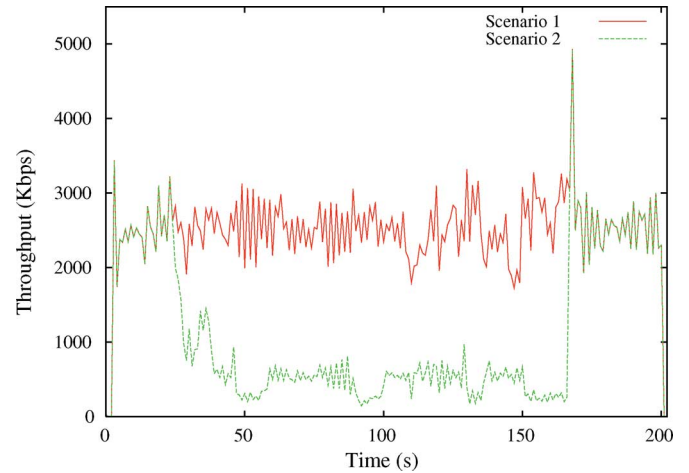


Fig. 8. Video throughput comparison for scenarios 1 and 2.

transmission rate 54 Mbit/s with both scenarios (1 and 2) as depicted in Fig. 9 and Fig. 10, respectively.

Afterward, the client begins to move and stays away from the AG [50s, 160s]. The signal strength shows a significant degradation and the frame error rate increases consequently. The “Sample” RCA which is based on transmission time estimation decreases the physical rate (Fig. 9 and Fig. 10) because the current transmission rate is taking too much times to transmit a single packet due to an increase in the mean retransmission attempts. Fig. 8 illustrates the response of our cross-layer adaptation to this rate changing in scenario 2. In fact, the scenario 2 adapts the video throughput while the scenario 1 maintains the same video throughput causing even higher packet loss rates (counterproductive behavior).

When the mobile client reaches the location B (in Fig. 6), the signal strength falls to a very low level causing RCA to use the lowest rate (1 or 2 Mbit/s) to maintain the transmission. Since scenario 1 is not aware of link quality degradation, we notice some spikes in packet loss measurements with a mean ratio reaching 20% (Fig. 7). In contrast, with the cross-layer adaptation in scenario 2, the video throughput is decreased using transrating.

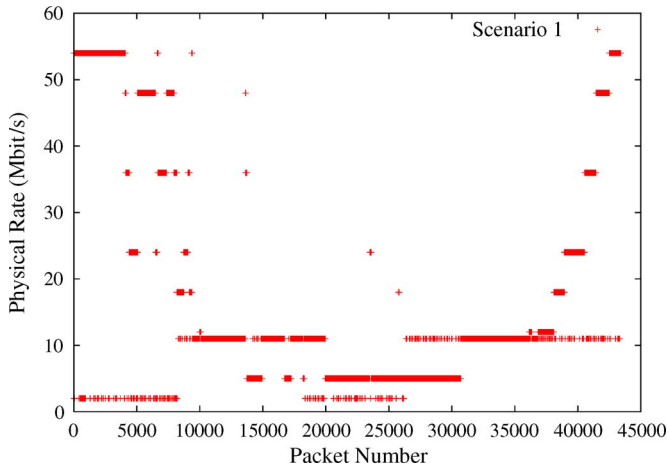


Fig. 9. Per-packet physical rate variation for scenario 1.

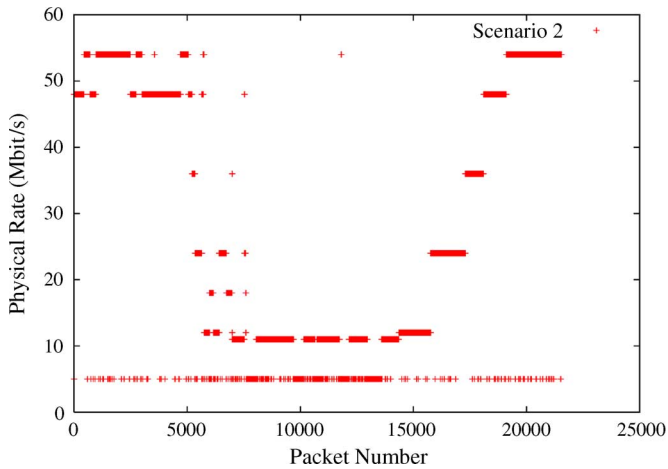


Fig. 10. Per-packet physical rate variation for scenario 2.

