# SCALABLE RESOURCE MANAGEMENT FOR VIDEO STREAMING OVER IEEE802.11A/E

*Y. Andreopoulos[1], M. van der Schaar[1], Z. Hu[2], S. Heo[3], and S. Suh[3]*

[1]University of California Los Angeles
Dept. of Electrical Engineering
54-147 Eng. IV Building, 420 Westwood Plaza,
Los Angeles, CA 90095-1594, USA

[2]University of California Davis
Dept. of Electrical and Computer Engineering
2346 Ac. Surge III, One Shields Ave. 3129
Davis, CA, 95616, USA

[3]Samsung Electronics
Software Center, Systems R&D Laboratory,
416, Maetan-3Dong, Yeongtong-Gu,
Suwon-City, Gyeonggi-Do, 442 742, Korea

## ABSTRACT

Delay-constrained streaming of fully-scalable video over IEEE 802.11a/e wireless (WLANs) is of great interest for many emerging multimedia applications. In this paper, we consider the problem of video transmission over HCF[1] Controlled Channel Access (HCCA), which is part of the new medium access control (MAC) protocol of IEEE 802.11e. A cross-layer optimization across the MAC and application layers is used in order to exploit the features provided by the new HCCA standard, as well as by the versatility of new state-of-the-art scalable video coding algorithms. Under pre-determined delay constraints for streaming, the proposed cross-layer strategy leads to a larger number of stations being simultaneously admitted (without any loss in the video quality) than in systems that utilize application-layer only optimizations. At the same time, the fine-grain layering provided by the scalable bitstream facilitates prioritization and unequal retransmissions of packets at the MAC layer thereby enabling graceful quality degradation under channel-capacity limitations and delay constraints. The expected gains offered by the optimized solutions proposed in this paper are established through simulations.

## 1. INTRODUCTION

IEEE 802.11 WLANs [2] have emerged as a prevailing technology for the (indoor) broadband wireless access. Today, the deployed IEEE 802.11 networks can be considered as a wireless Ethernet, which supports only a best-effort service (not guaranteeing any service level to users/applications). For this reason, the IEEE 802.11 Working Group recently defined a new supplement (part "e") to the existing legacy Medium Access Control (MAC) sub-layer of the standard, in order to support QoS [3]. A new medium access method called Hybrid Coordination Function has been proposed in the 802.11e draft [3], which combines a contention channel access mechanism, referred to as Enhanced Distributed Channel Access (EDCA), and a polling-based channel access mechanism, referred to as HCF Controlled Channel Access (HCCA). EDCA and HCCA operate simultaneously.

Research issues of 802.11e HCF scheduling have recently started to gain some attention. Initial contributions [4] [5] were mainly concerned with the feasibility of the EDCA and HCCA mechanisms of HCF for multimedia transmission. These approaches perform optimization only at the application layer or at the MAC layer, and thereby do not achieve the significant gains offered by cross-layer optimization [7].

In order to accommodate delay and transmission requirements, we perform cross-layer optimization between the MAC and application layers and measure the benefits in terms of individual stations performance as well as the overall system performance. The following steps are involved in the proposed optimization:

- Unlike conventional wireless streaming solutions, where each video flow is admitted individually by the Admission Control Unit (ACU) co-located with the QoS-enhanced Access Point (QAP), the application-layer video flow is divided into *sub-flows* based on the delay requirements of individual video frames [7] (Section 2).
- Based on the delay requirements of each flow, the optimal transmission scheduling strategy is established in Section 3.1. It is shown that, under error-free transmission during the contention-free periods, our algorithm increases the number of admitted stations without any compromise in the video quality.
- The inherent prioritization and graceful degradation properties of scalable coding are utilized in Section 3.2 in order to provide an optimized framework that defines the maximum retry limit for each MAC service data unit (MSDU) in the video sub-flows, given the delay constraint and distortion impact for each sub-flow's transmission duration. This allows an already admitted application/sub-flow to continue its transmission even if the channel conditions worsen, without (significantly) compromising the video quality. This graceful degradation is extremely important for real-time video applications, where a renegotiation of the TSPEC parameters would have a disrupting effect on the video quality that is unacceptable for the end user.

