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Adaptive and ubiquitous video streaming over Wireless Mesh Networks

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Abstract In recent years, with the dramatic improvement on scalability of H.264/MPEG-4 standard and growing demand for new multimedia services have spurred the research on scalable video streaming over wireless networks in both industry and academia. Video streaming applications are increasingly being deployed in Wireless Mesh Networks (WMNs). However, robust streaming of video over WMNs poses many challenges due to varying nature of wireless networks. Bit-errors, packet-losses and burst-packet-losses are very common in such type of networks, which severely influence the perceived video quality at receiving end. Therefore, a carefully-designed error recovery scheme must be employed. In this paper, we propose an interactive and ubiquitous video streaming scheme for Scalable Video Coding (SVC) based video streaming over WMNs towards heterogeneous receivers. Intelligently taking the benefit of path diversity, the proposed scheme initially calculates the quality of all candidate paths and then based on quality of path it decides adaptively the size and level of error protection for all packets in order to combat the effect of losses on perceived quality of reconstructed video at receiving end. Our experimental results show that the proposed streaming approach can react to varying channel conditions with less degradation in video quality.

1. Introduction

In recent years, with the dramatic improvement on scalability of H.264/MPEG-4 standard and growing demand for new multimedia services have spurred the research on scalable video streaming over wireless networks in both industry and academia. Robust streaming of video over wireless networks is fraught with many challenges including bit-errors, packet-losses and burst-packet-losses due to varying nature of wireless networks. In the case of Wireless Mesh Networks (WMN), we have multiple challenges. These challenges are diverse nature of topology, non existence of fixed infrastructure, and due to varying number of hops it becomes more difficult to maintain

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the required delay and quality of service (QoS). Since WMN is more error prone than classic wireless networks we need to address all possible error protection methods both at server and receiver ends as we can not rely on middle infrastructure. The errors in WMN severely influence the perceived video quality at receiving end. Wireless transmission at high packet rates is often characterized by burst-packet-loss behavior, i.e. if one packet is lost there is more chance that consecutive packets will also be lost. The effect of high error rates can be more devastating for streaming of compressed video such H.264/MPEG-4 SVC, which uses motion-compensated prediction. Although, motion-compensated prediction can achieve high compression efficiency, but it is not designed for streaming over error-prone channels. In motion-compensated based coding, the video sequence consists of two types of video frames, intra-frames (I-frames) and inter-frames (P or B-frames). The intra-frames are encoded by only removing spatial redundancy present in the frame. P-frames are encoded through motion estimation using preceding I or P-frames as a reference frame. B-frames are encoded bi-directionally using the preceding and succeeding reference frames. This poses a severe problem called error propagation, where the errors due to packet loss in a reference frame propagate to all of the dependent frames leading a severe degradation in perceived video quality that can be long-lasting. Thus carefully-designed error recovery scheme is essential for providing reliable and robust video streaming.

Many approaches dealing with the error recovery have been proposed in literature such as error resilience, error concealment, forward error correction (FEC) and automatic repeat request (ARQ). However, none of these approaches can fulfill all quality criteria by itself. A scalable bit-stream of video consists of a base layer and one or more enhancement layers. The base layer provides a basic level of video quality and is decodable independently of the enhancement layers. While on the other hand, the enhancement layers serve only to refine the base layer quality and are not useful alone. Thus, the base layer represents the most important part of video data, which makes the performance of streaming applications that employ layered representation sensitive to losses of base layer packets. Therefore, the base layer needs to be protected more strongly as compared to enhancement layers. However, assigning unequal error protection to scalable video is more complex and difficult than non-scalable video due to layered structure.

In this paper we propose and implement a hybrid error control scheme for Scalable Video Coding based streaming over WMNs which is a combination of adaptive unequal error protection, adaptive packet size assignment and path diversity. The work presented in this paper is an extension of Kormentzas (2010). The work in Kormentzas (2010) focuses on the fixed packet size (FPS) with fixed unequal error protection (FUEP). In this work, we extend the work in the following three directions.

1. Fixed packet size with adaptive unequal error protection (FPS + AUEP).
2. Adaptive packet size with fixed unequal error protection (APS + FUEP).
3. Adaptive packet size with adaptive unequal error protection(APS + AUEP).

It is important to mention here that prior work in the field of ad hoc networks has extensively taken advantage of the path diversity available through layered video coding or Multi Description Coding (MDC). The key question is how to utilize this path diversity more efficiently. In our proposed scheme the nodes are able to send periodically their state information to all of their neighbor nodes. Thus, based on node's state information, such as delay, jitter, loss rate and throughput, first the source node calculates the quality of all candidate paths using gray relational analysis (GRA). Then based on quality of path, it decides adaptively the size and level of error protection for all packets in order to combat the effect of losses on perceived quality of reconstructed video. The proposed scheme is implemented using HD/SD SVC-based video streams on the real platform rather than using any simulation tool as explained in Section 6. The performance comparisons of proposed scheme with some other existing schemes were performed. The video quality is measured based on most widely used metrics such as peak signal to noise ratio (PSNR) and Video Quality Model (VQM). The VQM tool (Wolf, 2006), which implements the International Telecommunication Unit (ITU-T) J.144 recommendation (Objective perceptual video, 2003) compares the original video stream with the reconstructed video stream using television model and reports a metric between 0 and 1. Lower VQM scores correspond to better video quality. After a series of repeatable experiments on test-bed, our results show that the proposed streaming approach can react to varying channel conditions with less degradation in video quality.

Rest of the paper is organized as follows: Section 2 describes the related work. Brief overview of scalable video is provided in Section 3. The proposed path selection and adaptive unequal error protection with adaptive packet size assignment schemes are explained in Sections 4 and 5 respectively. An overview of test platform and experimental results are presented in Section 6. Finally, this work is concluded in Section 7.

2. Related work

In this section we briefly review the previous work on the error control for video streaming. Basically the error control schemes can be classified into four categories e.g. error resilience, error concealment, forward error correction and automatic repeat request. The first type involves the design of smart encoders which attempt to limit the scope of the visual damage caused by lost data. The second type deals with designing of smart decoders, which attempt to hide the lost data using received data. The third type involves adding redundant data while the last one involves retransmission of lost data. The error control for Scalable Video Coding has attracted much attention due to its importance and many studies have been proposed so for. As discussed in Section 3, in scalable video bi-stream different layers have different importance so they should be protected unequally according to their importance. However, assigning unequal error protection to scalable video is more complex than non-scalable video due to layered structure of SVC.

Many studies have been conducted to tackle the problem of Unequal Error Protection (UEP) for SVC by appropriate consideration of the various frame types such as in Ang et al. (2003), Marx and Farah (2004), Fang and Chau (2005) and Helle et al. (2013). Some researchers have focused on applying

UEP to different layers according to their importance (Cai et al., 2004; vvan der Schaar and Radha, 2001; Costa et al., 2004; Adzic et al., 2014; Bursalioglu and Caire, 2011). However, UEP can be fixed or adaptive. Fixed unequal error protection schemes can be associated with providing strong protection under reliable channel conditions resulting wastage of bandwidth or providing weak protection under bad channel conditions, hence resulting severe degradation in perceived video quality. It was observed that adaptive protection level performs much better than non-adaptive (Siruvuri et al., 2009). A novel adaptive unequal error protection for scalable video over wireless networks was proposed in Naghdinezhad and Fatemi (2007). Experimental results show a significant improvement of 1.27 dB as compared with conventional methods. Another adaptive systematic lossy error protection scheme was presented in Ramon et al. (2009) for broadcast applications in which the Wyner–Ziv (WZ) stream is obtained by frequency filtering in the transform domain. The scheme is based on frequency filtering and unequal error protection. The ratio of error resilience varies adaptively according to characteristics of the compressed bit-streams. The authors in Liang et al. (2007) and Liang et al. (2008) demonstrated that using content adaptive unequal error protection or feedback aided unequal error protection can improve error resilience performance. A channel adaptive UEP scheme was proposed in Dick et al. (2005), which adjusts the channel coding in the base station thus can benefit from efficient hardware implementation enabling energy efficient data streaming over wireless links. A joint source and channel UEP scheme for SVC streaming was proposed for high speed packet access networks in Mansour et al. (2008). The proposed approach uses the video priority information along with channel quality information to set the channel coding rate in order to maximize the video quality.

