On the Quality Assessment of H.264/AVC Video under Seamless Handoffs

Ilias Politis, Tasos Dagiuklas, and Lampros Dounis

Dept. of Telecommunication Systems and Networks, TEI of Mesolonghi, National Road Nafpaktos-Antirrion, 30300, Greece
{Ilpolitis,dounis}@gmail.com, ntan@teimes.gr

Abstract. This paper examines the quality assessment of video streaming applications in a heterogeneous wireless environment, where the user hands off across inter-technology radio access networks. Three different scenarios have been considered: scenario with seamless handover using the media handover framework to initiate handover, seamless handoff combined with rate adaptation that is based on Rate-Distortion and seamless handoff with rate adaptation that is optimized using network bandwidth and packet loss parameters. The results from two video sequences have shown that both objective quality evaluation and the subjective evaluation (double stimulus-SDSCE, DSCQS) are optimized under the combined seamless handover and rate adaptation functionalities.

Keywords: Mobile Video Delivery, Subjective and Objective Video Quality Evaluation.

1 Introduction

Due to rapid growth of wireless communications, multimedia applications are becoming increasingly popular. In the recent years, this progress has also been aided by the proliferation of technologies such as 3G/3G+/LTE/WiFi, and the trend has been to allocate these services more and more also on mobile users [1]. Video transmission over heterogeneous wireless networks poses many challenges, including the issue of coping with losses due to physical impairments and network congestion, as well as maintenance of Quality of Service (QoS) and session continuity [2]. A key change is to support and maintain video quality while the user moves across heterogeneous networks. It has been reported that in order to support and maintain video quality, handover functions may be triggered not only from the physical layer but also from the network (rapid increase to packet loss) and the application layer (PSNR drops substantially).

The aim of this paper is to evaluate the quality of video streaming applications in different mobility scenarios using both objective and subjective methodologies. Three different scenarios have been considered: The first one considers an on-going video session that is seamless transferred in a vertical handoff function under the Media Independent Handover (MIH) functionality. In the second scenario, handover is

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combined with rate adaptation using the Rate-Distortion functionality of the encoder. In the third scenario handover is combined with a novel rate adaptation that is based on the optimizing technique by taking into account the available bandwidth of the new access network and the packet loss.

The paper is structured as follows: Seamless Handoff and Video Rate Adaptation are presented in section 2; section 3 presents the testbed platform in order to carry out the experimentation results. Subjective evaluation using double stimulus methodologies is presented in section 4. Conclusions are presented in section 5.

2 Seamless Handoff

Handover is the process of network association of a Mobile Terminal while it moves across different access point [3]. Handover aims to accomplish the following goals:

- Nomadicity: It is the ability of the user to change his network point of attachment while he/she is on the move.
- Session Continuity: It is the ability that the mobile terminal can switch to a new network point of attachment while maintaining the ongoing session towards the new point of attachment.
- Seamless handoff: It aims to minimize the packet loss while the session is associated with the new point of attachment. It is sometimes referred to as smooth handoff.

One of the standards that have specified a framework for seamless mobility is the IEEE 802.21 standard. This section describes a seamless mobility framework that combines the Media Independent Handover framework (MIH), which is responsible for seamless handover management across heterogeneous access networks and the available BW based rate control scheme, described above.

2.1 Media Independent Handover Framework Overview

IEEE created the 802.21 standard in order to challenge one of the main issues in wireless mobility, seamless handovers across inter-technology RATs (Radio Access Technology) networks [4]. In particularly, mobility protocols such as Mobile IP are suffering from sensible latency and they have not knowledge about the application layer parameters and candidate network conditions. IEEE 802.21 proposes the MIH framework where mobile nodes and the network exchange information and commands for an optimal handover. Moreover, for hiding the heterogeneity of the MAC and physical layers, MIH inserts an intermediate layer between layer 3 (and above) and the divert Layer 2 technology specifics, the Media Independent Handover Function (MIHF). The MIH framework describes three different types of communication that act as services: Event Service, Command Service and Information Service, as illustrated in Fig. 1.

