Quality of Experience Adaptation Controllers for Voice and Video in Wireless Networks

David Rodrigues[1], Eduardo Cerqueira[2] and Edmundo Monteiro[1]

[1] University of Coimbra, Pinhal de Marrocos, 3030-290 Coimbra, Portugal
{drod, edmundo}@dei.uc.pt
[2] Federal University of Para, Rua Augusto Correa, 01, 66075-110 Belem, Brazil
cerqueira@ufpa.br

Abstract. Real-time multimedia will be among the most important applications in next generation mobile networks. However, the efficient delivery of these applications to guarantee Quality of Experience (QoE) to fixed and mobile users in multimedia-aware networks is still a research challenge. This paper proposes new approaches to keep voice and video applications with an acceptable quality level in mobile wireless networks with scarce resources and investigates the impact of different video coding parameters on the QoE of voice sessions and real video sequences. Simulation experiments were carried out in a mobile multimedia system to show the benefits of the proposed QoE optimizations, by analyzing well-known QoE objective and subjective metrics

Keywords: Quality of Service, Quality of Experience, Multimedia, Wireless

1 Introduction

In recent years, the advancement of mobile communications and the popularization of multimedia services have emerged to offer a novel and comfortable living style for costumers. In this context, it is expected the delivery of mobile multimedia content anytime, anywhere and with an acceptable quality level. In mobile systems, the IEEE 802.11, which defines a Medium Access Control (MAC) and several Physical Layers (PHYs), is the most popular wireless technology used in Wireless Local Area Networks (WLANs). However, due to the lack of Quality of Service (QoS) and Quality of Experience (QoE) support, the scientific community and the industry are researching new approaches to improve the quality level of multimedia delivery and user perception, while optimizing the usage of wireless network resources.

To cope with QoS issues in WLANs, the IEEE 802.11e working group was created, where the draft version [1] brought new MAC improvements incorporated in the IEEE 802.11 standard in 2007 [2]. To provide QoS assurance, eight User Priorities (UPs) were defined. Each packet is assigned to an UP and mapped to an Access Category (AC). Each AC is directly mapped to a queue, where several queues have different priorities and applications must be put to them according to requirements, policies, content characteristics, among other parameters.

Regarding multimedia video and voice encoding, several Coder-Decoders (CODECs) were developed to reduce the bandwidth required to distribute multimedia content. This improvement is specially needed in mobile networks with limited resources. The Moving Picture Experts Group (MPEG) codifier is widely used and employs a structure composed of 3 frame types named *I*, *P* and *B* as shown in Figure 1 [3][4]. *I* frames are encoded using spatial compression and without reference to other frames in the sequence. To achieve temporal compression, *P* frames (predicted frames) are reconstructed using motion prediction from the last *I* or *P* frame. As a result, *P* frames have a better compression ratio than *I* frames but depending on the amount of motion present in the sequence. *B* frames (bidirectional frames) archive the better compression ratio using prediction from the last and next *I* or *P* frame. The sequence of frames that depend on an I frames is called Group of Pictures (GOP).

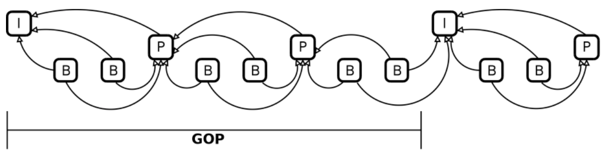


Fig. 1. MPEG Structure

To ensure end-to-end quality level control for multimedia content, new QoS schemes and metrics were created and developed to provide information about the network conditions, such as loss, delay and jitter. However, these QoS metrics only provide information from the network (packet) point of view, as well as, QoS control solutions fail in subjectivity aspects related with human perception. To cope with these limitations and to assess the quality level of multimedia applications at the user-level, QoE metrics and controllers were introduced [5]. QoE metrics provide a deeper analysis of the flows, taking into account the specificity of each multimedia application, as for example, CODEC type, luminance and contrast, as well as, loss and delay tolerance [6-7].