Small packet length increases the header overhead while on the other hand long packet length will tend to increase the packet error rate. The suitable packet length can be obtained through mathematical analysis of current wireless channel status. Based on adaptive packet length and unequal error protection, a video transmission mechanism was proposed in Lee et al. (2009), which has smoother quality degradation on video quality. Numerical results presented in Xiao et al. (2005) show that adaptive scheme combined with automatic repeat request can obtain a good performance. Another adaptive packet and block length forward error correction control mechanism was proposed in Tsai et al. (2010), which obtained better recovery performance than conventional forward error correction schemes.

Due to erroneous nature of Wireless Mesh Networks, a single path often cannot meet the requirement of video transmission. Consequently, multi-path transmission is needed but the key question is how to utilize the path diversity more efficiently. The early work presented in Golubchik et al. (2002) and Li et al. (2004) establishes the generic framework for multi-path streaming, which emerged as an effective technique to overcome some of the limitations of wireless networks. The specific advantages brought by the multi-path streaming consist of aggregated network bandwidth, packet loss decorrelation and delay reduction. The use of multi-path also allows increasing the streaming bandwidth by balancing the load over multiple network paths between source and destination. Experimental work presented in Apostolopoulos et al. (2002) shows that the combination of multiple description coding and path diversity provide improved error resilience for streaming media over best-effort networks. To improve the end-to-end reliability, forward error correction across multiple independent paths was applied in Fashandi et al. (2007), which provides significant performance improvement as compared to other alternatives. Problems of robust video streaming in multi-hop networks by relying on delay-constrained and distortion-aware scheduling, path diversity and retransmission of important video packets over multiple links to maximize the perceived video quality at receiving end were discussed in Tong et al. (2007). Another feasible path selection algorithm was proposed in Prakash and Selvan (2008), which addresses issues that pertain to finding a feasible path subject to delay and cost constraints and it offers high success rate in finding feasible path. In order to select the best path among all candidate paths, some metric e.g. predicted throughput, delay, packet loss or even energy consumption of any node, is required. The work presented in Ju and Evans (2008) provides a method for multi-path selection based on parameters prediction. The results of Muscariello et al. (2009) suggest that next generation networks should evolve the more meshed topologies in order to exploit path diversity and implement multi-path routing strategies. Another autonomic flow based path selection mechanism was proposed in Xiaoli et al. (2009) to autonomically and efficiently select the best path for traffic flow.

Contrary to the existing work, our proposed scheme provides two stage unequal error protection for SVC-based video streaming. In the first stage, we perform appropriate path selection for different layers and then in the second stage we assign adaptive unequal error protection and packet size. In the proposed scheme the destination nodes send their information in a periodic manner enabling source nodes to calculate the network topology based on gray relational analysis; hence dynamically adjusting the video quality depending upon the information received.

3. Scalable Video Coding

Scalable Video Coding (Schwarz et al., 2006), an extension of H.264/MPEG-4 Advanced Video Coding (AVC) is a video coding technology that encodes the video at the highest resolution, and allows the bit-stream to be adapted to provide various lower resolutions. It provides the way to show graceful degradation of video quality while streaming over error-prone channels in wireless networks. Scalable encoded video data enables a decoder to decode selectively only part of the coded bit stream. The main idea behind the scalable video is to create a compressed bit-stream which can be used by different users according to their needs. The users can selectively decode the bit-stream according to their computational power and visualization capability to get the best quality video. To achieve the scalability, the video data are encoded into several layers, i.e. base layer and one or several enhancement layers. The base layer is coded in compliance with H.264/MPEG-4 AVC, and each H.264/MPEG-4 AVC standard decoder is capable of decoding the base layer of SVC bit-stream. A base layer encodes the lowest temporal, spatial and quality representation of the video stream while enhancement layers encode additional information. The lower layers contain lower resolution data. These data are more important because it provides basic

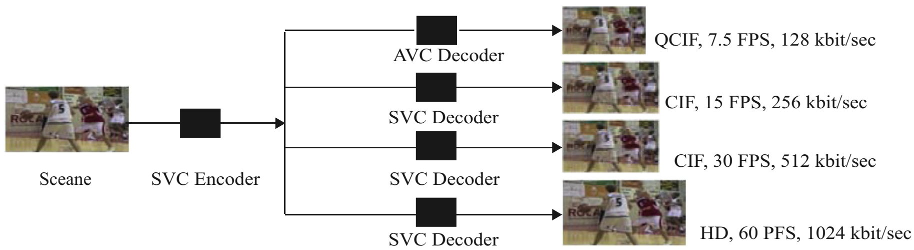


Figure 1 On-the fly adaption of streaming video contents.

video quality with low bit-rate. The higher layers contain the refinement data. It refines the lower resolution data to provide higher resolution video. The refinement data are less important and can be removed when the bandwidth or decoding capability is not sufficient. So that the more the layers that are used in the decoding process, the higher is the quality of the reconstructed video as shown in Fig. 1.

Scalable video can be scalable in different ways. As discussed in Yu et al. (2014), it can be spatially scalable accommodating a range of resolutions on visualizing screen. Spatial scalability is achieved by different encoder loop with an over-sampled pyramid for each resolution, including hybrid video coding with independent motion compensated prediction structure for each layer. The decoder operates only with a single motion compensated prediction loop. Therefore, inter layer dependent motion compensated prediction of lower layers is not needed. Scalable video can be temporally scalable offering different frame rates. This is generally enabled by restricting motion compensated prediction to reference pictures with a temporal layer equal or less than to the temporal layer of the picture to be predicted. Scalable video generally employs hierarchical B pictures to provide temporal scalability. Salable video can also be scalable in sense of signal-to-noise ratio (SNR), offering video at different quality levels to accommodate the difference in bit-rates of the transmission channels.

4. Wireless path diversity and path selection

Our proposed scheme provides two stage unequal error protection for SVC-based video streaming. The first stage is based on appropriate path selection for different layers according to their importance and the second stage is based on assigning adaptive unequal error protection and packet size. In the proposed scheme the nodes are able to send their state information periodically, so source node gets the state information and calculates the network topology based on gray relational analysis between source and destination. Gray method was developed by Deng (1989) and has been widely used to solve the problems of uncertainty under the discrete data and incomplete information. It is used to analyze the relationship grade from discrete sequences and select the best sequence. One of the sequences is defined as reference sequence presenting the ideal situation. The gray relationship between the reference sequence and the other sequences can be determined by calculating the gray relational coefficient (GRC) according to the level of similarity and variability. The sequence with the largest GRC is the most desirable one. The major advantage of gray relational analysis (GRA) method is that the results are based upon the original data with simple calculations. This technique is also effective for calculating the quality of paths in WMNs. GRA is usually implemented by following six steps:

1. Classifying the networks parameters by two situations (smaller-the-best, larger-the-best).
2. Defining the upper and lower bounds of the parameters.
3. Normalizing the parameters.
4. Defining the ideal situation.
5. Calculating the GRC.
6. Ranking the available paths according to the GRC values.

