High-Performance H.264/SVC Video Communications in 802.11e Ad Hoc Networks

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Abstract. This work focuses on improving the performance of video communications based on the recently developed H.264 Scalable Video Coding (SVC) standard over 802.11e wireless networks. The H.264/SVC standard is particularly suitable for wireless communications because of its compression efficiency and the ability to adapt to different network scenarios. The adoption of the H.264/SVC codec in 802.11 networks poses however some issues. In particular, since pictures are encoded into several small video units, the overhead imposed by the 802.11 contention-based channel access mechanism might be large. Thus, the strategy employed for the packetization of the video data plays a key role in determining the performance of the network. This work proposes two network adaptation strategies for H.264/SVC video to efficiently use the QoS-enabled 802.11e extension of the 802.11 standard by designing a scheme for joint optimization of video data aggregation and unequal error protection. Simulations of video transmissions in a realistic home networking scenario characterized by direct communications between devices in ad hoc mode show that the proposed strategies reduce the packet loss rate and significantly improve the quality of the communication with PSNR gains up to about 2 dB. Moreover, the performance of the proposed low-complexity strategy is close to that of the optimal, high-complexity, strategy.

1 Introduction

Scalable video coding is increasingly used in multimedia communications since the recently developed H.264 scalable video coding (SVC) standard [5] achieves nearly the same compression efficiency of the state-of-the-art non-scalable H.264 advanced video coding (AVC) standard. A scalable video stream consists of a low-bandwidth substream (base layer) which represents a low-quality version of the video (e.g., low temporal o spatial resolution) and one or more substreams (enhancement layers) which, together with the base layer, allow the reconstruction of the full quality video. Scalability allows the transmission of the same encoded bitstream over different types of networks to client devices with different processing capabilities. Enhancement layers, in fact, can be dropped in case of network congestion or if the client is unable to decode them due to, e.g., processing power constraints. Therefore, scalability is particularly valuable in communications over wireless networks, where neither constant link quality nor minimum bandwidth can be easily guaranteed due the intrinsic unreliability of the medium.

The increasing popularity of multimedia-capable devices equipped with an 802.11 wireless network interface is also fostering the development of more and more efficient wireless video communication schemes. Moreover, QoS-enabled extensions such as the 802.11e [1] can be exploited in conjunction with the unequal perceptual importance of multimedia data to design error protection

schemes specifically for video. For instance, Ksentini [6] exploited the different access categories provided by the 802.11e protocol to grant better protection to the most important parts of AVC streams. More recently, the work in [8] showed that, given the same error protection scheme, the SVC codec provides better video quality than AVC since it better adapts to different network conditions.
SVC communications over 802.11-based networks raise however some new peculiar issues with respect to traditional AVC communications due to the high number of channel accesses they require. Such issues are especially evident in ad hoc scenarios, where the absence of a centralized infrastructure does not allow to coordinate and optimize channel usage. The decomposition of each picture in base and enhancement information results in fact in a large number of packets offered to the network in the common case that each information unit is transmitted as a new packet, as recommended by the RFC standard [3]. Transmitting a high number of small packets on a 802.11 network performs worse than transmitting a small number of large packets due to the overhead associated with channel access contentions [11]. This issue also affects VoIP applications, limiting the number and the quality of concurrent communications that the network can sustain.

While much attention has been devoted to the case of VoIP communications (see, e.g., [10]),video communication has received less attention. In this paper we propose and test two strategies which mitigate the channel contention overhead by aggregating multiple video information into a single packet while taking into account the varying perceptual importance of video data. The performance of the proposed strategies is evaluated by simulation in an 802.11e network scenario using both network and application level metrics. To the best of our knowledge, no previous work has explored the impact of aggregating SVC video data in terms of application level metrics such as the perceived video quality.

The paper is organized as follows. Sec. 2 provides the technical background, followed, in Sec. 3, by the proposal of possible strategies to improve the communication performance. After simulation setup (Sec. 4), results are reported in Sec. 5. Conclusions are drawn in Sec. 6.