This paper proposes and evaluates new QoE-based multimedia packet control approaches for IEEE 802.11e mobile networks. The proposed mechanisms were designed to optimize multimedia content delivery according to multimedia content characteristics, user perception and frame importance and dependence. In order to show the benefits of the proposed solutions on the user perception, simulation experiments were carried out in multimedia wireless scenarios. The results presented the efficiency of each QoE-aware approach when used in mobile environments with different audio flows and real video sequences, and network conditions.

The remainder of this paper is organized as follows. Section 2 presents the related work. Section 3 details the advanced QoE control mechanisms. Section 4 presents the simulation environment and the performance evaluation for each mechanism. Finally, Section 5 concludes the paper and describes the future work.

2 Related Work

Several voice and video-related solutions were proposed to work in systems with different Coder-Decoders (CODECs) and different wired/wireless QoS models. An IEEE 802.11 system to improve multimedia quality is proposed and tested in [8]. This proposal adds a second queue to Distributed Coordination Function (DCF) in order to give priority to multimedia streams, and therefore reduces their delay. However, this approach is obsolete and does not focus on the new Hybrid Coordination Function (HCF) operation mode introduced by the IEEE 802.11e draft, which already includes several classes and queues that allow different priorities. Furthermore, few details are provided about the simulations results and environments. The conclusions are only based on network performance measurements, such as delay and losses, and no user level assessments were accomplished.

Another approach uses a cross-layer architecture to map packets with different priorities to different Enhanced Distributed Channel Access (EDCA) queues of IEEE 802.11 [9]. However, this proposal is not in accordance with the recommendations of the IEEE 802.11 standard, as well as, additional traffic placed into the highest priority class may damages voice flows and no scenario related to this problem was analyzed. Furthermore, the mapping of packets from the same flow into different queues increases the jitter and generates losses. Finally, the simulations only take into accounting QoS metrics and no user-based measurements are performed.

A cross-layer video quality proposed approach is deployed in [10]. The signaling from the MAC layer to the upper layers provides further information to the encoder to adjust the video rate and quality to the current network conditions. Thus, in a major interference scenario, the encoder adapts its flows to a lower rate in order to reduce the loss and delay. However, because only local information from the MAC layer is used for video rate adaptation, adjustment cannot be done according to end-to-end path conditions. Therefore, it is not suitable for heterogeneous next generation mobile networks with different requirements and mobile devices.

In [11] a framework for MPEG video delivery over heterogeneous networks (Differentiated Service (DiffServ) and IEEE 802.11e) is analyzed. However, the proposed solution only uses QoS metrics and PSNR in the evaluation process and no HVS-based metrics are measured, failing in human perception assessments. Moreover, the mapping of packets from the same flow into different queues increases the jitter and generates losses

Video streaming traffic delivery in an IP/UMTS network is studied in [12]. In IP networks, video quality is enhanced by mapping video packets, according to their importance, to different traffic classes in a DiffServ environment. Packets with lower importance are mapped to queues with higher drop probability. When packets arrive to the UMTS network, DiffServ traffic classes are mapped to UMTS traffic classes. The evaluation was carried out taking into account a MPEG-4 scalable video stream. However, the paper only assumes the existence of a fixed and simple scenario and is only suitable for scalable MPEG-4 CODEC over DiffServ and UMTS networks, which reduce the implementation in Emerging Wireless Networks.

Other proposals aim to provide QoE-aware assessment in wireless systems for video and voice applications [13-16]. However, they are focused on fixed or/and wired scenarios, do not consider intra-frame dependence neither mobile IEEE 802.11 environments in their approaches.

The analysis of the related work has shown that QoE is a key topic for the success of next generation wireless systems, but current packet control approaches do not assure quality level support based on user perception in IEEE 802.11e networks neither mobility situations. Furthermore, the dependency of each frame of a sequence during the adaptation has rarely been addressed. Existing solutions also fail in capturing QoE metrics during their evaluation processes.

3. Advanced QoE Control Mechanisms

Traditional QoS traffic conditioners, as supported in current DiffServ, IEEE 802.11e and IEEE 802.16 QoS model, provide unfair and poor QoE-aware support for video and voice applications. For example, video streams are encoded so that different frame types have different importance in the final quality from the user point of view. Therefore, by taking into account this property, traffic conditioner elements (i.e., classifier, marker and dropper) can be configured to improve the user experience without increasing the usage of network resources or damaging other ongoing flows.