In order to justify the proposed algorithms and methods, simulation results are presented in Section 4. Our conclusions are drawn in Section 5.

## 2. ENHANCED VIDEO STREAMING OVER IEEE 802.11E – THE SUB-FLOW CONCEPT

Admission control is one of the most essential components in IEEE 802.11e. This is due to the fact that, to ensure user satisfaction, it is essential that, once admitted, a video stream is guaranteed QoS for its lifetime. Thus, there is a need to control how many streams are admitted to the system and what should be the wireless resources allocated to each stream within the duration of each Service Interval (SI), denoted by $t_{\mathrm{SI}}$. For each flow $i$, the admission control unit can calculate the transmission opportunity (TXOP) duration required to service all the MSDUs within $t_{\mathrm{SI}}$ as [7] [8]:

$$t_{\mathrm{TXOP},i} = N_i'\left(L_i/R_i + T_{\mathrm{overhead},i}\right) \tag{1}$$

with $T_{\mathrm{overhead},i}$ the required overheads for MSDU acknowledgements and station polling, $L_i$ the nominal MSDU size, $R_i$ the minimum physical-layer rate, and $N_i' = \left\lceil g_i' \cdot t_{\mathrm{SI}}/L_i \right\rceil$ the number of MSDUs per SI, which can be calculated based on the effective bandwidth $g_i'$ of flow $i$ [7] [8]. Naturally, a necessary condition for non-violation of the initially-negotiated QoS requirements is that $R_i \geq g_i'$. For each new video flow $i$, we

[1] HCF: Hybrid Coordination Function

can express the admission control in terms of the TXOP duration for all the existing flows in the system as:

$$t_{\mathrm{TXOP},i} / t_{\mathrm{SI}} + \sum_{j=1}^{i-1} t_{\mathrm{TXOP},j} / t_{\mathrm{SI}} + t_{\mathrm{TXOP,other}} \leq (T - T_{\mathrm{CP}}) / T \quad (2)$$

where $t_{\mathrm{TXOP,other}}$ indicates the TXOP allocated to non-video traffic, $T$ is the beacon interval [3] and $T_{\mathrm{CP}}$ is the time reserved for the contention period, i.e. for EDCA traffic.

The admission control expressed by (2) is very useful, as it can be used for the construction of a round-robin, standard-compliant scheduler. In particular, normative behavior set by the IEEE 802.11e draft [3] requires that the ACU grants every flow $i$ the negotiated time $t_{\mathrm{TXOP},i}$. Hence, for every video flow, (1) and (2) can be used. The remaining unknown parameter is $t_{\mathrm{SI}}$, which, for a total of $n$ flows, is typically calculated as:

$$t_{\mathrm{SI}} = 0.5 \min\{d_1, \ldots, d_n\} \quad (3)$$

where $d_i$, $1 \leq i \leq n$ is the delay requirement for flow $i$. Notice that within the $n$ flows, several can be video flows, audio flows, or other delay-stringent multimedia. In addition, the factor $0.5$ is used to accommodate the jitter constraints demanded by the particular applications [4].

The aforementioned simple scheduling can be quite inefficient for real-time video streaming applications because the video traffic varies over time and consists of frames/packets with considerably varying sizes and different delay constraints. To improve the overall system utilization as well as the performance of the admitted stations, we introduced the sub-flow concept [7] [8] in which a video flow (bitstream) is divided into several sub-flows based on their delay constraints as well as based on the relative priority in terms of the overall distortion of the decoded video. The application layer enables each sub-flow of the video to interface with the MAC as a separate flow. Each sub-flow has a different priority (determined by its distortion impact) and delay constraint. A sub-flow has its own TSPEC parameters and is admitted independently by the ACU.