The Media Independent Event Service (MIES) is a communication procedure where indications for handoff (events) are passed to the MIH users for further handling. The Media Independent Command Service (MICS) provides a set of handover commands in order for the MIH users to be able to implement their handover decisions. The Media Independent Information Service (MIIS) is a database that contains all the available information about the network ranging from channel parameters to presence of application layer services. It is used by mobility protocols in order to find appropriate networks that can facilitate a handover.

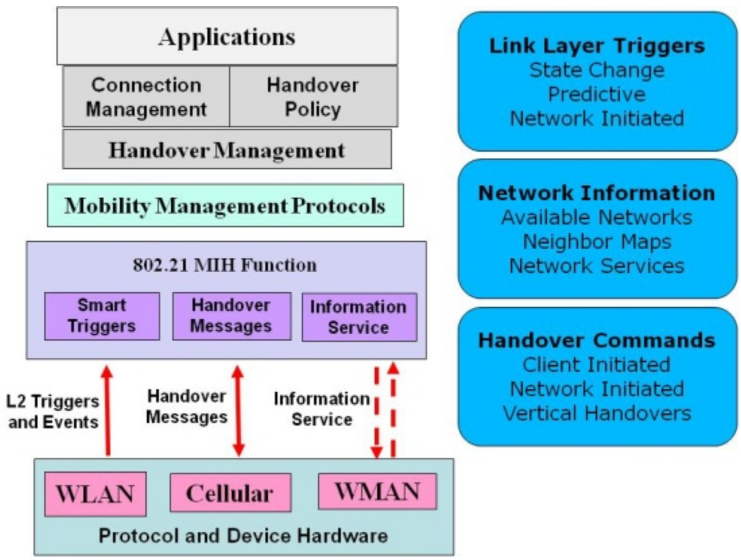

Fig. 1. IEEE 802.21 Media Independent Handover framework

MIH can be considered as a co-operative decision making scheme for QoE-aware handoff policy. This means that triggers from both MT and the radio access networks for the handover decisions. There are scenarios for handover from QoE based triggering [5], [6].

- Network Load increases at the current wireless network leading to congestion and packet loss.
- Application Deteriorates. This can be verified through video QoS monitoring (MOS, PSNR) generating alarms when these parameters are sharply altered.

2.2 Application QoE Triggering

In contrast to previous studies that consider only physical and network layer statistics, this paper proposes a handover functionality that can be triggered also by estimating PSNR. Monitoring of QoE is based on the RTP Control Protocol Extended Report (RTCP-XR) as defined in [7], which provides a useful set of fields providing information for video performance analysis. Important information that is related to video is the following:

- The packet loss within I/B/P frames.
- Knowledge of GoP structure and key coding parameters to estimate PSNR

The process of estimating the PSNR of real time video streaming requires the video client to send an RTCP-XR report to the application server with the ID numbers of the lost RTP packets, per video frame. The application server can then estimate the current value of PSNR by a video distortion model that calculates the distortion of the received video due to packet losses in real time. Specifically, the proposed distortion prediction model is a recursive formula that takes into account the correlation among video frames during the intra-frame period. The distortion model incorporates the random behavior of losses in the wireless medium (isolated losses, burst of losses, losses separated by a lag). More information about the predicted distortion model and Video QoS Monitoring can be found in [8] and [9].

2.3 Vertical Handoff Policy

A handover scheme, as part of the MIH framework is responsible for deciding whether a handover is needed based on physical, network and application layer statistics, collected from both the MT and the access networks. The handover policy includes three phases [4]:

- Decision phase – all the handover related information (e.g. Signal-to-Noise-Ratio, delay, jitter, Peak Signal-to-Noise-Ratio, packet loss, etc), from the mobile terminal, the currently selected access network and the already discovered neighboring networks are retrieved from the MIIS through the MIH entity.
- Initiation phase – the collected parameters are evaluated and compared against a set of predetermined threshold values. These thresholds are either determined by the network provider or are specified in the user profile. The MIES service is responsible for comparing the collected statistics with the threshold values and for informing the command service when one or more thresholds are violated and the handover criteria are matched.
- Execution phases – the MIH triggers the Mobile IP module which is responsible for performing the actual handover and bidding with the new point of attachment, ensuring seamless service continuity.