Multimedia flows are different in terms of encoder parameters, intra-frame dependence, as well as other QoS and QoE requirements. Hence, new mechanisms need to be used to improve the quality level of multimedia sessions and optimize network resources. In this context, multimedia QoE-based approaches are required to operate also in IEEE 802.11e that will be one of the most important access network technology in emerging wireless networks. The mechanisms need to be implemented taking into account the impact of multimedia flow on the end-user perception in congestion situations and to adapt traffic controllers according to the current network conditions, user experience, frame importance and dependence.

In order to satisfy these requirements, this section presents three mechanisms to improve video and voice quality level in wireless system, where the first two mechanisms are focused on advanced drop schemes based on intra-flow packet control, without changing the mapping of the different flows to the IEEE 802.11e access categories. The last mechanism improves the application quality, by mapping the different video frames to the existing queues in order to assure higher priority to frames with higher importance in congestion periods.

As illustrated in Figure 1, the MPEG encoding structure turns some frames more important than others. Because all frames in a GOP depend on *I* frames, these are the most important ones. *P* frames are also important, since part of the GOP depends on them. Finally, *B* frames can be dropped with minimal impact on the other frames as well as on the user experience.

Considering these proprieties, the Advanced Mechanism Priority Drop (AMPD) includes an advanced packet drop scheme that performs its multimedia quality level management based on the importance of each frame of CODECs. If a packet containing an *I* frame is marked to be dropped, it will be beneficial to check whether a packet corresponding to a *P* or *B* frame is currently in the buffer. Since the number of frames that depends on a *P* or *B* frame is reduced, the loss of these frame types will have a reduced impact on the final quality and therefore on the user perception. The same process happens when a *P* frame is marked to be dropped and a *B* frame is in the queue. It will be beneficial to drop the *B* frame that is already in the buffer and enqueue the incoming *P* frame.

Another mechanism, called The Advanced Mechanism Broken Drop (AMBD) aims to keep multimedia applications with an acceptable quality level, while reducing the waste of network resources based on the management of intra-frames dependency and user perception. AMBD recovers a packet marked to be discarded if there is a packet in the queue containing a frame with broken dependencies, that is, a frame that cannot be completely reconstructed on the receiver side. As shown in Figure 2, if the *P2* frame is lost, then *B3*, *B4*, *B5* and *B6* cannot be totally reconstructed by the receiver and will waste scarce wireless resources. Therefore, if the *P3* frame is marked to be drop and the *P2* frame has been dropped before, it is preferable to verify if there is a packet in the queue that contains a *B2*, *B3*, *B4*, *B5* or *B6* frame to be discarded and enqueue the incoming packet with the *P3* frame.

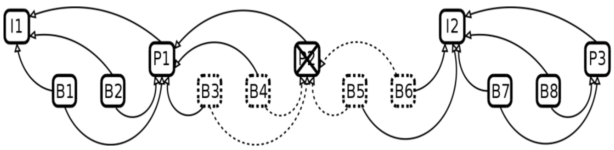


Fig. 2: MPEG Structure with broken dependences

Taking the frame dependence into account, when a packet is marked to be dropped, it is checked if it has broken dependencies. If so, it is dropped. If not, a packet with broken dependencies is searched in the queue. If it is found, the packet from the queue is dropped and the incoming packet is enqueued. Otherwise, the incoming packet is rejected.

Finally, The Advanced Mechanism Split Flow (AMSF) mechanism offers an approach to improve the video streams quality using an enhanced multimedia mapping process, which enqueues different packets to distinct ACs according to the different frames importance. At least, the high priority applications and frames will be allocated into most important classes. The remainder frames are mapped to a less significant class. This method can be used when the packet re-ordering is not crucial. For instance, it can be suitable for scheduled video and audio, where it is more important to ensure an intelligible audio flow than a perfect video.

As suggested in Table 1, priority packets containing *I* frames are mapped to AC_VO, intermediate priority packets carrying *P* frames are mapped to AC_VI and packets with *B* frames, and therefore less important, are mapped to AC_BE. The mapping of other flows remains as defined in the IEEE 802.11e standard.