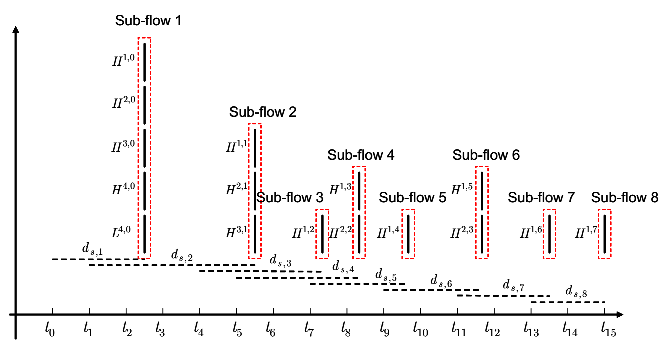


Figure 1. Sub-flows with different transmission durations due to additional delay permitted. Each $d_{s,i}$, $1 \leq i \leq 2^{D-1}$ corresponds to additional transmission time for sub-flow $i$. For all cases, we assume that an upper bound for the additional delay is set, denoted by $d_{\max}$, and we have: $\max\{d_{s,1}, \ldots d_{s,2^{D-1}}\} \leq d_{\max}$.

An example of the sub-flow based scheduling is given in Figure 1. We utilized a scalable MCTF-based video coder where, for each group-of-pictures (GOP), the temporal decomposition generates a hierarchy of high-pass (error) frames [6] denoted as $H^{t,j}$, with $t$ indicating the temporal decomposition level and $j$ the frame number within that level [6], and an "average" frame $L^{\max\{t\},0}$ [6]. Notice that the frames of the temporal decomposition are sorted in $N_s$ sub-flows according to their delay deadlines at the decoder [7] (which depend on the inverse MCTF that reconstructs the output video; $N_s = 8$ in our example). As a result, each sub-flow $i$ can have a varying transmission duration $d_{s,i}$ (Figure 1), which is upper-bounded by the video streaming delay deadline.

Our aim is to use the sub-flow mechanism to provide a joint application-MAC optimization that maximizes the number of admitted wireless stations while optimizing the video quality for each admitted station. Given the channel conditions, the ACU and the cooperating wireless stations have to determine for each application the number of sub-flows the application-layer can transmit, as well as their protection strategies (e.g. MAC retry limits per sub-flow), while maximizing the number of admitted wireless stations in the network.

## 3. Optimization of the number of Admitted Stations under Delay Constraints

Although the use of sub-flows permits the increase in the number of admitted stations in the HCCA traffic [7], if additional delay is permitted in the transmission of each sub-flow traffic, an optimal scheduling algorithm can yield further improvements. A visual example of such a case for one GOP of video data can be seen in Figure 1, where each increase in the transmission duration of each sub-flow $i$, $d_{s,i}$, provides the opportunity for traffic smoothing. In order to accommodate delay requirements, we have $\max\{d_{s,1}, \ldots d_{s,2^{D-1}}\} \leq d_{\max}$ with $d_{\max}$ the delay deadline set by the chosen streaming scenario.

### 3.1 Optimization of the Number of Admitted Stations

Each increase in the transmission duration of sub-flow $i$ is reflected by a change in $t_{\mathrm{TXOP},i}$. If we assume $N_{\mathrm{QSTA}}$ total stations, each of which has one video flow under HCCA, and each flow consists of $N_s$ subflows, we can introduce the average transmission-opportunity duration for each video flow as:

$$\overline{t_{\mathrm{TXOP}}} = 1 / N_s \cdot \sum_{i=1}^{N_s} t_{\mathrm{TXOP},i} . \quad (4)$$

Following the admission control expressed in (2), if we assume no residual HCCA traffic, i.e. $t_{\mathrm{TXOP,other}} = 0$, by replacing the average transmission-opportunity duration for each station by (4), we get the maximum number of admitted QoS-enhanced stations (QSTA) carrying video data as:

$$N_{\mathrm{QSTA}} = \left\lfloor t_{\mathrm{SI}} \cdot (1 - T_{\mathrm{CP}} \cdot T^{-1}) / \overline{t_{\mathrm{TXOP}}} \right\rfloor . \quad (5)$$

The last equation indicates that maximizing $N_{\mathrm{QSTA}}$ is equivalent to minimizing $\overline{t_{\mathrm{TXOP}}}$ since the other parameters are unaffected by changes in the transmission duration. As a result, by limiting the optimization to one GOP (since the video-flow traffic is periodic for each GOP [7]) the minimization problem now becomes:

**Primary problem:** $\{t^*_{s,1}, \ldots, t^*_{s,2^{D-1}}\} = \arg\min \sum_{i=1}^{2^{D-1}} g'_{s,i}$ . (6)

such that $\forall i : 1 \leq i \leq 2^{D-1}$ we have:

$$\sum_{j=1}^{i} t^*_{s,i} \leq \sum_{j=1}^{i} (t_{s,i} + d_{\max}) . \quad (7)$$

In (6) and (7), $\{t^*_{s,1}, \ldots, t^*_{s,2^{D-1}}\}$ are the optimal transmission durations corresponding to sub-flows $1 \leq i \leq 2^{D-1}$, $g'_{s,i}$ is the effective bandwidth defined by $g'_{s,i} = b_{s,i} / t_{s,i}$, with $b_{s,i}$ the size (in bits) of sub-flow $i$, and $t_{s,i}$ the original transmission duration of sub-flow $i$. Notice that this definition of $g'_{s,i}$ assumes that constant-bitrate (CBR) transmission occurs for the duration of sub-flow $i$ using a conventional leaky-bucket model [5] [7]. In order to facilitate the optimization process, the optimization problem stated in (6), (7) can be expressed in a dual form as:

**Dual problem:** $\{b^*_{s,1}, \ldots, b^*_{s,2^{D-1}}\} = \arg\min \sum_{i=1}^{2^{D-1}} b_{s,i} / t^{\max}_{s,i}$ . (8)

such that $\forall i : 1 \leq i \leq 2^{D-1}$ we have:

$$\sum_{j=1}^{i} b_{s,i}^{*} \geq \sum_{j=1}^{i} b_{s,i} \ . \tag{9}$$

In (8) and (9), $\{b_{s,1}^{*},\ldots,b_{s,2^{D-1}}^{*}\}$ are the optimal sub-flow sizes, and $t_{s,1}^{\max} = t_{s,1} + d_{\max}$, $t_{s,i}^{\max} = t_{s,i}$ for $2 \leq i \leq 2^{D-1}$ represent the maximum permissible transmission durations for each sub-flow.

Under CBR transmission for each sub-flow, the Primary and Dual problems stated above provide the same solution [8]. For example, by deriving the optimal sub-flow sizes we can establish the optimal transmission duration corresponding to the size of the first sub-flow as:

$$t_{s,1}^{*} = t_{s,1}^{\max} \cdot b_{s,1} \big/ b_{s,1}^{*} \tag{10}$$

and in the general case of sub-flow $i$ as:

$$t_{s,i}^{*} = \sum_{k=1}^{i} t_{s,k}^{\max} \cdot \sum_{k=1}^{i} b_{s,k} \cdot \left( \sum_{k=1}^{i} b_{s,k}^{*} \right)^{-1} \tag{11}$$

Nevertheless, one difference of practical significance is that the Dual problem facilitates the application of linear programming techniques, namely the simplex minimization, for the establishment of the optimal solution. This ensures optimality with low complexity, as the algorithm converges in a number of steps proportional to the total number of sub-flows, $N_s$. The simplex optimization scans through all the vertices of the $N_s$-dimensional simplex in order to establish the point corresponding to the minimum of (8). It is important to mention that, in order to formulate a bounded problem for this purpose, we need to impose an upper-bound to the maximum number of bits transmitted in the time interval corresponding to one GOP. Hence, we introduced an additional constraint to the problem given by:

$$\sum_{j=1}^{2^{D-1}-1} b_{s,j}^{*} \leq \sum_{j=1}^{2^{D-1}-1} \left( R_j \cdot t_{s,j}^{\max} \right) \tag{12}$$

which corresponds to the physical constraint that the maximum number of bits transmitted during the duration of one GOP together with the packetization overhead introduced at the various layers cannot exceed the mean amount of bits transmitted by the physical-layer during this time.