For the purpose of selecting appropriate paths for different layers, we consider the network-layer metrics such as delay ζ, jitter θ, loss rate σ and throughput α. Delay, jitter and loss rate belong to the smaller-the-best category while throughput belongs to larger-the-best. Before calculating the GRC, the data need to be normalized to eliminate the dimensional units. We use programmable Simple Network Management Protocol (SNMP) traps to calculate all the possible network candidate paths $(P_1, P_2, \ldots, P_n)$, delay, jitter, throughput and packet loss rate with a frequency of 10 seconds. While first SNMP trap carries the required information at the time of network startup. These SNMP traps also permit us to identify rouge access points and change in topology. Assuming that n possible network candidate paths $(P_1, P_2, \ldots, P_n)$ are compared, and each network candidate path has k parameters, the upper bound (u_j) is defiend as max $\{P_1(j), P_2(j), \ldots, P_n(j)\}$ and the lower bound (l_j) as min $\{P_1(j), P_2(j), \ldots, P_n(j)\}$, where $j = 1, 2, \ldots, k$. For smaller-the-best parameters the normalized value of $P_i(j)$ parameter can be calculated as follows:

$$P_i^*(j) = \frac{(u_j) - p_i(j)}{(u_j) - (l_j)} \tag{1}$$

Similarly, for the larger-the-best parameters the normalized value $P_i(j)$ can be calculated as follows:

$$P_i^*(j) = \frac{p_i(j) - (l_j)}{(u_j) - (l_j)} \tag{2}$$

Network path attributes can be represented as a row matrix, where the elements of the matrix are the normalized values of k different network path attributes.

$$P = [P^*(1), P^*(2), P^*(3), \ldots, P^*(k)] \tag{3}$$

While $P_i^*(j)$ parameters are maximized in 1, the most preferable network path can be always described as $P_1^*(j) = 1$, where $j = 1, 2, \ldots, k$, and k is the number of network path parameters used for the decision making. Using the behavior of the normalizing algorithm, the ideal network path can be determined as $S = [1, 1, 1, \ldots, 1]$. If there are N available network paths to choose from, the previous row matrix (3) can be extended to a $N \times k$ matrix, which contains all the parameters that play role in the appropriate network path selection procedure. The matrix can be determined as follows:

$$P_N = \begin{bmatrix} P_1^*(1), P_1^*(2), P_1^*(3), \ldots, P_1^*(k) \\ P_2^*(1), P_2^*(2), P_2^*(3), \ldots, P_2^*(k) \\ \ldots\ldots \\ P_N^*(1), P_N^*(2), P_N^*(3), \ldots, P_N^*(k) \end{bmatrix} \tag{4}$$

The final step is to calculate the GRC as follows:

$$GRC_i = \frac{1}{\sum_{j=1}^{k} w_j |p_i^*(j) - 1| + 1} \tag{5}$$

where w_j is the weight of each parameter and i $(1 \leqslant I \leqslant N)$ is the network index. The path with the largest GRC is the most appropriate path. The source node calculates the GRC for all available paths. As in SVC bit-stream different layers have different priority, the base layer has highest priority and enhancement layer one has lower priority than base layer and enhancement layer two has lower priority than enhancement layer one and so on. So according to priority of layers appropriate paths are assigned to each layer in such a way that base layer stream has highest priority. Therefore, it should be transmitted through the highest quality path (the path with highest GRC value) and while highest enhancement layer has lowest priority so it should be transmitted through lowest quality path (the path with lowest GRC value). It means more important data are transmitting through more reliable path with less error rate. The path with the largest GRC is the most reliable path and vice versa. Thus the scheme ranks all candidate paths according to their robustness which is based on GRC values.

As shown in Table 1, during test we had five candidate paths and we calculated the quality of each path and ranked them according to their robustness. Thus according to Table 1, the most robust path is path 5 and the worst path is path 1. Path 2 is the second robust path and path 4 is third best path. As we used HD/SD SVC-based streams with one base layer and two enhancement layers in our tests, so therefore, we selected the first three most robust paths for video streaming which are P_5, P_2 and P_4 for base layer, enhancement layer one and enhancement layer two respectively. All other paths are ignored (path 1 and 3 in this case) which leads to more important data through more robust path with less error rate probability. So this is the first stage of providing UEP in our proposed scheme.

5. Adaptive UEP and packet size assignment

Now after the appropriate paths are assigned to each layer, the second step is to adaptively assign UEP and Packet Size (PS) to all layers based on path quality. The bit-streams of all layers are interleaved into one Block Of Packets called (BOP) as shown in Fig. 2. The transmitted packets are the rows of the BOP. The source data with length r_i in layer i are grouped into k_i packets, where $i = 1 \sim l$, with column width of s_i. The n is number of packets and the remaining $n - k_i$ packets in the BOP are filled with channel coding redundancy. Therefore, k_i specifies the protection level of the layer i. The BOP buffer size r is assumed to be enough to satisfy delay and buffer constraints for real-time streaming. The length of packet header is s_h. If the number of packets n is known, then the packet size $s = r/n$. Now the first constraint obtained from BOP data structure for forward error correction assignment is as follows:

$$S = S_k + \sum_{i=1}^{l} S_i = S_k + \sum_{i=1}^{l} \frac{r_i}{k_i} \tag{6}$$

Each Group Of Pictures (GOP) can be packed into a fixed number of block of pictures. In our proposed scheme, one GOP is equal to one BOP. In SVC bit-steam different layers have different priorities; therefore, SVC-based encoded video explicitly requires an unequal error protection scheme, yielding another restriction for forward error correction assignment as follows:

$$0 \leqslant k_1 \leqslant k_2 \leqslant \cdots \leqslant k_l \leqslant n \tag{7}$$

We elucidate four different adaptive assignment schemes in this section for scalable video transmission over error-prone wireless networks and finally suggest the best one through experimental results in Section 7. The four schemes are as under:

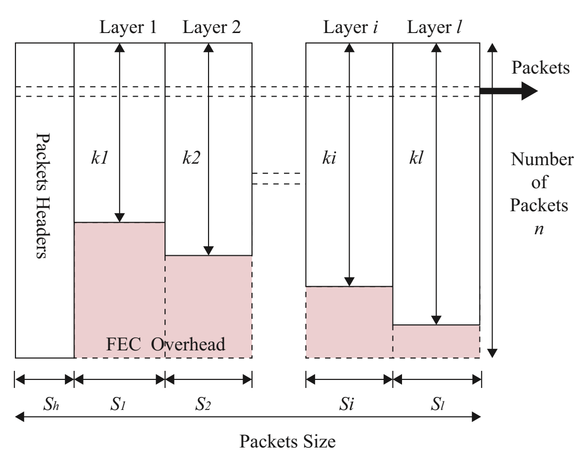


Figure 2 Date structure of block of pictures (BOP).

Table 1 Parameters for appropriate path selection decision making.

Path	GRC	Delay (ζ)		Jitter (θ)		Loss rate (σ)		Throughput (α)	
		ms	Norm	Ratio	Norm	ms	Norm	mbps	Norm
1	0.25	169	0.00	0.300	0.00	05	1.00	08	0.00
2	0.42	096	0.56	0.002	1.00	07	0.86	23	0.20
3	0.27	150	0.15	0.017	0.95	19	0.00	19	0.15
4	0.38	125	0.34	0.027	0.92	07	0.86	25	0.23
5	0.50	039	1.00	0.050	0.84	17	0.14	82	1.00

1. Fixed packet size with fixed unequal error protection (FPS + FUEP).
2. Fixed packet size with adaptive unequal error protection (FPS + AUEP).
3. Adaptive packet size with fixed unequal error protection (APS + FUEP).
4. Adaptive packet size with adaptive unequal error protection(APS + AUEP).