2 Background

2.1 The H.264 Scalable Video Coding Standard

The H.264/AVC standard splits the functionalities of the encoder between the Video Coding Layer and the Network Abstraction Layer (NAL) [7] components. The former layer encompasses all the encoder core functionalities such as macroblock encoding. The latter layer facilitates the transport of video over packet networks by encapsulating each piece of encoded data into an independently decodable transport unit known as NAL Unit (NALU). Each NALU is prefixed by a header which specifies the type of data that the NALU contains, e.g., picture slices, parameter sets, etc. As in other video compression standards, the dependencies among the pictures are constrained within the so called *Group of Pictures* (GOP). Figure 1 depicts a typical AVC GOP: each box represents a NALU, the letter inside the box the picture type (Intra, Predictively or Bipredictively coded), the subscript number display order of the picture and the arrows show the decoding dependencies.

The H.264/SVC amendment extends H.264/AVC with temporal, spatial and fidelity scalability options, encoding a video as an independently decodable base layer and one or more enhancement layers. The header of an SVC NALU is extended with extra fields, such as the Temporal Index (TID) field, which provide information about the type of the enhancement information and their level

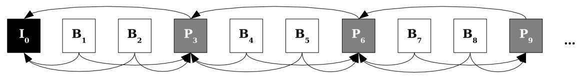

Fig. 1. A typical H.264/AVC GOP structure.

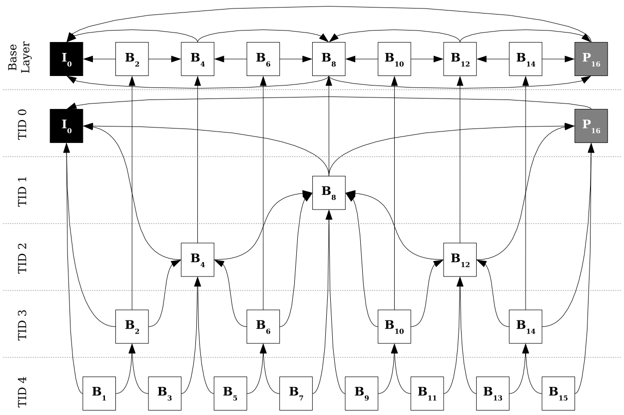

Fig. 2. GOP structure of the H.264/SVC encoding scheme used in this work.

in the decoding hierarchy. Figure 2 depicts the GOP structure of a typical H.264/SVC encoding scheme which encompasses the AVC-compatible base layer, one spatial and one temporal SVC enhancement layers. Each GOP is 16 frames long and every 32 frames a picture is Intra-coded, therefore the video encoding scheme is $I_0B \ldots BP_{16}B \ldots BI_{32}$.

From Fig. 1 and 2 it is clear that the SVC codec encodes each GOP in a higher number of NALUs and employs a more complex prediction scheme than AVC. The base layer is used to predict some enhancement NALUs while the other NALUs with a higher TID value depends on those with lower TID value as indicated by the arrows. In case a NALU is lost, different error propagation paths are possible. The loss of any base layer NALU generates distortion which affects both the base and the enhancement layers. The loss of an enhancement NALU, instead, generates distortion which affects the enhancement layers with a higher TID index only. Therefore, an efficient protection strategy might protect each NALU according to its potential to propagate distortion to other pictures.

2.2 The 802.11 and 802.11e Communication Standards

The 802.11 standard supports unstructured communications by means of the Distributed Coordination Function (DCF), which is based on a Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) channel access mechanism. At each host, packets expecting transmission are enqueued whilst the channel is busy. When the medium becomes idle, the host waits for a time interval called

DCF Inter Frame Space (DIFS) and, if the medium is still idle, the hosts defers the transmission for the so called backoff interval, i.e., an additional randomly chosen number of transmission slots. Then, if no transmission on the channel is detected, the host transmits the packet and waits for an acknowledgment from the destination. Should noise on the channel or a packet collision make the frame undecodable, the sender would receive no acknowledgment. Thus, the transmitter starts again the transmission of the packet increasing the contention window (CW), which leads, on average, to longer transmission delays.