Finally, although the optimization problem is defined and solved for the duration of one GOP, if access to additional sub-flows from consecutive GOPs is possible (e.g. in the case of off-line encoding), they can be included in the optimization problem of (8), (9) following the same rationale. Experimental results with real video data utilizing the proposed optimization approach are presented in Section 4.

### 3.2 Packet Scheduling and Retransmissions

For the admitted sub-flows of a QSTA, the application and MAC layers can cooperate to improve the multimedia quality by adapting the retry limit. The previous studies [1] propose cross-layer strategies for 802.11WLANs that are not HCCA enabled and also they do not explicitly consider the delay bound set by the application for the various packets/flows. Here, the goal of packet scheduling and prioritized MAC retransmissions is to minimize the playback distortion for a video streaming session over an 802.11a/e HCCA WLAN, under delay constraints.

Due to limits imposed by link-adaptation to different physical-layer rates as well as delay constraints, the retransmission bound for the earlier-transmitted packets can be higher than the maximum retransmissions allowed for the remainder set of packets. Hence, under a scheme allowing for unequal video-packet retransmissions, a higher probability for correct reception can be provided to the first subsets of video packets. This motivates packet prioritization at the application layer depending on the video-data significance (incurred distortion due to losing the packet). Notice that this is readily achieved in MCTF-based coding, as the layered compressed information of each video frame can be encapsulated in MSDUs, which can be transmitted in decreasing significance.

Given the set of distortion-ordered MSDUs for each sub-flow $i$ as well as the transmission duration $t_{s,i}^{*}$, we establish which subset should be transmitted as well as the maximum permissible number of video-packet retransmissions in case of errors.

In this work we assume that the transmission channel is an independent, identically distributed error channel. Thus the channel causes errors independently in each MSDU frame and the error probability is the same for all MSDU frames with the same length at all times. Let $p_b(m)$ be the bit error probability in physical-layer mode $m$. Then, the error probability of an MSDU frame of size $L_i$ (belonging to sub-flow $i$) in physical-layer mode $m$ is a function of bit error probability $p_b(m)$ and is defined as:

$$p_e(m, L_i) = 1 - [1 - p_b(m)]^{L_i} \ . \tag{13}$$

Let $N_{\text{retry}}^{\max}(j)$ be the maximum number of retries of MSDU $j$ belonging to sub-flow $i$. Notice that the value of $N_{\text{retry}}^{\max}(j)$ depends on the position of the MSDU in the transmission queue (derived based on the criteria outlined before), as well as on the available transmission duration for the current sub-flow. The probability of unsuccessful transmission after $N_{\text{retry}}^{\max}(j)$ retransmissions is:

$$P_e(m, L_i, N_{\text{retry}}^{\max}(j)) = [\, p_e(m, L_i)]^{N_{\text{retry}}^{\max}(j)+1} \ . \tag{14}$$

In addition, based on $N_{\text{retry}}^{\max}(j)$, we can find the average number of transmissions for the $j$-th MSDU until we reach the retry limit [8]:

$$N_{\text{average}}(j) = \left( 1 - [\, p_e(m, L)]^{N_{\text{retry}}^{\max}(j)+1} \right) \big/ [1 - p_e(m, L)] \ . \tag{15}$$

The corresponding average time to transmit the MSDU using the guaranteed channel rate $g_i'$ for sub-flow $i$ is given by:

$$T_{\text{average}} = N_{\text{average}}(j) \left( L_i \big/ g_i' + T_{\text{ACK}} \right) . \tag{16}$$

where $T_{\text{ACK}}$ is the overhead for the transmission of the acknowledgment. Assuming that the maximum time before the MSDU expires is $T_{\max}$, we have:

$$T_{\text{average}} \leq T_{\max} \ . \tag{17}$$

Due to CBR transmission for the duration of each sub-flow, the MSDUs are evenly distributed with an interval $\alpha_i$ (which is the MSDU arrival interval for sub-flow $i$). Assuming that the transmission duration for sub-flow $i$ is $t_{s,i}^{*}$ (estimated by the optimization of Subsection 3.1), for the $j$-th MSDU frame of that sub-flow we have:

$$T_{\max} = t_{s,i}^{*} - \alpha_i \sum_{k=1}^{j-1} N_{\text{retry}}^{\text{actual}}(k) \tag{18}$$

where $N_{\text{retry}}^{\text{actual}}(k)$, $0 \leq N_{\text{retry}}^{\text{actual}}(k) \leq N_{\text{retry}}^{\max}(k)$, is the actual number of retries for each MSDU frame $k$ (that precedes MSDU $j$) until an acknowledgment has been received, or the maximum number of retries has been performed. Notice that $N_{\text{retry}}^{\text{actual}}(k)$ can be determined dynamically based on feedback from the MAC layer. The last equation can be used in conjunction with (14) and (17) to establish the bound for the maximum-allowable number of retries for the current MSDU $j$.

Notice that the estimated maximum number of retries determined by (18) can be negative, depending on whether we exceeded the available bandwidth for sub-flow $i$ or not. In such a case, the remaining MSDUs of the current sub-flow are simply discarded.

We outline the steps performed during the actual streaming process for each sub-flow $i$ in Table 1. Some of the last MSDUs of each sub-flow will not be transmitted whenever the channel condition deteriorates since the transmission duration (deadline) determined by the simplex optimization of the previous subsection do not take into account the retransmissions that will occur based on the algorithm of Table 1. Nevertheless, the use of a scalable video coding, as well as the prioritized generation and transmission of

MSDUs (per sub-flow) based on the expected distortion reduction, ensure that near-optimal adaptation of the video quality will occur based on the instantaneous channel capacity since the MSDUs with the most important video data will be transmitted first.

**Initialization:** Establish $p_b(m)$ based on the utilized physical layer mode. Calculate $p_e(m, L_i)$ from (13).
**For each MSDU frame** $j$ **:**

1. Establish $T_{\max}$ based on (18). Calculate $N_{\text{retry}}^{\max}(j)$ based on (15)-(17). Set $\text{current\_retries} = 0$ .
2. If $N_{\text{retry}}^{\max}(j) \geq 0$
   Set $\text{current\_ACK} = \text{FALSE}$ ; go to Step 3.
   else
   Terminate the current sub-flow transmission.
3. While $\text{current\_ACK} = \text{FALSE}$ & $\text{current\_retries} \leq N_{\text{retry}}^{\max}$
   Transmit the current MSDU.
   Set: $\text{current\_retries} \leftarrow \text{current\_retries}+1$
   Set $\text{current\_ACK}$ to $\text{TRUE}$ or $\text{FALSE}$ depending on MAC-layer feedback.
4. Set $N_{\text{retry}}^{\text{actual}}(j) = \text{current\_retries}$

Table 1. Transmission of MSDUs of each sub-flow $i$ .

## 4. EXPERIMENTAL RESULTS

We have evaluated the advantages offered by the proposed approach using the ns-2 simulator package of Ansel et al [5]. HCCA was used to stream a number of video flows generated by wavelet-based scalable video coding [7] [8] and EDCA was used for the remaining traffic within the contention period. Indicative results are presented in Table 2 with the settings $T = 100$ ms, $t_{\text{SI}} = 50$ ms , $T_{\text{CP}} = 60$ ms . The mean video rate was set to 2 Mbps. The optimization algorithm of Section 3.1 was used in the case of video transmission based on sub-flows in order to determine the application-layer optimal transmission duration for each sub-flow. Our results indicate that a significantly-higher number of stations could be admitted in the case of the sub-flow-based transmission. Notice that both cases are compliant to the IEEE 802.11e definition of the standard and both transmit the same amount of video data. Hence, the utilization of sub-flows under the proposed optimization offers a significant advantage.

| | Token Rate (kbps) | TXOP (ms) | Stations admitted |
|---|---|---|---|
| Global Flow | 4664 | 9.33 | 2 |
| Sub-flows 1~8 | 1385 (mean) | 3.35 (mean) | 5 |

Table 2. Measured token rates (based on leaky-bucket traffic smoothing), TXOPs and the total number of admitted stations based on ns-2 HCCA simulations, with $d_{\max} = 200$ ms .