Fig. 3 shows the block of pictures structure of FPS + AUEP, APS + FUEP and APS + AUEP under bad and good channel conditions. In Fig. 3(a and b) the packet size and number of packets are fixed, but the unequal error protection is adaptive. Due to error-prone nature of WMNs, when the channel condition is bad, it calls for an increased forward error correction ratio as shown in Fig. 3(b). The variations in BOP structure under different channel conditions is shown in Fig. 3(c and d), where the packet size is adaptive and forward error correction is fixed. Normally small packet size is suitable under bad channel conditions to reduce the packet error rate resulting in improved video quality at receiving end. But, increase in number of packets leads to header overheads. However, the packet size used in our tests was less than 1500 bytes because of the Maximum Transmission Unit (MTU). Finally, Fig. 3(e and f) represents the BOP structure under different channel conditions for adaptive packet size and adaptive forward error correction. However, the amount of forward error protection required to be added and packet size are key issues. Both of them are directly related with perceived video quality at destination node. In our proposed scheme both the packet size and protection are based on channel conditions.

For adaptive assignment of unequal error protection and packet size the algorithms use Table 2 for decision making. As we can see in Table 2 that based on quality of path, the error protection and packet size are assigned to different layers of scalable video adaptively according to their importance. For example during good channel conditions when the GRC value is around 1, which means the path is almost same as ideal path. The probability for error occurrence is around zero. There is almost no need for error protection redundancy. Furthermore, we can increase the packet size to maximum limit in order to reduce the overhead due to packet headers. While on the other hand, during bad channel conditions, such as in case when GRC value is around 0.1, which means the path is unreliable. Obviously there is a call for increased ratio of error protection as well as small packet size.

6. Performance evaluation

Performance evaluation of our proposed technique is performed through experimental measurements. For this purpose, we have developed a test bed platform. This section is mainly divided into two halves. In the first part, we discuss the network topology, test bed platform used in this work. In this part we further discuss video content server, quality measurement probes, types of video clients, communication system and SVC media gateway. These are the criteria which form the base of our experimentation. Finally, based on these criteria, we present detailed result analysis in the second half of this section.

6.1. Network topology

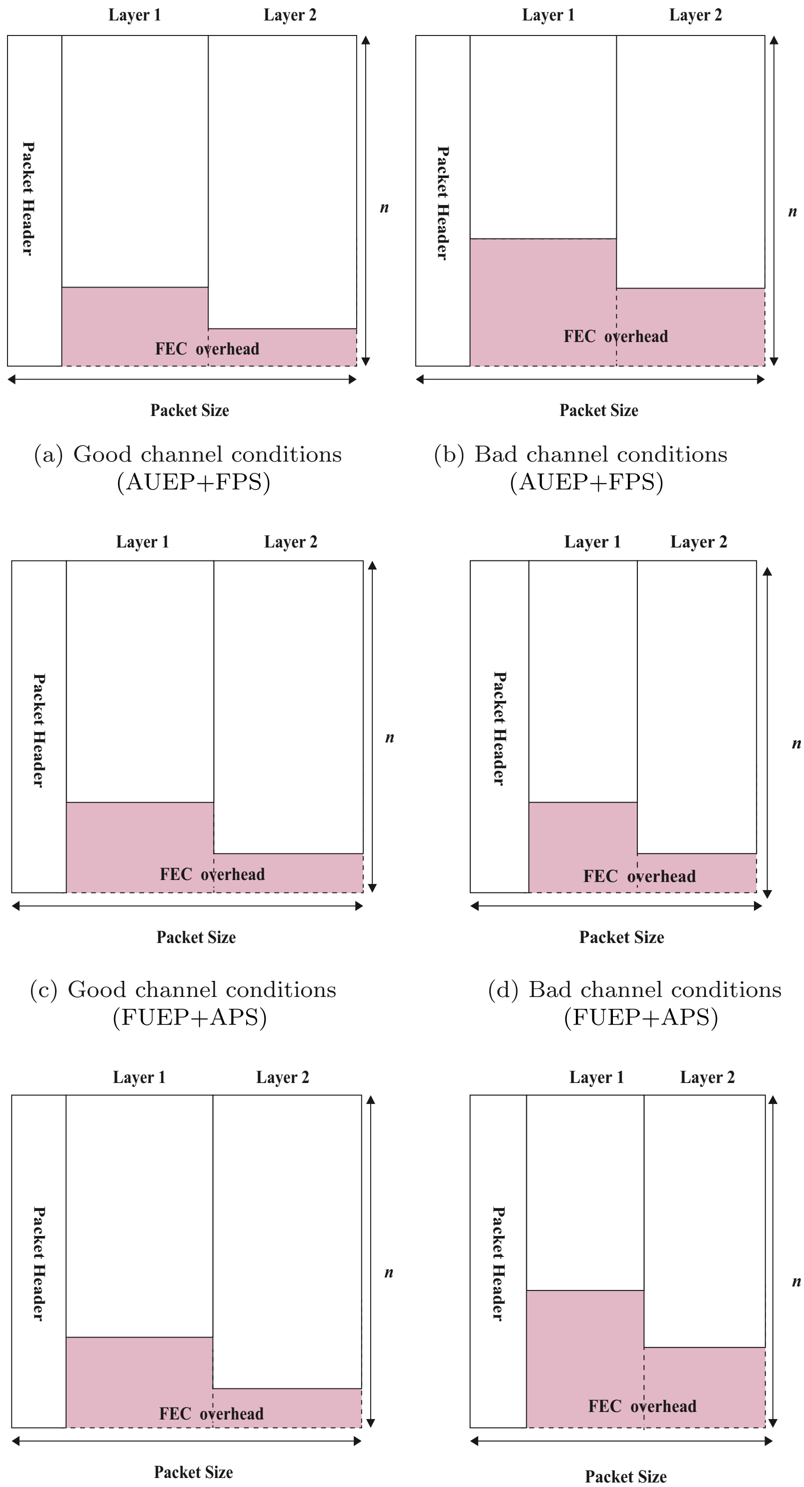
The main goal of this study is to develop efficient solutions for robust scalable video streaming over error-prone channels based on joint source-channel coding ensuring a graceful degradation in perceived video quality at receiving end. This approach allows for strategies where the source coding, channel coding, modulation and network parameters are jointly determined to yield the best end-to-end system performance. The platform offers a solution for scalable video encoding, streaming and quality monitoring and adaptation over wired (xDSL), wireless (WiFi) and residential networks. It supports both live multicast and on-demand unicast video services. As described in Wenger et al. (2007), the most common network distribution modes in context of scalable video are:

1. Multicast/broadcast on the server side, with a Media Aware Network Element (MANE) to aggregate and/or trim sessions. The Network Abstraction Layer (NAL) units of the aggregated and/or trimmed sessions are conveyed jointly on a single transport address, and in a single Real Time Protocol (RTP) session.
2. Multicast/broadcast of video data to receivers with heterogeneous connectivity, where layers are transported in separate RTP sessions on separate transport addresses.
3. Starting from a layered representation in a file, the server generates and sends one RTP session containing possibly more than one layer.