The recently standardized IEEE 802.11e amendment [1] introduces the Enhanced Distributed Channel Access (EDCA) mechanism which extends the DCF mechanism with QoS support. For each host station, four distinct transmission queues known as Access Categories (ACs) are introduced in place of the unique queue offered by the 802.11 standard. The CW and DIFS values differ from queue to queue, resulting in high priority and low priority queues characterized by unequal chances of getting access to the channel. As a result, packets in high priority queues are elected for transmission before others, resulting in an effective intra and inter hosts traffic prioritization mechanism.

The efficiency of the contention-based DCF/EDCA mechanisms decrease when the number of channel accesses increases due to, e.g., a higher number of packets to send or an increase of the number of hosts participating in the network, since that packets have to wait in queue more time before being transmitted. Moreover, a high number of channel accesses by multiple hosts leads to a higher collision probability and higher packet loss rates.

Previous work showed that the DCF/EDCA performance can be significantly improved by reducing the number of accesses to the channel, e.g., by transmitting many data units as a single large packet. The 802.11 MAC mechanism is indeed provided with a mechanism which can aggregate small packets in the MAC queue. This mechanism is however designed for generic data and does not take into account application layer constraints such as, for example, the maximum allowed delay. Moreover, such mechanism cannot discriminate among the different types of multimedia streams nor it does consider the different perceptual importance of the various parts of a multimedia stream.

3 Problem Analysis and Solution

Generally, a video sequence encoded in SVC format requires the transmission of a higher number of RTP packets compared to a non-scalable AVC sequence. According to the SVC scheme depicted in Figure 2, every odd-numbered picture (namely the first, the third, etc.) is encoded as one base and one enhancement NALU, that is, on average, each two pictures three NALUs are generated. On the contrary, in the case of the AVC codec each picture can be encoded a single NALU (Figure 1). When the encoded video is transmitted over an IP network, each NALU is encapsulated into a single RTP packet (unless fragmentation is required) according to the RFC draft [3]. The higher number of NALUs of an SVC video thus implies the transmission of a higher number of packets with respect to a non scalable AVC video.

In this work a set of well known test video sequences, each 9$\sim$10s long, is encoded using the AVC JM [12] and the SVC JSVM [13] reference encoders provided by the ISO/MPEG Joint Video Team. Table 1 reports the bitrate, the encoding PSNR, the number and the average size of the encoded VCL NALUs as well as the number and the average size of the resulting RTP packets. A sample of the RTP packet size distribution for the Foreman sequence is also shown in Figure 3 for

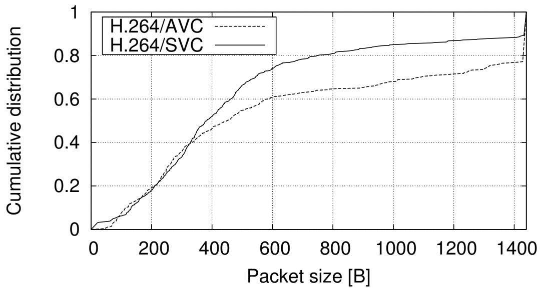

Fig. 3. Cumulative packet size distribution for the Foreman sequence encoded in H.264/AVC and H.264/SVC format. The transmission of an SVC video implies the transmission of a higher number of smaller packets with respect to the AVC video.

	H.264/AVC						H.264/SVC					
Sequence	PSNR [dB]	Bitrate [kb/s]	# of NALUs	NALU size	# RTP pkts.	RTP pkt. size	PSNR [dB]	Bitrate [kb/s]	# of NALUs	NALU size	# RTP pkts.	RTP pkt. size
Coastguard	30.57	326	298	1361	499	825	30.28	337	450	934	600	703
Football	31.34	615	259	2562	604	1111	31.07	571	390	1583	724	867
Foreman	33.29	205	298	854	386	672	33.13	212	450	586	522	521
Soccer	32.47	302	298	1259	425	895	32.98	331	390	915	476	766
Tempete	30.63	365	259	1523	432	926	30.46	377	390	1045	536	775

Table 1. Characteristics of the test video sequences encoded in H.264/AVC and H.264/SVC format.

both the AVC and SVC formats. The figures confirm that an SVC video communication implies the transmission of a higher number of smaller RTP packets with respect to its AVC counterpart, with the drawbacks highlighted in Section 2.2.