Table 1: Flows & Queues Mapping in AMSF

Priority	Flow	Access Category
Highest	VoIP and I-Frames	AC_VO
	P-Frames	AC_VI
	Best-Effort and B-Frames	AC_BE
Lowest	Background	AC_BK

Depending on local policies, agreements and available wireless resources in each class, the mapping of frames can be configured to operate in a different way to avoid the session blocking and the re-ordering of packets, for example, by mapping all frames of a multimedia application to a less suitable class in congestion periods.

4 Performance Evaluation

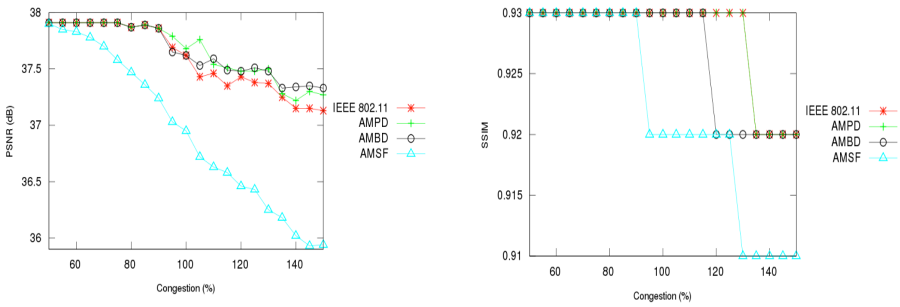
This section presents the simulation scenarios, results and discussions of the proposed QoE-aware control mechanisms. First, an overall comparison of the mechanisms is performed. Afterwards, a depth analysis of the mechanism with the best performance, i.e. AMPD, is performed. The objective of the experiments is to verify the impact of each mechanism on the user experience in mobile networks, by measuring objective (Peak Signal to Noise Ratio (PSNR) and Structural Similarity (SSIM)) and subjective QoE metrics (Mean Opinion Score (MOS))

The PSNR is the most traditional objective metric and compares frame by frame the quality of the video received by the user with the original one. The value of PSNR is expressed in dB (decibels). For a video to be considered with good quality according to the MOS scale, it should have an average PSNR of at least 30dB. The SSIM is a measurement of the video structural distortion, trying to get a better correlation with the user's subjective impression where values vary between 0 and 1, where 1 is the best value.

In order to define a scenario similar to a real system, simulations were carried out, where the Network Simulator 2 (NS2) with real video sequences in IEEE 802.11e environments. Experiments with Mobile IP (MIP) were used to verify the overall improvements of each mechanism on the performance of the mobile network. In each scenario, four flows were used: VoIP, video, best-effort and background traffic. Each flow type was mapped to a different IEEE 802.11e AC in accordance with its priority: VoIP flow to AC_VO, video stream to AC_VI (except for AMSF), best-effort traffic to AC_BE and background traffic to AC_BK.

Two well-known Common Intermediate Format (CIF) video sequences, News and Foreman, were used with 30 frames per second (fps), 2 *B* frames between each *P* frames, a bitrate of 320 kbps and a GOP size of 10. These values were changed for the AMPD analysis. Additionally, the Evalvid video control tool [17] was used to add real video sequence support to NS2. Regarding VoIP flows, they were encoded using the ITU-T G.729 standard, with a rate of 8 kbps and without Voice Activity Detection (VAD). Audio file support for NS2 was included using the VoIP packet trace module implemented by the Technical University of Berlin [18]. Best-effort and background flows were simulated with Constant Bit Rate (CBR).

QoE-aware simulation results for the video flows with different wireless congestion rates are depicted in Figures 3(a) and 3(b), for PSNR and SSIM assessments, respectively.