We rely on the last two modes and named them live and Video On Demand (VOD) respectively. In the live scenario each layer is transmitted in its own IP multicast group. The gateway/modem subscribes to layers via IP multicast mechanisms Internet Group Management Protocol (IGMP) depending on user-selected layouts as shown in Fig. 4. As this scenario is designed to optimize the network traffic in the core network, therefore, the gateway/modem node may convert a RTP multicast bit-stream into a RTP unicast for a mobile application for high definition TV for instance. While in VOD scenario, (Fig. 5), the content provider aggregates multiple SVC layers into single RTP session. Thus as this scenario supports personalized layout, the composing process is performed by the content server for each of the receiving end point. The platform integrates the following two modes of transmission as described in Handley et al. (2006):

1. *Single-Session Transmission (SST):* In which all SVC data are carried in a single RTP session. This mode should be used in point-to-point unicast applications or generally whenever the potential benefit of using multiple RTP sessions does not justify the added complexity.
2. *Multi-Session Transmission (MST):* In which two or more RTP sessions are used to carry the SVC data. The MST should be used in a multicast session when different receivers may request different layers of the SVC bit-stream.

According to the current draft of the RTP payload format for SVC video (Wenger et al., 2009), three packetization modes namely Single NAL Unit Mode, Non-Interleaved Mode and Inter-leaved Modes in case of SST and three packetization modes namely Non-Interleaved Timestamp based (NI-T), Non-interleaved Cross Session Decoding Order Number



(a) Good channel conditions (AUEP+FPS)

(b) Bad channel conditions (AUEP+FPS)

(c) Good channel conditions (FUEP+APS)

(d) Bad channel conditions (FUEP+APS)

(e) Good channel conditions (AUEP+APS)

(f) Bad channel conditions (AUEP+APS)

Figure 3 block of pictures (BOP) structure under different channel condition.

(CS-DON) based mode (NI-C) and Non interleaved combined timestamp and CS-DON mode (NI-TC) in case of MST were retained. While Interleaved CS-DON (IC) mode is not retained because it requires relatively high end-to-end latency and the decoding order recovery process is not as straightforward as in non-interleaved modes. The signaling of SVC streams is based on the Session Description Protocol (SDP) (Handley et al., 2006). The SDP is intended for describing multimedia

Table 2 AUEP and PS assignment under different channel conditions.

GRC	Base layer		E-layer 1		E-layer 2		E-layer 3	
	UEP (%)	PS (bytes)	UEP (%)	PS (bytes)	UEP (%)	PS (bytes)	UEP (%)	PS (bytes)
0.9–0001	03	1024	00	1096	00	1168	00	1240
0.8–0.89	06	0952	03	1024	00	1096	00	1168
0.7–0.79	09	0880	06	0952	03	1024	00	1096
0.6–0.69	12	0808	09	0880	06	0952	03	1024
0.5–0.59	15	0736	12	0808	09	0880	06	0952
0.4–0.49	18	0664	15	0736	12	0808	09	0880
0.3–0.39	21	0592	18	0664	15	0736	12	0808
0.2–0.29	24	0520	21	0592	18	0664	15	0736
0.1–0.19	27	0448	24	0520	21	0592	18	0664

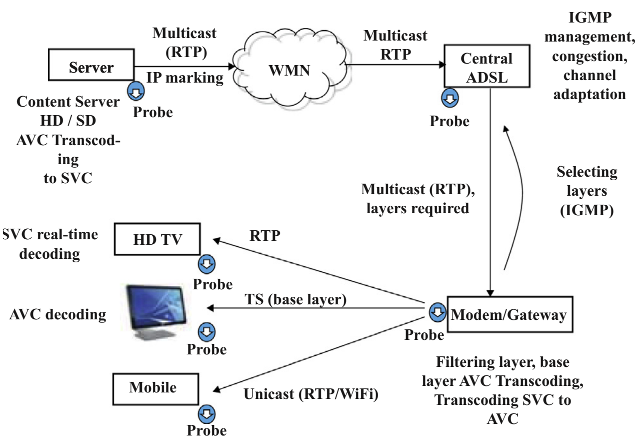


Figure 4 Network topology for live streaming.

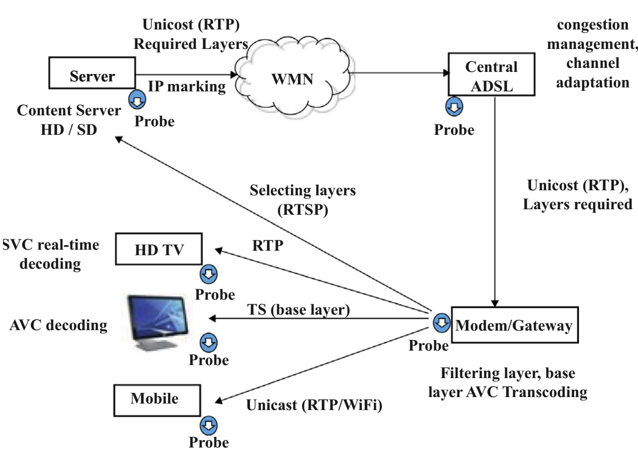


Figure 5 Network topology for VOD streaming.

communication sessions for the purposes of session announcement, session invitation and parameter negotiation. The SDP does not deliver media itself but it provides information about media streams contained in media session. Due to introduction of scalability, SDP defines a set of rules on signaling media decoding dependencies. Two types of dependencies, layered/hierarchical decoding dependencies and multiple description decoding dependencies can be distinguished (Schierl and

Figure 6 Test platform.

Wengeret, 2009). In both cases SDP provides information about the potential dependencies between layers and media formats which allows for signaling a range of transport addresses in a certain media description. In our study SDP is conveyed by the Real Time Streaming Protocol (RTSP). Thus the gateway/modem node acts as an RTSP proxy relying on RTSP messages from terminals to the streaming server. The picture of test platform (shown in Fig. 6) developed integrates the two modes of transmission as described above and revolves around the following five main modules.

6.1.1. Scalable video content server

The content server covers different sub-modules including SVC encoder, SVC streamer, video data base, packet/stream buffers, sending controller, QoS measurement probes and protection module. The SVC encoder encodes offline all video sequences and are stored in video data base. The SVC encoder used in test platform is an optimized version of Joint Scalable Video Model (JSVM). The experimental validation shows considerable improvement in encoding time as compared to original JSVM. The requested bit-streams are moved to the stream buffer and the streamer which accepts commands from the sending controller, segments each bit-stream into video packets. The video packets are put into the packet buffer as the RTP payload. The sending controller interacts with the receiving controller to create a media session for video transmission then server transmits pre-encoded SVC video packets over RTP/UDP.

6.1.2. Network QoS measurement probes

Two main types can be considered in the description of network characteristics: capabilities and conditions. The capabilities define static attributes of a network while the conditions describe dynamic behavior of network. We are more concerned with conditions which specify attributes that describe the available bandwidth, packet loss rate, delay, jitter etc. Many protocols can be used for sending back the value of performance metric such as MPEG-21 Event Reporting (ER), Digital Items Adaption (DIA) and Digital Right Management (DRM). The DIA specifies metadata for assisting the adaption of digital items according to constraints on storage, transmission and consumption, thereby enabling various types of quality of services. Real-time transport control protocol also provides information about the quality of real time media flows through its feedback reports. We adopted Real Time Control Protocol (RTCP) for transferring the network condition reports from receiver to ServMon agent in Digital Subscriber Line Access Multiplexer (DSLAM) as shown in Fig. 7. Usually this information is transferred periodically through receiver report message in RTCP to synchronization source. In fact, ServMon (Server Monitoring) developed in FP6 european project ENTHRONE, in DSLAM exploits and translates these reports into perceptual quality (MOS) before sending them to adaption agent. This may be located in DSLAM (IPTV) or in a remote server (VOD) by using the Simple Object Access Protocol (SOAP).