In this work we propose and test two different algorithms to reduce the number of channel accesses required by an SVC communication. Both algorithms aggregate multiple NALUs into a single RTP packet on a GOP basis, according to the RFC draft [3], therefore creating a standard-compliant packet flow. The I-type and P-type NALUs usually exceed in size the network MTU, therefore they cannot be aggregated together. On the contrary, the B-type enhancement NALUs with high TID index are good candidates for aggregation since they are copious and their size is much lower than the MTU.

The first proposed algorithm is referred to as *TID-based* in the rest of the paper and aggregates NALUs with identical TID in decoding order unless the network MTU (or a desired RTP packet size threshold) is exceeded. The number of NALUs which can be encapsulated into a single packet is highly variable and depends on the MTU, on the threshold imposed on the RTP packets size and on the video sequence. With reference to Figure 2, the proposed strategy would aggregate NALUs B_4 and B_{12} into a single RTP packet, B_2, B_6, B_{10} and B_{14} into a single packet and $B_1 \ldots B_{15}$ together. It is however unlikely that all the enhancement NALUs with the same TID can be aggregated into the same RTP packet given the MTU and the constraint on the maximum packet

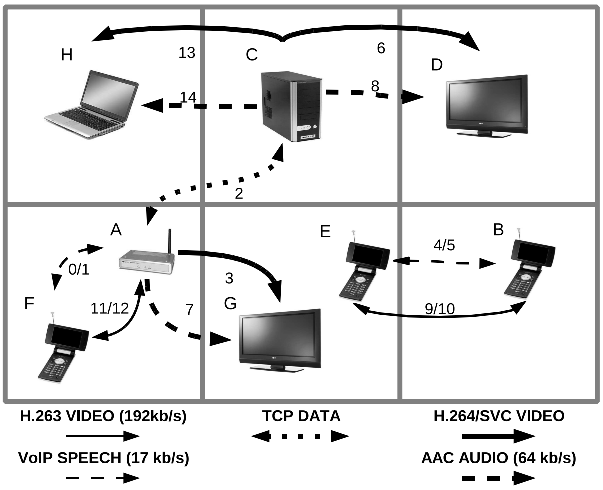

Fig. 4. The simulated network topology.

size. For example, NALUs B_2 and B_6 will be aggregated in a single packet as well as NALUs B_{10} and B_{14}, depending on their size. After aggregation, existing prioritization schemes for SVC video based on the TID value [8] can be easily applied.

The second algorithm is referred to as *Optimal* and aggregates NALUs regardless of the TID. The algorithm is optimal in the sense that, by means of an exhaustive search, it finds the combination of NALUs that minimizes the number of packets offered to the network. Therefore its computational complexity is high and in this work is used mainly as a reference for the *TID-based* algorithm. Since the *Optimal* strategy may aggregate NALUs with different TID into a single packet, the issue of determining the perceptual importance of such packets arises. In this work the importance $T(p)$ of a packet p is determined as shown in Eq. (1):

$$T(p) = \frac{\sum_i T(i) \cdot S(i)}{\sum_i S(i)}. \tag{1}$$

$T(p)$ is the weighted average of the TID values $T(i)$ of each packet i, the weights being the packet sizes $S(i)$, excluding the RTP header overhead. The TID of base layer NALUs is assumed to be equal to -1, thus packets transporting perceptually important NALUs will have their $T(p)$ value close to zero if not negative. Clearly, lower TID values represents higher importance. For example, a 1200-byte packet which includes an 800-byte base layer NALU and a 400-byte enhancement NALU with TID equal to 3, has a $T(p)$ value equal to 0.33. The $T(p)$ values are computed for each packet, and then the same traffic prioritization approach used for the TID-based strategy can be applied.

4 Simulation Setup

The two proposed NALU aggregation strategies are evaluated using the *ns* [4] network simulator. Figure 4 illustrates the network setup, which represents a typical domestic environment where different types of hosts, each located in a different room of a building, compete for the access to the channel. Host A is the gateway which provides internet access to all the other hosts in the building. Host G is a TV box which receives a multimedia stream composed of an SVC video and an AAC audio substreams from a content provider located in the internet. Host F is a videoconferencing device which communicates with a remote host located in the internet by sending and receiving VoIP traffic and low bitrate H.263 video. Hosts B and E are videophones which communicate among themselves, each generating traffic whose characteristics, in type and bandwidth, are similar to the ones of node F. Host C is a PC which exchanges data with a host on the internet via a TCP connection, and it additionally acts as a domestic media server which streams AAC audio and SVC video to hosts D and H.