2.4 Network Selection

In the context of MIH, it is necessary to incorporate a mechanism that selects the access technology that is the most suitable according to the needs of the user at each moment. This decision is based on QoS parameters/criteria that must be optimized depending on the available access networks. In this paper, the network selection scheme combines two Multi Attribute Decision Making (MADM) algorithms methods, the Analytic Hierarchy Process (AHP) method [10] and the Total Order Preference by Similarity to the Ideal Solution (TOPSIS) method [11]. The first one determines weights of the criteria and the second one calculates the final access network ranking.

AHP Method

In the case of AHP, the following parameters are considered: Throughput, Packet Loss and SNR. All the handover parameters are compared pairwise according to their levels of influence with respect and the comparison results are inserted in a square matrix using the following rule: A=[aij]nxn where n are the number of factors. Each element represents a handover parameter with a value that implies the extent, at which an element is more important to another from 1 to 9. The value 1 defines equal importance between the two elements and value 9 defines extreme importance.

The weights vector w is calculated through the following repetitive process:

- The elements of each line of the matrix are added up: $s_i = \sum a_{ij}$ for each i.
- In each line of matrix, the weight of each element is estimated by calculating the quotient of the value s_i via the sum of all elements of the matrix: $w_i = s_i / \sum \sum a_{ij}$. The elements of the received vector w are normalized, so that their sum equals to 1.
- The square of the matrix is calculated and all the procedure steps are repeated until two successive approaches do not differ considerably in the frame of the desirable precision.

TOPSIS Method

In TOPSIS method, the best radio access must have the shortest distance from the positive ideal solution and the longest distance from the negative ideal solution. It comprise the following steps:

- Based on the scores achieved for each one of the selected criteria (attributes), the Network Matrix is expressed as:

$$NW_{ij} = \begin{array}{c} A_1 \\ A_2 \\ M \\ A_m \\ \\ \end{array} \begin{array}{c} \begin{bmatrix} d_{11} & d_{12} & \cdots & d_{1m} \\ d_{21} & d_{22} & \cdots & d_{2m} \\ M & M & \cdots & M \\ A_{m1} & A_{m2} & \cdots & A_{mm} \end{bmatrix} \\ \begin{matrix} C_1 & C_2 & \cdots & C_m \end{matrix} \end{array}$$

  where A_1, A_2, ... , A_n are the possible network alternatives and C_1, C_2, …, C_m are the criteria, which measure the performance of the alternatives. Each element dij of the Network Matrix NW_{ij} is the rating of the alternative Ai with respect to the criterion C_j.

- The normalized value of r_{ij} is computed as: $r_{ij} = \dfrac{d_{ij}}{\sqrt{\sum_{i=1}^{n} d_{ij}^2}}$ where i={1,2, …,n}, j={1,2, …, m}

- The normalized weights are determined according to the following: $NW_{normij} = NW_{ij} * r_{ij} * w_j$
- Determination of the positive and negative ideal solutions using the following formula: $A^+ = \{\max_j (NWnorm_{ij}) \mid j \in I_b, \min_j (NWnorm_{ij}) \mid j \in I_c\}$ and $A^- = \{\min_j (NWnorm_{ij}) \mid j \in I_b, \max_j (NWnorm_{ij}) \mid j \in I_c\}$ where I_b denotes the set with the benefit criteria, and I_c denotes the set with the cost criteria
- The distance of each alternative from the ideal and the negative ideal solution is given by the following formulas: $S_{+i} = \sqrt{\left(NWnorm_{ij} - NWnorm_j^+\right)^2}$ and $S_{-i} = \sqrt{\left(NWnorm_{ij} - NWnorm_j^-\right)^2}$
- The relative closeness determines the relative closeness of each alternative Ai from the ideal solution using the following formula: $C_i = \frac{S_i^+}{S_i^+ + S_i^-}$, i={1,…,n}
- The best candidate networks are ranked according to the Ci values in descending order.

2.5 Seamless Handoff and Rate Adaptation

Seamless handover can be combined with rate adaptation in order to optimize QoS/QoE of an on-going video session. In case, where the mobile user moves to a network with less available bandwidth than the current one, seamless handoff can be combined with rate control so that QoE/QoS is optimized. This can be transformed to an optimization problem (determine the best Quantization Parameter that optimizes PSNR under certain networking conditions) [12].