6.1.3. Video clients

The video clients or customer's terminals cover different submodules including SVC decoder, stream/packet buffer, receiving controller, QoS measurement probes and video display (HD and Mobile). At the client side we used Joint Scalable Video Model (JSVM) decoder in order to improve its speed and experimental validation shows considerable improvement in decoding time. The decoder aims to provide a real-time fast and robust decoding framework for SVC. The buffers are used to smooth out the video stream due to changes in bandwidth, packet losses or jitters. Video frames are segmented in different size packets according to channel conditions which arrive at the receiver with an inter-arrival rate. All packets are decompressed and assembled to form a frame again and stored in a buffer and waits there until they are fetched by a display process. If the HD enhancement layer is no longer available due to network disruptions, the receivers automatically resize the image HD to SD for a better user experience. The WiFi connected mobile terminals decode the AVC bit-stream in only 720 × 576 SD resolution.

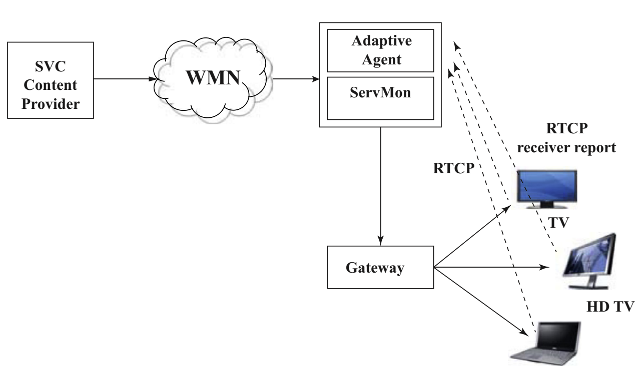


Figure 7 Real-time transport control protocol for transporting metrics.

6.1.4. Communication system

The communication system is a collection of following individual communication networks:

1. The IP Wireless Mesh Network backbone of a national operator supports IP multicast protocols using HP (2015) network emulator.
2. ADSL access network, emulating different profiles of ADSL lines in accordance to European Telecommunications Standards Institute (ETSI) specifications.
3. The home network that provides two communication technologies, wired Ethernet and wireless 802.11g.
4. The network traffic generator to generate high volume of IP/TCP/UD traffic from clients to server to stress test routers, servers and firewalls under extreme network load. Traffic is divided into a client and a server. The server needs to be run on one interface of the router and the client on as many other interfaces as we need to. This additional traffic can cause traffic congestion in the network resulting in losses and transmission delays of video service target packets.

Table 3 Parameters of video sequences used in test.

Video sequence		Basket	Stadium
Resolution		1280 × 720	640 × 480
Profile_idc		77 (Main)	66 (Baseline)
Level_idc		30 (3.0)	12 (1.2)
Coding syntax		CAVLC	CAVLC
Bit rate	AVC	1495.85	191.88
(kbps)	SVC	4089.10	719.17
Frame rate	AVC	25	15
(fps)	SVC	25	15

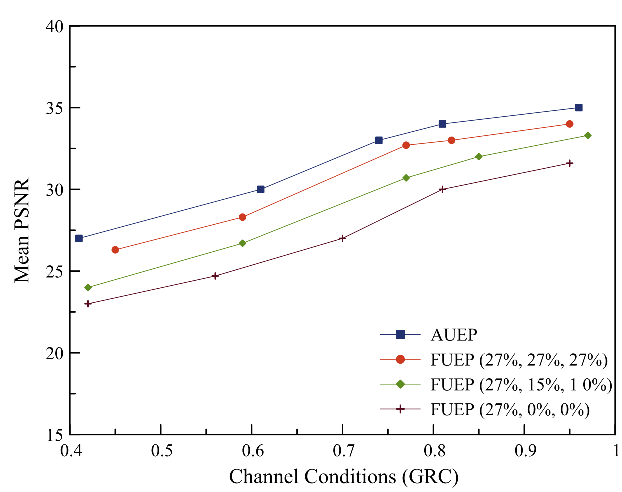


Figure 8 Performance comparison of proposed AUEP with other three FUEP schemes for basket sequence.

6.1.5. SVC media gateway

The gateway acts as a multimedia aware network element, i.e. it is able to aggregate or separate RTP sessions function of the elements that constitute the network. In our case the gateway is able to aggregate the RTP sessions in a single RTP session at receiver on the receiving end, i.e. HD TV or mobile client. The gateway is therefore, likely to select SVC layers, repackage them according to the client's connections. The gateway connects via IGMP for multicast address based on layers it wishes to receive and unsubscribes if network disturbances are too important for a particular enhancement layer. As the base layer carries most important part of video data, therefore, it is important that it should be transmitted to the gateway without disruption. To do this, we mark the Type Of Service (TOS) field of IP datagrams containing information from the AVC layer for transmitting to the DSLAM with a high priority and a significant level of FEC.

6.2. Results analysis

In the following sub-sections, we present experimental results in order to evaluate the performance of proposed scheme in different scenarios. We used the test platform as described in Section 6.1 for SVC-based streaming. We used HD/SD SVC bit-streams with one base layer and two enhancement layers. All the video sequences are encoded using SVC encoder which is an enhanced version of SVC reference software. The characteristics of video sequences are summarized in Table 3. We also consider some back ground traffic in order to achieve different channel conditions.

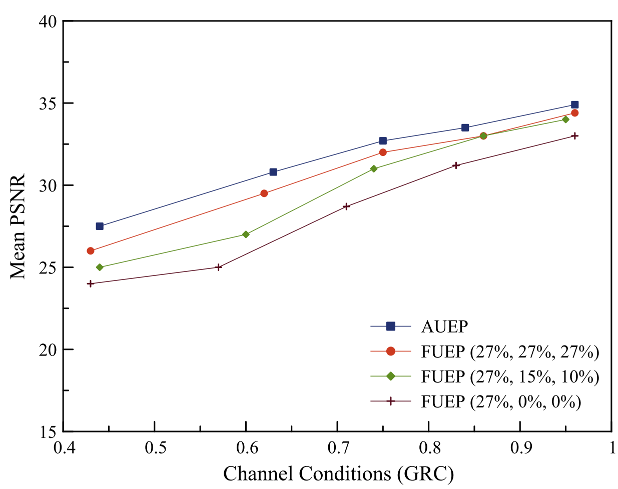


Figure 9 Performance comparison of proposed AUEP with other three FUEP schemes for stadium sequence.

6.2.1. Evaluation of AUEP + PFS

Initially, we considered the scenario in which UEP is adaptive and PS is fixed. We compare the performance of our proposed AUEP scheme with following three fixed error protection schemes using both sequences basket and stadium:

1. Protecting all layers with fixed error protection.
2. Protecting all layers with fixed but unequal error protection.
3. Protection only base layer with fixed error protection.

The performance comparisons are shown in Figs. 8 and 9. The percentages in parentheses in the legends show the protection ratio for the base layer, enhancement layer 1 and enhancement layer 2 respectively. If we consider the only two cases AUEP and FUEP (27%, 27%, 27%) for all layers, the ratio of error protection for FUEP (27%, 27%, 27%) is more than proposed AUEP. But still the graph shows improved PSNR for proposed AUEP. This is due to appropriate path assignment to different layers according to their importance, which means we are transmitting the base layer through most reliable path, enhancement layer 1 through second most reliable path and enhancement layer 2 through third most reliable path, hence resulting in improved video quality with less overhead. Furthermore, the non-adaptive schemes can be associated with providing strong protection under reliable channel conditions, which is useless and wastage of bandwidth, or providing weak protection under bad channel conditions resulting severe degradation in video quality. Table 4 shows the objective parameters for measuring the perceptual effects of the usual types of television impairments under television model VQM_T. As shown in Table 4, the VQM scores for the scenario AUEP are 0.16 and 0.17 respectively for both video sequences, which

Table 4 Summary of objective parameters for AUEP + FPS scheme

Video sequences	Scenarios	VMQ score	Blurring (%)	Jerky motions (%)	Global noise (%)	Error blocks (%)
Basket	AUEP	0.16	0.6	13	0.8	6
	FUEP (27%, 27%, 27%)	0.19	0.6	13	0.8	9
	FUEP (27%, 15%, 10%)	0.24	0.8	15	0.9	9
	FUEP (27%, 0%, 0%)	0.29	0.7	14	0.8	13
Stadium	AUEP	0.17	0.6	09	0.7	7
	FUEP (27%, 27%, 27%)	0.19	0.7	12	0.6	12
	FUEP (27%, 15%, 10%)	0.20	0.8	12	0.7	12
	FUEP (27%, 0%, 0%)	0.27	0.7	15	0.8	13

is much better than all other FUEP scenarios. The results explicitly show that only AUEP can cope with different channel conditions with smoother degradation in perceived video quality.