In such a scenario data flows with different bandwidth and delay requirements coexist. For example, a VoIP call requires limited bandwidth, although it loads the network with a high number of small packet which have tight maximum delay and jitter requirements. Videoconferencing traffic demands more bandwidth than VoIP and, similarly, requires timely packet delivery. Maximum delay requirements for streaming of pre-recorded contents are less stringent, albeit a minimum bandwidth is required to ensure a smooth playback. Therefore, the traffic which loads the network is categorized in four classes with different QoS requirements. The highest importance Class A encompasses the four VoIP streams. Class B includes the four H.263 video streams and the 50% perceptually most important traffic, i.e., approximately all the base layer of the three SVC flows. Class C encompasses the three AAC streams and the 25% of the SVC enhancement traffic, i.e., the part with low TID values. Finally, the background TCP traffic and the remaining SVC enhancement traffic are assigned to the lowest importance Class D. The four traffic classes defined in the 802.11e standard are exploited by mapping Class A, B, C and D to AC3, AC2, AC1 and AC0, respectively.

In our simulations the 802.11e extension for *ns* developed by the Berlin Technische Universität [2] is used and the bandwidth of the channel is set to 11 Mb/s. The 802.11e MAC is modified adding a timeout mechanism to the transmission queues which drops a packet after 0.5 seconds of stay in the queue, as recommended by the 802.11 standard. Each SVC bitstream is fed to a RTP packetizer which implements the two NALU aggregation algorithms described in Section 3. Two sets of simulations evaluate the performance under different maximum RTP packet size constraints (750 and 1350 bytes). The same traffic trace is used to generate data flows #3, #6 and #13, namely, the three SVC flows. The video sequences are decoded using a frame copy error concealment technique, then they are visually inspected and PSNR is computed. Both error free and noisy channel conditions ($2.5 \cdot 10^{-5}$ byte error rate at the physical level) are simulated.

5 Results

Table 2 shows the performance of the two NALU aggregation strategies (*TID-based* and *Optimal*) presented in Section 3 both in error-free and noisy channel conditions (respectively, upper and lower half of the table). The table reports the number of SVC packets offered to the network, the byte loss rate (BLR) for the video flow at the application level, the packet triptime and the PSNR of the received video (flow #3). The number of packets offered to the network is a function

	Sequence	No aggregation				Max pkt. size [B]	TID-based strategy				Optimal strategy			
		# RTP pkts.	BLR [%]	Trip time [s]	PSNR [dB]		# RTP pkts.	BLR [%]	Trip time [s]	PSNR [dB]	# RTP pkts.	BLR [%]	Trip time [s]	PSNR [dB]
Error-free channel	Coastguard	617	9.91	0.65	**27.94**	750	556	7.45	0.62	**28.14**	448	2.60	0.59	**29.29**
						1350	491	2.88	0.49	**28.62**	359	1.08	0.47	**29.93**
	Football	724	28.09	0.86	**21.70**	750	720	28.67	0.86	**21.56**	647	27.02	0.84	**22.82**
						1350	704	28.11	0.85	**21.82**	601	27.39	0.87	**23.36**
	Foreman	522	5.45	0.55	**32.15**	750	447	1.51	0.39	**32.76**	338	0.08	0.31	**33.10**
						1350	382	0.03	0.22	**33.11**	237	0.00	0.17	**33.13**
	Soccer	477	8.41	0.60	**30.64**	750	472	7.62	0.58	**30.82**	441	5.72	0.59	**31.44**
						1350	416	2.88	0.47	**32.08**	322	1.03	0.43	**32.65**
	Tempete	536	12.22	0.64	**28.36**	750	523	10.45	0.62	**28.98**	469	9.20	0.65	**28.80**
						1350	453	6.17	0.57	**28.10**	342	1.38	0.47	**29.99**
Noisy channel	Coastguard	617	17.69	0.82	**26.22**	750	556	15.51	0.82	**26.06**	448	10.61	0.80	**27.92**
						1350	491	11.83	0.67	**26.46**	359	6.81	0.75	**28.61**
	Football	724	36.70	1.07	**18.84**	750	720	37.19	1.09	**18.70**	647	34.99	1.07	**20.36**
						1350	704	37.21	1.06	**19.33**	601	36.82	1.11	**21.17**
	Foreman	522	9.99	0.70	**31.22**	750	447	4.65	0.59	**31.95**	338	2.02	0.51	**32.57**
						1350	382	0.54	0.41	**32.93**	237	0.05	0.27	**33.11**
	Soccer	477	13.98	0.76	**28.09**	750	472	12.73	0.73	**28.76**	441	11.90	0.76	**29.09**
						1350	416	10.36	0.68	**27.96**	322	5.37	0.69	**31.21**
	Tempete	536	19.28	0.82	**25.02**	750	523	18.56	0.80	**23.91**	469	16.05	0.82	**26.30**
						1350	453	14.49	0.71	**23.40**	342	10.09	0.71	**27.64**