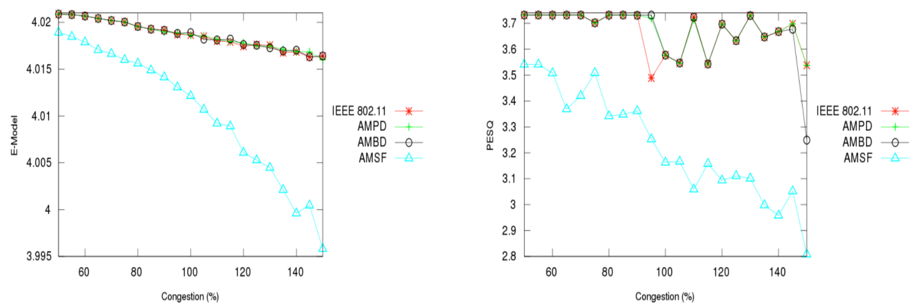
(a) PSNR (b) SSIM

Fig. 3: QoE assessment for the video flows with different congestion rates

Regarding IEEE 802.11e, the packet loss rate remains low despite of high values of congestion generated by background traffic in the network, which aims to assure quality level support for video flows. Only flows with lower priority, this is, background and best-effort traffics, are strongly affected. Therefore, the QoE assessments reveal a good quality of video for all values of congestion. However, the IEEE 802.11e quality of service controller does not distinguish between *I*, *P* and *B* frames and random values of loss exist for the different frames types. Therefore, video quality improvements can be achieved if the importance of each frame type is taken into consideration.

AMPD achieves a better *I* frame protection owing to the advanced drop mechanism. Thus, for all values of congestion, no *I* frame packets have been discarded and the best values of PSNR for all values of congestion are achieved. Despite a large number of dropped frames, including *I* frames, AMBD shows intermediate values of quality thanks to the discard of packets with broken dependencies, this is, packets that contain frames which cannot totally be reconstructed by the decoder on the mobile device. Finally, AMSF shows no loss in terms of *I* and *P* frames due to low congestion values achieved by the frames division over different queues. However, because *B* frames are mapped to a queue with less priority and in concurrence with best-effort traffic, loss rates for this type of frames raise, reaching values up to 80% when the congestion is of 150%. User experience is highly affected by such values, presenting lower quality assessments than those achieved with traditional IEEE 802.11e quality of service support. Based on the mapping of PSNR values to MOS, AMPD keeps multimedia applications with excellent quality level during all experiments.

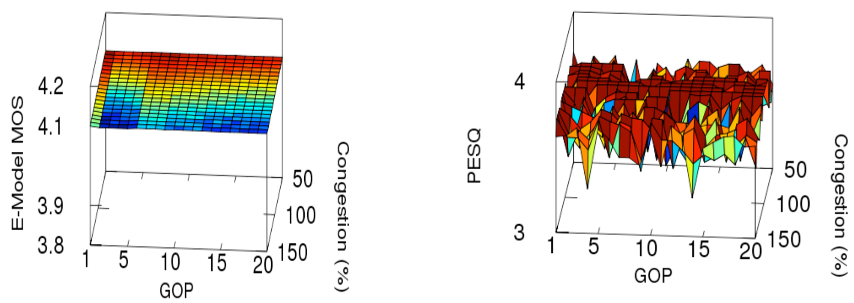
Regarding VoIP flows, E-Model and PESQ evaluations are depicted in Figures 4(a) and 4(b), respectively. Because no packet loss occurs in the VoIP traffic, as it has the highest priority and enough network resources, IEEE 802.11e only shows a small decrease in VoIP quality level due to the delay and jitter increase with the network load. Moreover, because AMPD and AMBD only affect video traffic, VoIP quality for these mechanisms is equal to IEEE 802.11e. However, besides the video quality degradation for AMSF, VoIP quality is also affected for this mechanism. Because *I* frames are mapped to AC_VO, the delay and jitter for voice traffic increase, leading to a degradation in VoIP quality.



(a) E-Model MOS (b) PESQ

Fig. 4: QoE assessment for the VoIP flows

For all values of GOP size used for video encoding and network congestion, no packet loss occurs in the VoIP traffic, as it has the highest priority and enough network resources. However, Figures 5(a) and 5(b) show a decrease in the quality assessed with the increase of the congestion. This decrease occurs only due to the increase of the delay and jitter with the network load. However, the quality decrease is minimal due to the protection achieved by the IEEE 802.11e QoS support. Different values of GOP size do not affect the voice perception either. Because video traffic is separated from VoIP, a change in video encoding does not affect the voice quality.