In the context of this paper, a rate control scheme has been used that maximises perceived video quality based on the currently available bandwidth (BW). The bandwidth availability is estimated based on RTCP feedback from the mobile terminal.

The proposed real time video rate control framework requires that pre-encoded video sequences have been tested over different network condition. This is required in order to extract important statistical information. Such statistical information regards the relationship between encoding distortion and QP, sending bit rate and QP. The above relationships are obtained through experimentation.

A Rate Control Module (RCM) is defined as an entity within the video encoder that stores the aforementioned statistical information, collects real-time information from the network and sends feedback to the encoder in order to control the sending rate. Without loss of generality, we assume that this control is applied to each video frame. RCM receives periodic feedback with information regarding current available

BW and packet losses, and decides upon the optimum QP value that maximizes the perceived video QoS (PSNR at the decoder). RCM's decision is forwarded to the video encoder, which selects the optimum QP parameter, as shown in Fig. 2.

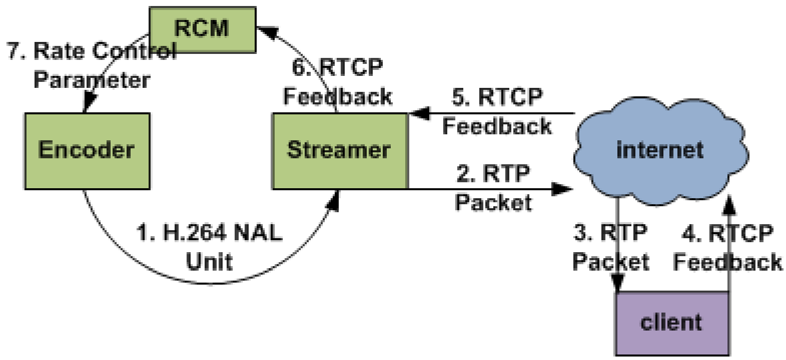

Fig. 2. Video Rate Control Scheme

The perceived video QoS in terms of PSNR is maximized by selecting the optimum value of the QP parameter, according to the current available bandwidth (BW). Without loss of generality, the term available bandwidth refers to the capacity of the access network that becomes available to the user. That is the bottleneck between the video encoder and the end-user. This network capacity is periodically monitored by probing the network with a predefined stream of dummy RTP packets. The user is informed for the available network capacity (or available BW) by the periodic RTCP messages that carry this information to the message header. In order to determine the optimum QP, the RCM is based on pre-stored PSNR versus QP data for different BW conditions [12].

This relationship is illustrated in Fig. 3, where a video sequence has been encoded at different QP values and is transmitted multiple times over a network with varying load conditions. It is evident that for low available bandwidth, perceived PSNR increases with QP, due to the fact that larger QP results in lower video rate. The PSNR reaches a certain peak value, which is the maximum perceived quality that a video user can have for specific network conditions. This is the point where both coding distortion and packet loss have the least impact on the perceived video quality for a given available bandwidth. Any further increase to the QP value will result in higher coding distortion and smaller video transmission rates (packet loss increases) that deteriorates the perceived video QoS. As the available BW becomes higher, PSNR reaches its peak value earlier (i.e. at smaller QP), shifting this point towards the left. Furthermore, Fig. 4 illustrates the surface fitted model of Fig. 3. This curve is true for one particular video sequence (NTIA gold fish pond) [13], however, the algorithm can be extended to other video sequences as well.