Table 2. Performance of the two NALU aggregation strategies.

of the considered aggregation strategy, of the maximum RTP packet size constraint and of the characteristics of the video sequence (see Table 1). The Foreman sequence, for example, is made of small-size NALUs, hence many NALUs can be encapsulated into one RTP packet. Thus, the number of accesses to the channel is noticeably reduced even if the maximum RTP packet size is set to half the MTU (750 bytes). The opposite considerations hold for sequences with large NALUs such as Football: multiple NALUs can be fitted into a single RTP packet only in a few cases and only if the threshold on the maximum RTP packet size is set close to the MTU. Obviously, the *Optimal* strategy is more effective than the *TID-based* strategy in reducing the number of channel accesses, which drops from 522 to 237 in the case of the Foreman sequence.

Simulations show that the byte loss rate and the packet triptime decrease when the number of accesses to the channel is reduced, both in error-free and noisy channel conditions. For instance, the Coastguard sequence shows 8% BLR reduction with the *TID-based* strategy and 9% with the *Optimal* strategy in error-free channel conditions. Such BLR reductions are due to the increased number of correctly received Class D packets, that is the lowest priority traffic class which encompasses mainly enhancement NALUs with TID equal to four. When the number of packets offered to the network decreases, the transmission queues become shorter and therefore less packets are dropped due to timeout expiration or queue saturation. Shorter queues imply lower triptimes, as

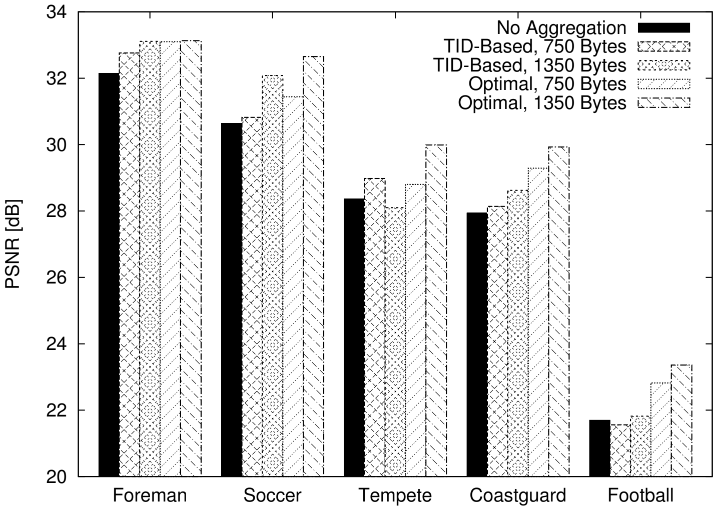

Fig. 5. PSNR of the received videos as a function of the NALU aggregation strategy and the limit to the size of the aggregated RTP packet.

in the case of the Foreman sequence, whose average triptime drops from more than half a second to less than 0.2 s in the case of the *Optimal* strategy.