(a) E-Model MOS (b) PESQ

Fig. 5: QoE assessments for VoIP for different values of GOP and congestion for AMPD

The same conclusions are achieved for different numbers of *B* frames and bitrate used for video encoding, as shown in Figures 6 and 7, respectively. Only the increase in congestion leads to a decrease in the voice quality due to the increase of delay and jitter.

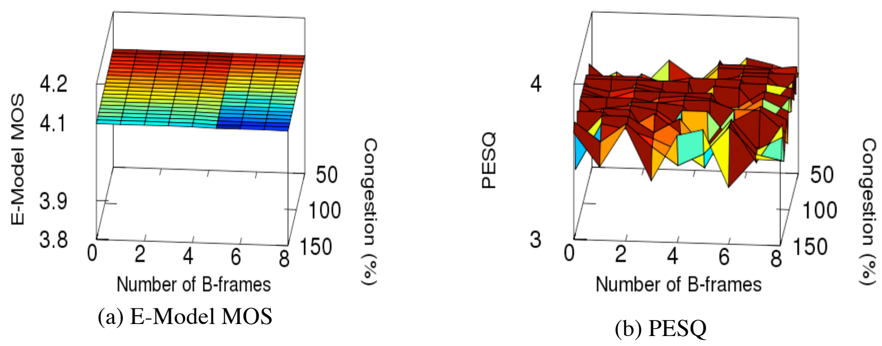


Fig. 6: QoE assessments for VoIP flows for different values of B-frames and congestion for AMPD

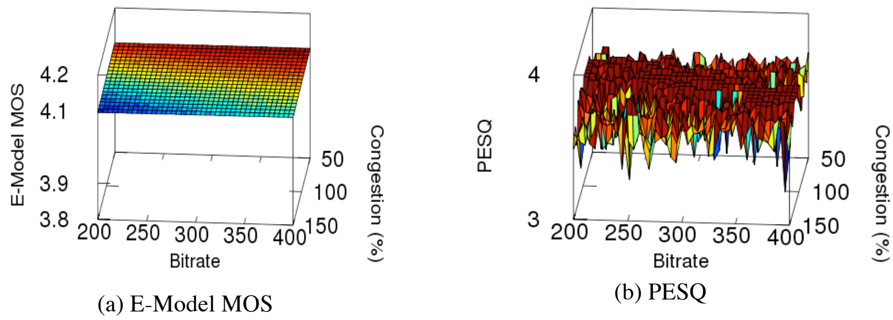
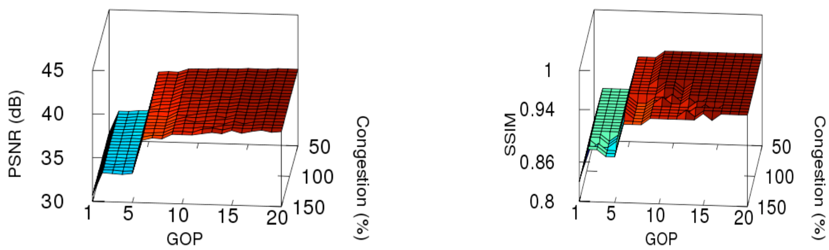


Fig. 7: QoE assessments for VoIP for different values of bitrate and congestion for AMPD

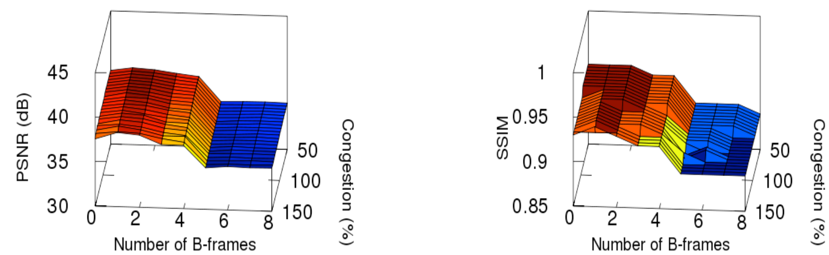
Regarding the video quality level, it varies according to the GOP size, the number of *B* frames used and bitrate. In Figures 8(a) and 8(b), the GOP size impact in user's experience is visible. For GOP values less than 5 the video quality is low, as it is evidenced by PSNR and SSIM evaluations. When small GOP size values are used, the quantity of *P* and *B* frames generated is low and therefore compression too. Thus, for the same data rate and lower compression, the video quality achieved by the encoder is lower. This is visible when the video quality is low despite the low network congestion. When large values of GOP size are used, and therefore when the number of *P* and *B* frames is high in relation to the number of *I* frames, the number of dropped packets with *I* frames reaches zero, because the advanced drop mechanism protects this important frame type. The same happens with the number of packet dropped with *P* frames. By increasing the GOP size, the number of *B* frames is increased and the drop of other frames becomes less probable. Evaluation results demonstrate best results for GOP sizes higher than 10. Because the most important frames are protected there are less broken dependencies and therefore quality values keep high even for large values of GOP size.