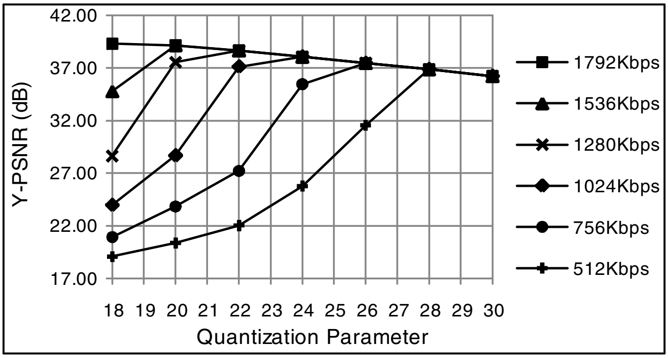

Fig. 3. Perceived PSNR vs. QP under different network conditions (available BW for video transmission)

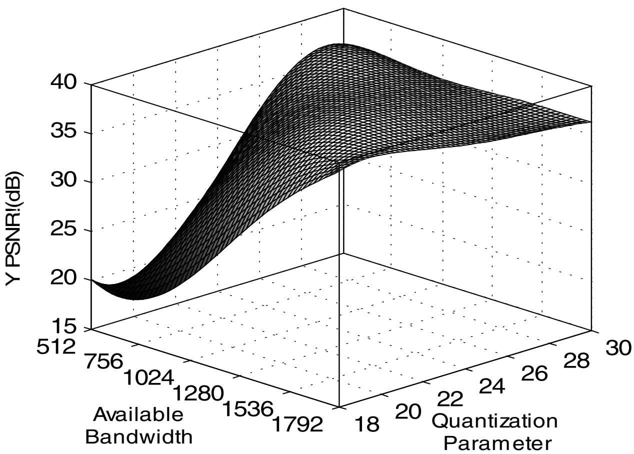

Fig. 4. Surface fitted model of perceived PSNR vs. QP under different network conditions (available BW for video transmission)

For the surface-fitting model a number of different polynomials and rational equations have been used, which resulted in the following polynomial equation with a reasonable fitting goodness.

$$PSNR = 605 - 1.2*BW - 57.6*QP + 4*10^{-3}*BW^2 + 0.12*BW*QP + 1.26*QP^2 - 3.7*10^{-3}*BW*QP^2 + 0.02*QP^3 - 6*10^{-4}*QP^4 \quad (1)$$

According to the BW based rate control algorithm, both the currently available BW and the PSNR versus QP relationship over different available BWs are regarded as input to the algorithm. BW conditions are collected from RCM via RTCP reports [14]. The algorithm optimizes the perceived PSNR by selecting the optimum QP value according to the network conditions. Depending on network conditions optimum QP can either be higher or lower than the current QP. As the network load increases (decreases), QP should increase (decrease) so that the sending rate is adapted according to the available bandwidth. The solution of the partial derivative of the three-dimensional function of Fig. 4 (Eq. 1), with respect to the QP given that the available BW information is collected by RTCP, returns the critical points (QP values) that maximize the perceived PSNR. Fig. 5 outlines the proposed BW based rate control algorithm.

Input
 # BW → Available bandwidth
#F(BW,QP) → Function describes the surface of the Bandwidth based rate control algorithm.

Output
#bestQP →Best Quantization Parameter for available BW
#maxPSNR → Estimated average PSNR for Best QP

$$[solutions] = \frac{\partial}{\partial QP} F(BW, QP) = 0$$
$$\forall iQP \in solutions$$
$$if\ iQP > 0\ \&\ \frac{\partial}{\partial iQP^2} F(BW, iQP) < 0$$
$$bestQP = iQP$$

Fig. 5. Proposed available BW based rate control scheme

Another approach of rate control is to use a Rate Distortion Model that is inherently implemented within the video encoder [15]. In this model, the targeted bit rate is provided to the encoder in order to select on the fly the appropriate QP parameter using the Rate Distortion Model. In this approach, when the user is handed over to a wireless network with less bandwidth than the current one, a signal is sent towards the encoder to adapt QP parameter accordingly.

3 Testbed Platform

In order to study the perceived video quality, an experimental test-bed was implemented.

- Two fully configurable 802.11e access points and one 3G access that allow the monitoring and collection of physical and network layer statistics.
- A MIH capable mobile terminal that can connect to any access network through two corresponding adapters. The MT monitors the status of the current connection and the availability of any other candidate RATs in each vicinity.