| **Foreman** Packet Loss Rate (%) | Optimized Retransm. (dB) | Fixed Retransm. (dB) |
|---|---|---|
| 0.00 | 37.93 | 37.93 |
| 10.00 | 36.11 | 34.96 |
| 20.00 | 33.73 | 32.25 |
| **Coastguard** Packet Loss Rate (%) | Optimized Retransm. (dB) | Fixed Retransm. (dB) |
| 0.00 | 36.03 | 36.03 |
| 10.00 | 34.79 | 33.45 |
| 20.00 | 32.81 | 29.66 |

Table 3. Average PSNR (Y channel – 5 runs per method/loss rate) for two typical CIF sequences under various packet loss rates.

In order to investigate the effect of retransmissions in the proposed system, we have enabled a uniform bit-error loss model in our simulations. The proposed optimized retransmission policy was compared against a conventional retransmission policy which does not set the number of retransmissions based on the expected loss rate. The results are presented in Table 3. It can be seen that, across the various error rates, the proposed retransmission policy offers an average benefit of approximately 1.8 dB in PSNR versus the conventional retransmission policy, while the gains may be as high as 3.0 dB in some cases.

## 5. CONCLUSIONS

We have investigated cross-layer optimization strategies for HCCA-based video streaming. The proposed methods maximize the number of admitted stations by creating multiple sub-flows from one global video flow, each with its own traffic specification. In order to achieve an optimal scheduling policy, a low complex linear-programming solution is proposed, which effectively allocates the optimal transmission opportunity to each generated sub-flow in order to maximize the utilization of the wireless medium under the contention-free period. Besides the proposed method for optimization of the video traffic under HCCA transmission, the retry limit for each packet is adaptively modified in order to accommodate transmission under random packet losses. The proposed algorithm can be easily coupled with link adaptation mechanisms in order to provide efficient adaptation to dynamic network behavior [8].

## ACKNOWLEDGEMENTS

This work was supported in part by Samsung Corporation, NSF CAREER, and UC Micro. The authors would like to thank Dr. Sai Shankar N. (Qualcomm Inc.) for technical discussions related to the topic of this paper.

## REFERENCES

[1] D. Majumdar, G. Sachs, I. V. Kozintsev, K. Ramchandran, ”Multicast and Unicast Real-Time Video Streaming Over Wireless LANs,” *IEEE Trans. on Circ. Syst. for Video Technol.*, vol. 12, no.6, pp. 524-534, June 2002.

[2] IEEE Std 802.11-1999, *Part 11: Wireless LAN MAC and PHY specifications,* Reference number ISO/IEC 8802-11:1999(E), IEEE Std 802.11, 1999 edition, 1999.

[3] IEEE 802.11e/D5.0, Draft Supplement to Part 11: Wireless Medium Access Control (MAC) and physical layer (PHY) specifications: Medium Access Control (MAC) Enhancements for Quality of Service (QoS), June 2003.

[4] A. Grilo, M. Macedo, and M. Nunes, “A scheduling algorithm for QoS support in IEEE 802.11E networks,” *IEEE Wireless Commun. Mag.*, vol. 10, no. 3, pp. 36-43, June 2003.

[5] P. Ansel, Q. Ni, and T. Turletti. "An Efficient Scheduling Scheme for IEEE 802.11E". *Proc. IEEE Workshop on Model. and Opt. in Mob., Ad-Hoc and Wireless Net.* (WiOpt 2004), Cambridge, UK, March 2004.

[6] J.-R. Ohm, "Advances in scalable video coding," *Proc. of the IEEE*, vol. 93, pp. 42-56, 2005

[7] S. Shankar, Z. Hu, and M. van der Schaar, “Cross-layer optimized transmission of wavelet video over IEEE 802.11a/e WLANs,” *Proc. Packet Video Workshop*, April 2004.

[8] M. van der Schaar, Y. Andreopoulos, and Z. Hu, “Optimized Scalable Video Streaming over IEEE802.11a/e HCCA Wireless Networks under Delay Constraints,” *IEEE Trans. Mobile Computing*, to appear.