(a) PSNR (b) SSIM

Fig. 8: QoE assessments for videos for different values of GOP and congestion for AMPD

Regarding the number of *B* frames used for encoding, the impact in quality can be seen in Figures 9(a) and 9(b). Because the GOP size is always 10, the *I* frames loss rate is always zero. However, for low values of *B* frames used, the number of *P* frames discarded is high. Because the loss of a *P* frame is more harmful compared to the loss of a *B* frame, the video quality is low for very small values of *B* frames used. Finally, for values higher than 2 *B* frames between each *P* frame, the encoding quality is degraded due to the increase of predictions.



(a) PSNR (b) SSIM

Fig. 9: QoE assessments for video flows for different values of B-frames and congestion for AMPD

Based on the simulation results, it is possible to conclude that for low GOP size values the mechanism has generally worst performances. Small GOP values lead to large number of *I* frames and turn the discovery of less important frames to be discarded less probable. On the other hand, large GOP values lead to higher numbers of dependencies. Therefore, a frame loss can induce a high number of discarded frames on the receiver side if broken dependencies exist. However, AMPD keeps a good quality due to the protection of most important frames.

Regarding the number of *B* frames, simulation results show that small values lead to a reduction in the quality level. A small number of *B* frames leads to a higher number of *P* frames. Since the number of *P* frames increases, the loss rate for these frames also increases.

Concerning the bitrate, simulation results present that different values do not have a visible impact on the proposed mechanisms. The quality variation is only related to the video quality achieved by the encoder when using different compression schemes. Overall simulations results show better quality when GOP size values higher than 10 and a maximum of 2 *B* frames between each *P* frame are used.

6 Conclusion

The usage of multimedia applications, such as voice and video streaming, is increasing in the Internet and will be very popular in next generation networks. However, existing wireless networking systems are not prepared to optimize resources and the quality level of current and new multimedia applications according to the user perception. Throughout this paper, various QoE advanced mechanisms to improve video flows were analyzed, compared, discussed and evaluated. Performance evaluation results show better video experience, without degrading VoIP flows quality, in IEEE 802.11e environments when advanced mechanisms are used. As demonstrated by simulations, AMPD presents the best video quality ratio.

Due to the mechanism simplicities, they can be implemented easily in real networking systems. Information about the frame types can be placed in the Real Time Protocol (RTP) header and acquired by any node through packet inspection. Service providers, such as mobile and multimedia markets, who increasingly provide audio-visual content, can increase their portfolio and revenue, by using QoE-aware packet controller mechanisms, which combine improvement in user perception, video and voice quality level, and wireless resources.

As future works, the proposed mechanisms will be adapted and evaluated in IEEE 802.11s, IEEE 802.16 and Long Term Evolution (LTE) systems. A test-bed environment will also be setup to confirm the simulation results and show the efficiency of the proposed solution in real scenarios.

Acknowledgment

The work is supported by FCT UBIQUIMESH project (PTDC/EEATEL/105472/2008) – Portugal and by National Council for Scientific and Technological Development (CNPq) and FAPESPA - Brazil.