This information is reported to the MIH (MIIS) through a client-server application, and based on the decision from the vertical handover functions it will be instructed by the MIP core

- A MIH server that hosts all MIH services MIIS, MIES, MICS as well as, the handover decision functionality
- A video server that consists of a fully configurable H.264/Advance Video Coding encoder and streamer [15], capable of exchanging RTCP-XR messages through a clients server application
- During the experiments, the network is stressed with background traffic based on a statistical video traffic model, which regards a number of multiplexed homogeneous and mutually independent video sources that transmit simultaneously. This model can accurately simulate the effect of aggregate video traffic from multiple video sources. Moreover, Dummynet [16] is used in order to emulate packet losses and network load.

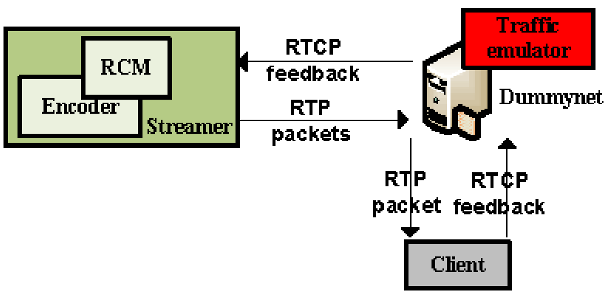

Fig. 6. Test-bed Platform

Fig. 7 depicts the objective quality (PSNR) of the 'Fish' video sequence temporal basis [13]. We consider that both WLANs and 3G Network are stressed with background traffic ranging from 50% to 75% of the total capacity.

The paper focuses on measuring the effect of seamless handoff on the perceived video quality. To this end three test-bed experiments have been carried out in a control laboratory environment. The handover is initiated by both application layer and network layer triggers (including PSNR drop and Packet loss increase due to mobility). The seamless handoff is executed by the MIH platform, described above.

In the first scenario, the impact of seamless handoff functionality on the perceived video quality is presented. Two other scenarios are considered where seamless handoff is combined with rate control. The first one considers the optimization functionality presented in section 2. The second one considers a rate distortion functionality that is inherent in the video encoder. It is obvious that the seamless handoff functionality and optimized rate control gives the optimum perceived video quality.

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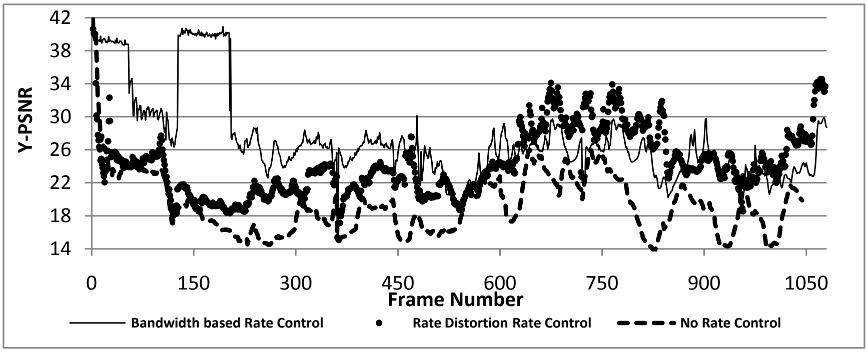


Fig. 7. Objective Video QoS for three video mobility scenarios

4 Subjective Evaluation

The subjective quality evaluation tests have been carried out using two different video sequences with the same spatial resolutions. The tests have been carried out using the recommendations by ITU-T BT.500 for laboratory environments [17]. The following parameters have been considered: daylight conditions, mid gray background using appropriate curtains. High quality LCD displays have been used for the subjective evaluation [18], [19].

We have used two high-quality raw videos that have been recorded with professional equipment using YUV 4:2:0 format. All videos are freely available for downloading from the Internet.

All subjects that participated at the evaluation are undergraduate students and faculty members of the TEI of Mesolonghi, Department of Telecommunication Systems and Networks, Greece[1]. A total of 80 subjects have been used to evaluate the videos. During the test setup phase, each subject gets familiar with scoring procedure and video artifacts. This will ensure that subjects will get familiar with the testing procedure and score video artifacts accordingly. Training videos have been used for this purpose. Both training and test videos have been impaired.