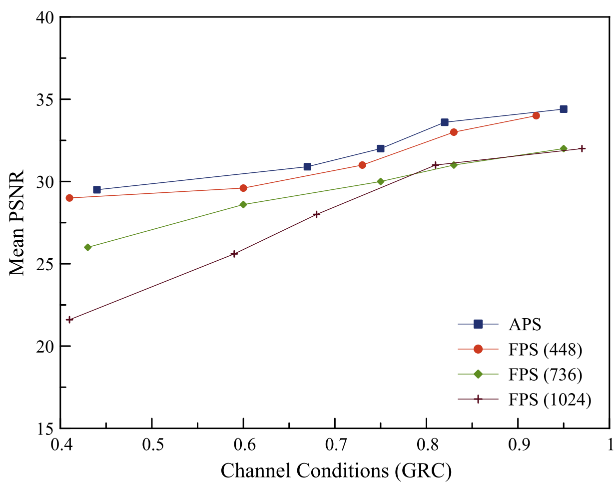


Figure 10 Performance comparison of proposed APS with three FPS schemes for basket sequence.

6.2.2. Evaluation of FUEP + APS

Secondly, we considered the scenario in which UEP is fixed and PS is adaptive. The performance comparisons of our proposed APS with some other fixed packet size schemes using both sequences basket and stadium are shown in Figs. 10 and 11. The non-adaptive fixed packet sizes are 448, 736 and 1024 bytes. The impact of packet loss on perceived video quality depends on several factors, including packet size. As shown in Figs. 10 and 11 , the length of the packet size severely influences the quality of reconstructed video. Hence, only the adaptive packet size assignment can cope with varying channel condition with lesser degradation in perceived video quality. Furthermore, it is noted that in fixed packet size schemes, the packets are either too large with high packet error under bad channel conditions or too small with larger headers overhead during good channel conditions. As we can see under reliable channel conditions the packet size for our proposed scheme is larger than other two fixed schemes but still the proposed scheme outperforms the other schemes, again this achievement is due to selection of best quality path. The ratio for fixed UEP were 27%, 15% and 10% for base layer, enhancement layer one and enhancement layer two respectively.

Table 5 shows the objective parameters for measuring the perceptual effects of the usual types of television impairments under television model VQM_T. As shown in Table 5, the VQM scores for the scenario APS are 0.19 and 0.18 respectively for

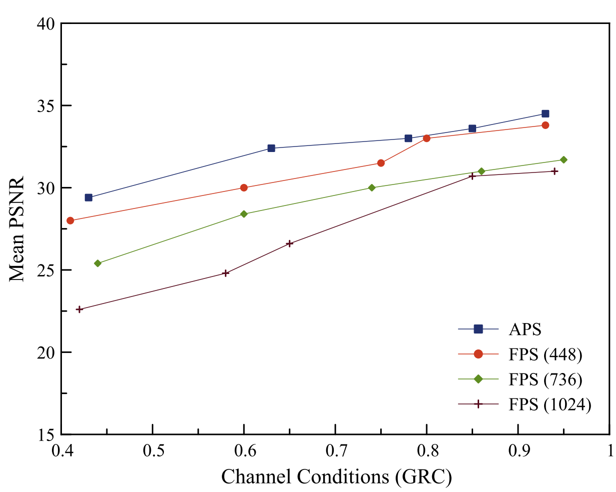


Figure 11 Performance comparison of proposed APS with three FPS schemes for stadium sequence.

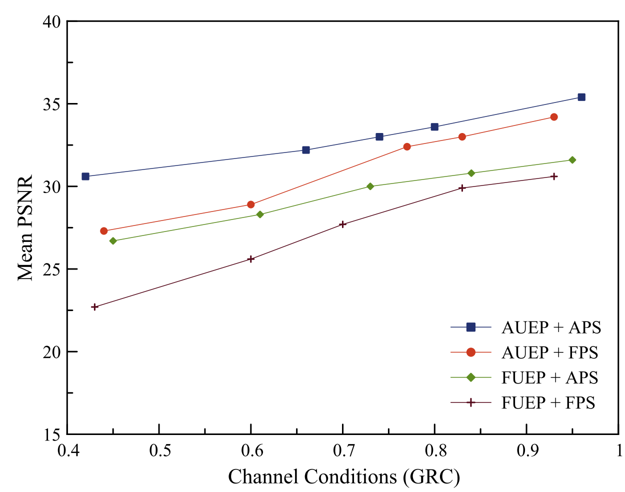


Figure 12 Performance comparison of four proposed schemes for basket sequence.

Table 5 Summary of objective parameters for FUEP + APS scheme.

Video sequences	Scenarios	VMQ score	Blurring (%)	Jerky motions (%)	Global noise (%)	Error blocks (%)
Basket	APS	0.19	0.8	16	0.5	7
	FPS (448)	0.20	0.8	16	0.8	9
	FPS (736)	0.25	0.9	19	0.7	11
	FPS (1024)	0.29	1.3	18	0.9	15
Stadium	APS	0.18	0.6	15	0.7	7
	FPS (448)	0.21	0.9	18	0.7	9
	FPS (736)	0.24	1.2	18	0.9	11
	FPS (1024)	0.27	1.7	21	0.8	14

both video sequences, which show better video quality as compared to all other FPS schemes. Hence, the results show that APS performs better under varying channel conditions with lesser degradation in transmitted video quality.

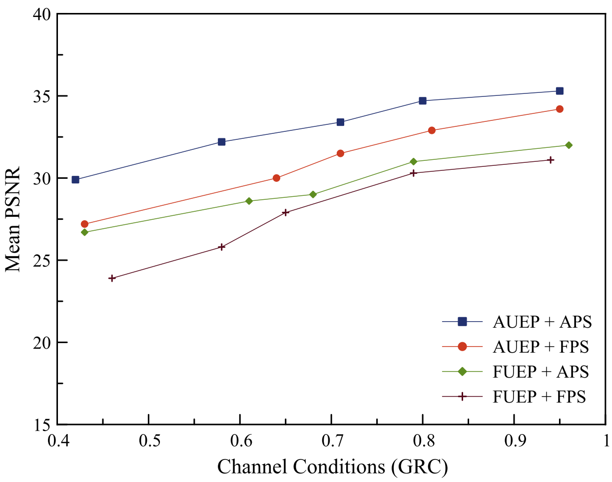


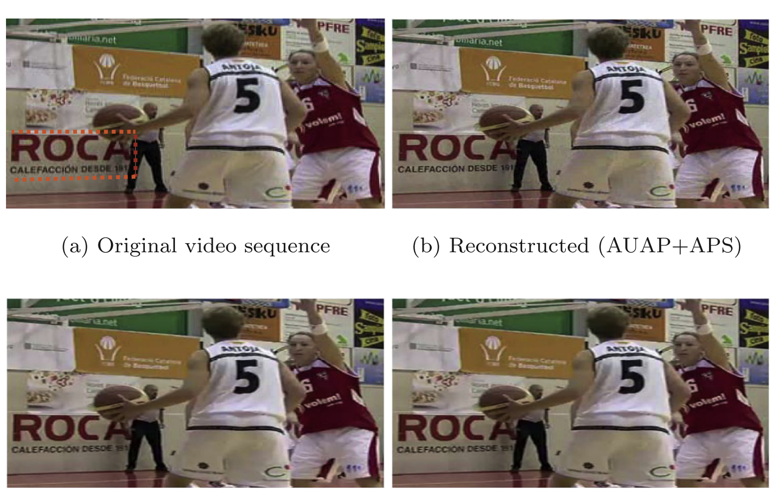
Figure 13 Performance comparison of four proposed schemes for stadium sequence.