Figure 5 shows that the *TID-based* and the *Optimal* strategies achieve PSNR gains in excess of 1 dB with respect to the standard case where no NALU aggregation is performed. Clearly, the *Optimal* strategy achieves the most considerable improvements in video quality (1.6 dB with the Football sequence and 1.8 dB with Tempete). The performance of the *TID-based* strategy is in many cases close to that of the *Optimal* strategy, whose computational complexity is however higher. The increase in video quality is mainly due to the correct reception of a higher number of NALUs with TID equal to four. Such NALUs noticeably increase the temporal resolution of the video sequences characterized by high levels of picture motion such as Soccer or Football, which in fact boast the most significant PSNR increases.

Aggregating NALUs improves the communication quality even in noisy channel conditions: a single transmission error in a packet causes the whole packet to be discarded, so the larger the packet is, the more likely it is affected by errors. However, if the aggregation strategy is effective in reducing the number of transmitted packets, the packet loss rate increase due to channel errors can be counterbalanced by the 802.11 automatic retransmission mechanism, leading to a reduction of the application-level BLR and an increase in video quality.

6 Conclusions

This work aims at improving the quality of H.264/SVC video communications over 802.11e networks. In particular, this work addresses the inefficiencies of the 802.11 channel access mechanism when a high number of small packets are offered to the network, as it is often the case of SVC communications. Two strategies with different computational complexity and effectiveness are proposed to aggregate multiple SVC data units prior to their transmission, also taking into account their perceptual importance. Both strategies produce RTP packets which are compliant with the RFC standards. The performance of the proposed strategies is evaluated in a domestic 802.11e wireless ad hoc network scenario in various channel conditions. Simulation results show that the proposed low-complexity strategy significantly improves channel utilization and provides significant quality gains as well as more timely packet delivery with respect to a traditional SVC communication. Moreover, the performance of the proposed low-complexity strategy is close to that of the optimal, high-complexity, strategy.

References

1. IEEE 802.11e Standard, 2005: Wireless LAN Medium Access Control and Physical Layer specifications Amendment 8: Medium Access Control Quality of Service Enhancements.
2. Sven Wiethölter and Marc Emmelmann and Christian Hoene and Adam Wolisz: TKN EDCA Model for ns-2. Telecommunication Networks Group, Technische Universität Berlin", 2006, TKN-06-003.
3. S. Wenger and Y. K. Wang and T. Schierl: RTP Payload Format for SVC Video. IETF draft-ietf-avt-rtp-svc-15 (work in progress), Nov. 2008.
4. UCB / LBNL / VINT: Network Simulator - NS - Version 2.31. http://www.isi.edu/nsnam/ns.
5. ITU-T & ISO/IEC Standard: Advanced video coding for generic audiovisual services. Annex G, Scalable Video Coding H.264 & 14496-10. Nov. 2007.
6. A. Ksentini and M. Naimi and A. Guéroui: Toward an improvement of H.264 Video transmission over IEEE 802.11e through a cross-layer architecture. IEEE Communications Society Magazine. Jan. 2006. Volume 44, pages 107–114.
7. H. Schwarz and D. Marpe and T. Wiegand: Overview of the Scalable Video Coding Extension of the H.264/AVC Standard. IEEE Transactions on Circuits and Systems for Video Technology. Sept. 2007, Volume 17, Number 9.
8. A. Fiandrotti and D. Gallucci and E. Masala and E. Magli: Traffic Prioritization of H.264/SVC Video over 802.11e Ad Hoc Wireless Networks. Proceedings of the 17th International Conference on Computer Communications and Networks, 2008 ICCCN'08. Aug. 2008.
9. ISO/IEC 8802-11, ANSI/IEEE Std 802.11 Standard: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications. 1999.
10. D. Niculescu and S. Ganguly and K. Kim and R. Izmailov: Performance of VoIP in a 802.11 Wireless Mesh Network. Proceedings of the IEEE INFOCOM. Apr. 2006.
11. G. Bianchi: Performance analysis of the IEEE 802.11 distributed coordination function. IEEE Journal on Selected Areas in Communications. Mar. 2000. Volume 18, Number=3, Pages 535–547.
12. H.264/AVC Reference Software JM 11.0, available online at http://iphome.hhi.de/suehring/tml/doc/lenc/html/index.html.
13. H.264/SVC Reference Software JSVM 9.4, available online at http://ip.hhi.de/imagecom_G1/savce/downloads/SVC-Reference-Software.htm.