The subjective evaluation study uses simultaneous double stimulus continuous evaluation that is described below. Under Double Stimulus framework, two video sequences are shown simultaneously to each subject. The first one is the original video sequence and the second one is the transmitted in a heterogeneous wireless environment. The subject is informed about the presence of the reference video (Stimulus A) and the distorted video (stimulus B), so that he/she continuously evaluates the test material.

4.1 SDSCE

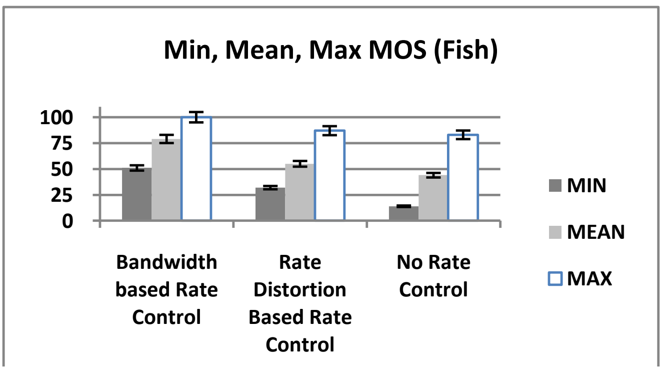
The votes are sampled every 0.5, as described from the Simultaneous Double Stimulus for Continuous Evaluation (SDSCE) in ITU BT-500. Two different

[1] `http://www.tesyd.teimes.gr/cones`

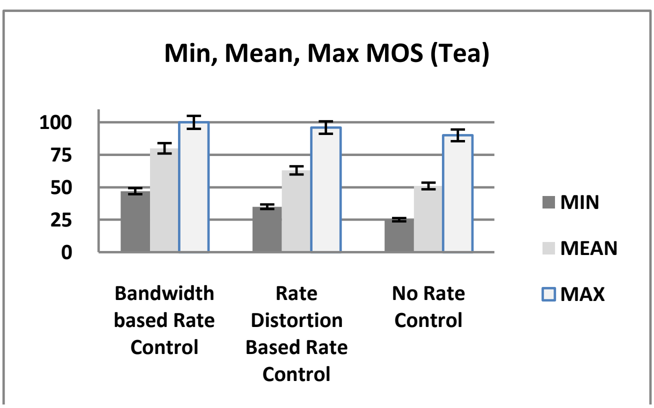
impaired video segments haven been evaluated. Each impaired video segment corresponds to a different wireless video transmission policy. The aim to evaluate the video sequences under the aforementioned framework: wireless video transmission in a heterogeneous wireless environment, wireless video transmission + rate adaptation (using the R-D of the encoder and the proposed bandwidth adaptation algorithm).

In this voting procedure, the next step regards the removal of vote outliers. These votes refer to the cases where the difference between mean subject vote and the mean vote for this test case from all other subjects exceeds 15%. This is a general rule that has been also used in other research works [18], [19].

The following figure illustrates the max, min and average MOS for both “Fish” and “Tea” video sequences. The best quality is experienced under the Bandwidth Based Rate Control, which is the seamless handover using the optimised rate adaptation functionality and the worst scenario is the one where the seamless handover is not combined with rate adaptation.



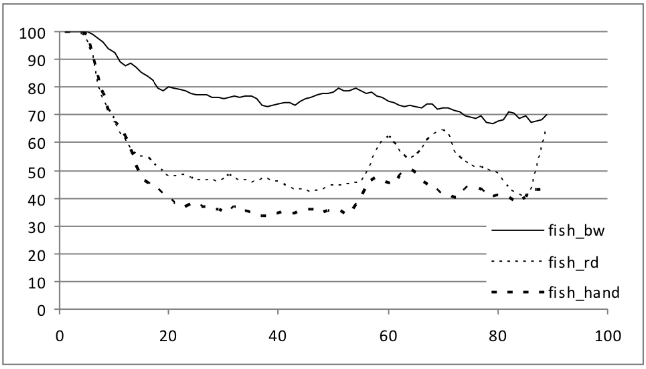
(a) ‘Fish’ Video Sequence Min, Mean, Max MOS (SDSCE)