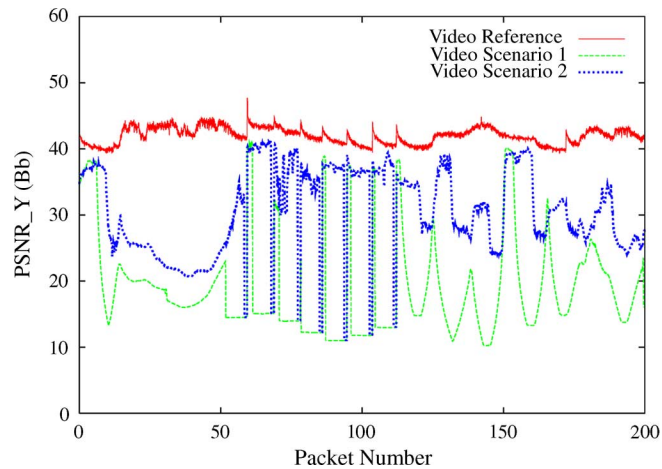


Fig. 11. PSNR comparison for scenarios 1 and 2.

The reduced video throughput can be perfectly accommodated by link-layer rate of 1 or 2 Mbit/s, which means no packet loss. Fig. 9 and Fig. 10 reveal, respectively, that in scenario 1 the physical rate decreases until 1 or 2 Mbit/s but in scenario 2 the lowest physical rate falls only to 5 Mbit/s. The reason is that the channel transmission is less saturated (low video throughput)



(a)



(b)



(c)

Fig. 12. Snapshot of the received and reference video quality for Frame #1642 at time 65s.

with the last scenario, and thus there are fewer collisions (transmission failure) and a short mean transmission time. All this will lead the Sample-RCA to further reduce the transmission rate with the scenario 1.

Finally, when the client returns back closer to the AG, the link quality is enhanced. The signal strength is improved and the AP needs less retransmission attempts with each frame. As a consequence, the “Sample” RCA increases the physical rate (Fig. 9 and Fig. 10) causing our cross-layer adaptation to enhance the video throughput (Fig. 8).

To further assess the user-perceived QoS gain induced by the cross-layer adaptation, we performed objective QoS measurement using PSNR (Peak Signal to Noise Ratio) metrics. The PSNR assessment estimates the reconstructed images quality compared to original ones. The Fig. 11 shows PSNR for scenario 1 and 2 between Frame #2000 and Frame #4000. These frames represent the time interval [40 s, 160 s] which is the lowest physical rate period.

Snapshots of the received video quality of Akyio MPEG-4 sequence stream transmitted under the same conditions for scenario 1 and scenario 2 for Frame #1642 (at time 65s) are given in Fig. 12(b) and (c) respectively. The original video quality is reported in Fig. 12(a). Clearly, our cross-layer optimization shows a significant perceived quality improvement when comparing scenario 1 and scenario 2. The gain is quantifiable in terms of loss ratio and perceived quality.

VII. CONCLUSION

The conception and deployment of a new industry-endorsed platform for the delivery of media over mobile networks may take several years before going from R&D stage to finally hit the market. It is primordial for main industrial actors to develop intermediary, cost effective, solutions towards achieving the convergence of broadcast and interactive-IP worlds to meet the current market demands.

In this paper, we proposed an Adaptation Gateway (AG) for a seamless cross-layer interworking between broadcasting network (DVB-T) and 802.11-based WLAN network for adaptive and interactive mobile TV service delivery. This gateway is media and channel aware by enabling a joint channel and video bit-rate adaptation in response to wireless link degradation (signal strength quality, packet loss, etc.). The AG is based on an emerging concept namely the cross-layer adaptation to allow the video throughput adaptation according to rate information provided by underlying network technologies. Our AG takes advantage of the rate variation capabilities provided by a recent wireless card to maintain better QoS for mobile users. The different experiments conducted validate our system and only proves the necessity of further cross-layer cooperation to achieve sustained QoS guarantees.

Future works will be focused on elaborating more sophisticated adaptation policies to take into account the multi-stream and multi-user environments. Moreover, we project to compare different RCAs implemented actually in order to select the most suitable configuration.

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