Reference

1. IEEE standard for local and metropolitan area networks part 16: Air interface for fixed and mobile broadband wireless access systems amendment 2: Physical and medium access control layers for combined fixed and mobile operation in licensed bands and corrigendum 1, 2006.
2. IEEE standard for information technology - telecommunications and information exchange between systems - local and metropolitan area networks specific requirements - part 11: Wireless LAN medium access control (MAC) and physical layer (PHY) specifications, 2007
3. N. Moussa, A. Soudani, and R. Tourki. Performances evaluation and enhancement of MPEG4 transmission over IEEE 802.11 WLAN. In *Design and Test of Integrated Systems in Nanoscale Technology, 2006. DTIS 2006. International Conference on*, pages 341–344, Tunis, September 2006.
4. Alexander Eichhorn. Modelling dependency in multimedia streams. In *Proceedings of the 14th annual ACM international conference on Multimedia*, pages 941–950, Santa Barbara, CA, USA, 2006. ACM.

5. Mu Mu, Eduardo Cerqueira, Fernando Boavida, and Andreas Mauthe. Quality of experience management framework for real-time multimedia applications. *Int. J. Internet Protoc. Technol.*, 4(1):54–64, 2009.
6. Eduardo Cerqueira, Lucjan Janowski, Mikolaj Leszczuk, Zdzislaw Papir, and Piotr Romaniak. Video artifacts assessment for live mobile streaming applications. In *FMN '09: Proceedings of the 2nd International Workshop on Future Multimedia Networking*, pages 242–247, Coimbra, Portugal, June 2009.
7. Serral-Gracia, R., Cerqueira, E., Curado, M., Monteiro, E., Yannuzzi M., Masip-Bruin X. "*An Overview of Quality of Experience Measurement Challenges for Video Applications in IP Networks*". In Proceedings of The International Conference on Wired/Wireless Internet Communications (WWIC 2010), Loule, 2010.
8. H.M. Liang, C.H. Ke, C.K. Shieh, W.S. Hwang, and N.K. Chilamkurti. Performance evaluation of 802.11e EDCF in the ad-hoc mode with real audio/video traffic. In Wireless and Optical Communications Networks, 2006 IFIP International Conference on, April 2006.
9. Do Minh Ngoc, Phan Anh Tan, Jong Soo Lee, and Hoon Oh. Improvement of IEEE 802.11 for multimedia traffic in wireless LAN. In Wireless Telecommunications Symposium, 2007. WTS 2007, pages 1–4, Pomona, CA, April 2007.
10. A. Ksentini, M. Naimi, and A. Gueroui. Toward an improvement of h.264 video transmission over IEEE 802.11e through a cross-layer architecture. Communications Magazine, IEEE, 44(1):107–114, 2006.
11. L. Haratcherev, J. Taal, K. Langendoen, R. Lagendijk, and H. Sips. Optimized video streaming over 802.11 by cross-layer signaling. Communications Magazine, IEEE, 44(1):115–121, 2006.
12. Chih-Heng Ke and Naveen Chilamkurti. A new framework for MPEG video delivery over heterogeneous networks. Computer Communications, 31(11):2656–2668, July 2008.
13. Thomas Pliakas, George Kormentzas, and Charalabos Skianis. End-to-End QoS issues of MPEG4-FGS video streaming traffic delivery in an IP/UMTS network. In Autonomic Management of Mobile Multimedia Services, pages 247–255. 2006.
14. Pablo Ameigeiras, Juan J. Ramos-Munoz, Jorge Navarro-Ortiz, Preben Mogensen, and Juan M. Lopez-Soler. Qoe oriented cross-layer design of a resource allocation algorithm in beyond 3g systems. *Computer Communications*, October 2009.
15. Lu LIU, Wen-an ZHOU, and Jun-de SONG. Quantitative and qualitative research on service quality evaluation system in ngn. volume 16, pages 71–77, 2009.
16. Tobias Hoßfeldand Andreas Binzenhöfer. Analysis of Skype VoIP traffic in UMTS: End-to-end QoS and QoE measurements. *Computer Networks*, Vol 55. N 3, pages 650-666, 2008.
17. EvalVid Tool. "A video quality evaluation tool-set". http://www.tkn.tu-berlin.de/research/evalvid/, Febuary 2011.
18. Predicting the perceptual service quality using a trace of VoIP packets. http://www.tkn.tu-berlin.de/research/qofis/, February 2011.