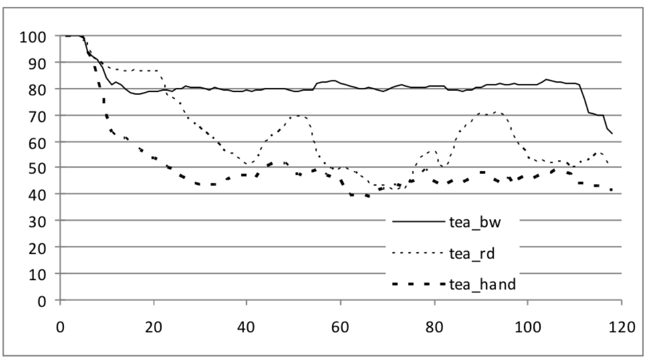
(b) ‘Tea’ Video Sequence Min, Mean, Max MOS (SDSCQE)

Fig. 8. Max, Mean and Min MOS under the three handover schemes

The above observations are verified by illustrating average MOS over frame for the three scenarios. Under the seamless handover scenario, the MT may switch to a network with much less available bandwidth. In the scenario, where rate adaptation is carried out using Rate Distortion, there is an improvement in the MOS due to the fact that video encoding is adapted to bandwidth constraints. This improvement is optimized when using novel rate control functionality.



(a) 'Fish' Video MOS over time (SDSCE)



(b) 'Tea' Video MOS over time (SDSCE)

Fig. 9. MOS versus time under the three different schemes

4.2 DSCQS

Within the Double Stimulus Continuous Quality Scale (DSCQS) methodology, two consecutive presentations of two stimuli take places [17]. There is a 10s duration of the reference and the test (distorted) video, separated by 1s grey frame. The above procedure is repeated three times, and at the last round the subject must vote for both the reference and the test video in the scale 0 to 100.

Similar to the above procedure (SDSCE), the outliers are removed in order to remove subjects whose scores deviate considerably from the votes of the other subjects. Under this methodology, the Differential MOS computed using the above formula: $DMOS_i = \frac{\sum_{j=1}^{N}\left(s_{ji}^{A} - s_{ji}^{B}\right)}{N}$, where N is the number of valid subjects and s_{ji}^{A}, s_{ji}^{B}, are the scores of the test and the reference video respectively. Using the DMOS, MOS can be computed for the i test using the following formula: $\left(MOS\right)_i = \frac{100-\left(DMOS\right)_i}{10}$

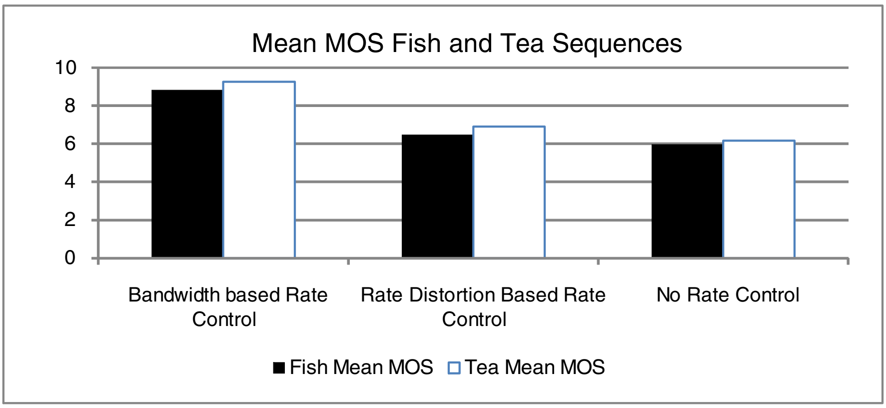


Fig. 10. Average MOS using DSCQS Methodology

5 Conclusions

This paper examines subjective and objective evaluation of video sequences under seamless handoff schemes. Handover decision is based on the Media Independent Handover Framework by collecting information from physical, network layer and application layer from both MT and network entities. Video Quality has been evaluated using three different cases: in the first cases video is received by considering only mobility function, in the second case mobility is combined with rate adaptation that uses Rate-Distortion function of the video encoder and in the third case mobility is combined with rate adaptation that is optimized by taking into account available bandwidth and packet loss of the network that the MT will handoff. Through experimentation from a testbed, both subjective and objective quality is optimized when handover is combined with rate control.

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