6.2.3. Evaluation of AUEP + APS

Finally, we considered the four different scenarios, i.e. AUEP + APS, AUEP + FPS, FUEP + APS and FUEP + FPS. The performance comparison of all schemes over different channel conditions using both sequence basket and stadium are shown in Figs. 12 and 13. The ratios of FUEP are 27%, 15% and 10% for base layer, enhancement layer 1 and enhancement layer 2 respectively and fixed packet size is 1024 bytes. As shown in Figs. 12 and 13, the adaptive schemes perform better than fixed schemes under all channel conditions. Furthermore, the protection level for FUEP schemes is large under good channel conditions, which is wastage of bandwidth, while on the other hand packet's size for FPS schemes is as well large under bed channel conditions resulting in degradation of perceived quality of reconstructed video. Table 6 shows the objective parameters for measuring the perceptual effects of the usual types of television impairments under television model VQM_T. As shown in Table 6, the VQM scores for the scenario AUEP + APS are 0.11 and 0.11 for both video sequences basket and stadium, which show better perceived video quality as compared to other FUEP and FPS schemes. All results show that only AUEP and APS can cope with different varying channel conditions with lesser degradation in transmitted video quality.

Table 6 Summary of objective parameters for AUEP + APS scheme.

Video sequences	Scenarios	VMQ score	Blurring (%)	Jerky motions (%)	Global noise (%)	Error blocks (%)
Basket	AUEP + APS	0.11	0.4	09	0.6	3
	AUEP + FPS	0.16	0.8	12	0.7	8
	FUEP + APS	0.21	0.8	12	0.9	12
	FUEP + FPS	0.26	0.9	21	0.8	17
Stadium	AUEP + APS	0.11	0.5	09	0.5	6
	AUEP + FPS	0.21	0.7	11	0.8	14
	FUEP + APS	0.21	0.7	12	0.9	18
	FUEP + FPS	0.24	0.8	17	0.9	24



(a) Original video sequence (b) Reconstructed (AUAP+APS)

(c) Reconstructed (AUEP+FPS) (d) Reconstructed (FUAP+APS)

Figure 14 Perceived video quality under three different scenarios for video sequence basket.

(a) Original sequence

(b) AUAP + APS scheme

(c) AUEP + FPS scheme

(d) FUAP + APS scheme

Figure 15 Zommed doted area of Fig. 14(a) for all reconstructed videos.

6.2.4. Perceived video quality evaluation

Snapshots for video sequence 1 under different transmission schemes are shown in Fig. 14. However, it's difficult to perceive the quality difference between snapshots because of small size, as the original video is 720_p. The doted area in Fig. 14(a) is zoomed in Fig. 15 for all snapshots. Now, we can see the difference between them. However, during performing tests and watching these reconstructed video on big screen, there was a clear quality difference between them. The results show that non-adaptive schemes are unable to cope with varying channel conditions.

6.2.5. Peak signal-to-noise ratio

Peak signal to noise ratio is a popular objective metric used to assess the quality of a video at receiving end, which is derived from the Root Mean Squared Error (RMSE). Hence, we calculated the PSNR by comparison of sender side original raw YUV format video sequence with receiver side processed raw YUV format video sequence using JSVM software. Table 7 shows the average PSNR results for both video sequences under different scenarios, while Figs. 16 and 17 show the PSNR graph for first 80 frames. It can be seen from both table and figure that our proposed adaptive unequal error protection and packet size assignment scheme achieves the best performance under varying channel conditions.

Table 7 Average PSNR results for both video sequences under different scenarios.

Video sequence	Scenario	Average PSNR		
		Y	U	V
Basket	AUEP + APS	37.4576	44.3071	44.9023
	AUEP + FPS	32.6601	38.2087	38.8089
	FUEP + APS	32.1002	38.3959	38.7899
Stadium	AUEP + APS	36.3423	43.7073	43.4432
	AUEP + FPS	30.3309	37.3055	37.3007
	FUEP + APS	31.0432	38.3971	38.4401

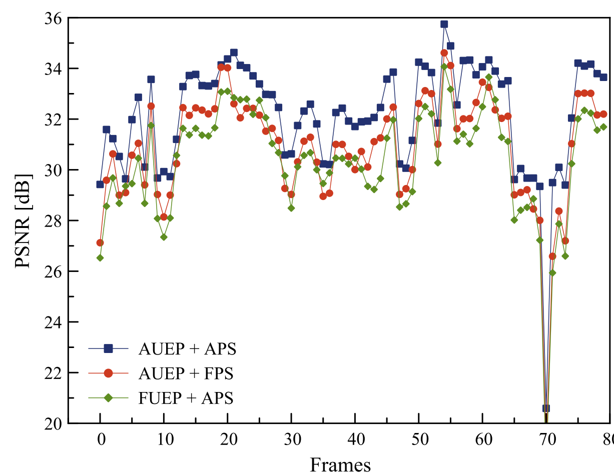


Figure 16 PSNR plot of first 80 frames under different scenarios for basket sequence.

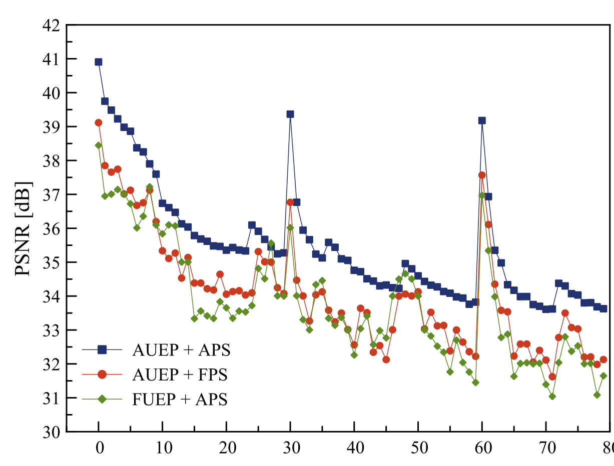


Figure 17 PSNR plot of first 80 frames under different scenarios for stadium sequence.

7. Conclusion

In this paper we proposed an interactive and ubiquitous video streaming scheme for Scalable Video Coding (SVC) based video streaming over error-prone WMNs towards heterogeneous receivers. Initially, the proposed scheme calculates the quality of all available paths using gray relational analysis and then based on quality of path it decides adaptively the size and level of error protection for all packets in order to combat the effect of losses on perceived video quality. The scheme is implemented and evaluated in real test-bed. The performance comparisons of proposed scheme with some other existing schemes were performed. After a series of repeatable experiments on the test-bed, our results show that the proposed streaming approach gives better performance compared to other existing schemes and can react to varying channel conditions with less degradation in video quality.

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