Control of QoE based on Algorithms for the Disposal of Packets concerned with Streaming Video in Wireless Networks

Paulo Bezerra [13] ***Adalberto Melo*** [23] ***Allan Costa*** [23] ***Carlos Quadros*** [13] ***Antônio Abelém*** [23] ***Eduardo Cerqueira*** [123]

Federal University of Pará (PPGEE/UFPA), Po Box 479 - 66075-110 Belém - PA - Brazil [1]
Federal University of Pará (PPGCC/UFPA), Po Box 479 - 66075-110 Belém - PA - Brazil [2]
Research Group on Computer Networks and Multimedia Communications (GERCOM/UFPA) [3]

Summary
Following decades of research, the multimedia networks remain a great challenge; however, the modern multimedia wireless networks with video streaming and Internet Protocol voice, have been attracting special attention due to factors such as mobility and heterogeneity in the devices that are used, which can influence the quality of a user's experience in a particular application or service. This article compares two algorithms for the disposal of packets in wireless networks based on the quality of the experience of the user with video applications when there is multimedia traffic congestion in the wireless networks and where the results obtained surpass those of the wireless networks which lack control of disposal.

Key words:
Algorithms for the disposal of packets; Video Streaming; Wireless Networks; Quality of Experience.

1. Introduction

In the last few years, the wireless networks have, to a great extent, evolved into Next Generation Networks (NGNs) and formed a new structure which enables there to be a convergence of data, voice and multimedia applications on a transport platform that allows applications such as IP voice, known as Voice over Internet Protocol (VoIP) to have mobile access to the Internet and video streaming.
One of the technologies used to make these services available is the Wireless Local Area Network (WLAN), which has components that are currently inexpensive enough to be used in periods of relaxation and can be used to share an Internet connection with the whole family. Developments were made in the transmission patterns of the protocol owners but at the end of the 1990s, these were replaced with patterns belonging to the various IEEE 802.11 Wi-Fi versions (Wireless Fidelity).
Currently, the Wi-Fi networks are the wireless connectivity standard for the local networks. As a proof of its success, one can cite the increasing importance of Hot Spot (the site where the Wi-Fi technology is available) and the growing number of portable computers equipped with wireless interfaces (standard 802.11).
Based on the increasing demand for multimedia services in recent years, such as video streaming, Voice over Internet Protocol (VoIP) and Internet Protocol Television (IPTV), together with the proliferation of mobile devices, the demand by users for this type of service has been increasing owing to the ease of access provided by any device.
Mobile and modern devices such as laptops, Personal Digital Assistants (PDAs), intelligent telephones and Portable Media Players (PMPs) have evolved into powerful portable computers which have access to multimedia material. There is a steady rise in the number of users making use of mobile devices to watch videos that are transmitted through the wireless networks and which require a broader range of material of a higher standard. The forecasts for the telecommunications market show that multimedia applications will be among the main services for the networks in the next generation.
These applications usually require time for a quick response on the network and have a low rate of packet loss along their routes in order to maintain their quality at an acceptable level. However, owing to the nature of the networks based on packets, there are several kinds of wireless network failings that can cause a considerable degradation of the flow, although the effect of these failings on the requests may be trivial if they are not noticed by the end users.
At the same time, owing to the limitations of the traditional Quality of Service (QoS) metrics regarding the evaluation of multimedia applications at the level of the user, new Quality of Experience (QoE) metrics have been introduced.[3].These metrics use a more in-depth analysis of the flows to ensure the evaluation is better; they take into account the specific features of each multimedia application such as the codifiers and decodifiers (CODECS) and the loss and delay tolerance.
This concept was also investigated with the aim of providing new network control functionalities and a QoE

Manuscript received February 5, 2012.
Manuscript revised February 20, 2012.

that ensures the spreading of multimedia material through cable and wireless networks with scant resources.

In one of the works that are described, the writers describe an algorithm to calculate the required bandwidth for Variable Bit Rate (VBR) multimedia services. The calculation is based on the mapping of a QoE descriptor to calculate the anticipated amount of loss [4].

This study will investigate two algorithms that employ an intelligent way of disposing of video streaming packets in wireless networks that take account of QoE where two metrics will be used to measure the degree of quality perceived by the user: the Peak Signal to Noise Ratio (PSNR) and the Structural Similarity Index (SSIM). Apart from this introductory section, the article comprises five other sections. In Section 2, there is an account of work related to the subject of this research. In Section 3, the QoE metrics are described and Section 4 outlines the algorithms for the planned disposal. Following this, in Section 5 a case study is examined in which simulations were carried out to verify the performance of each disposal algorithm and which took account of a group of images with 10,15 and 20 GOP (Group of Pictures) frameworks. Finally, in Section 6, there is a discussion of ideas and recommendations for further studies.

2. Related Work

Research studies have shown that the transmission of multimedia material through the Internet in real-time is becoming increasingly popular and taking up more resources of the world network of computers. At the same time, more attention is being paid to the quality observed by the end user, and an increasing amount of importance is being attached to concerns about the influence that multimedia applications in real-time can exert on other kinds of flow that travel through the Internet. However, the use of control algorithms for multimedia traffic envisages the guarantee of QoS and QoE for both fixed and mobile users in the wireless networks.

In one of the studies, the authors [04] provide an algorithm to calculate the required widthband for Multimedia Variable Bit Services (VBR). The calculation is based on the mapping of a QoE descriptor that assesses the quality perceived by the users and a QoS descriptor to calculate the anticipated amount of loss. Three metrics were used to measure the quality perceived by the user: the Peak Signal to Noise Ratio (PSNR), Structural Similarity Index (SSIM) and the Video Quality Model (VQM).

The model for the planned system estimates the width of the band that is needed for the degree of quality required for the flow. The reservation of the band is divided into two stages. In the first stage, the method determines the amount of loss for the end user. In the second stage, the width of the band required is calculated on the basis of the maximum loss permitted. This second procedure is designed by means of a form of alignment – Discrete Time Markov Chain – which determines the process of arrival to the video file and is based on the H.264/SVC codec. The video model outlined by the authors has not been devised to provide an exact description of the original video, but rather to allow an efficient reservation for the band.

In another study, the writers outline the techniques of Automatic Network Management (ANM) that automize the reservation of resources in traffic engineering through the metrics of user satisfaction. Even though assessing QoE requires a set of metrics that is able to evaluate the satisfaction of the end user in an objective way, in practice the evaluation becomes a complex process because it involves a series of subjective factors. These factors are not usually related to the performance of the network but may be, for example, to do with the humour of the user or the responses of the system, as opposed to the classic assessment platforms of QoS which are based on the network. The current limitations of the systems employed for the mediation of QoS with regard to subjective factors regarding human perception, are analysed by the authors with the aim of understanding the new challenges that arise in the evaluation of QoE in the delivery of multimedia systems. This analysis is carried out to allow a more precise assessment of the degree of quality experienced by the users. The authors evaluate multimedia traffic through the network by employing QoE mechanisms which seek to define which metrics are suited to an evaluation of quality. These metrics were divided into two distinct categories, which are called Direct Metrics when they are obtained form various types of data in different layers and requiring specific information about the performance of the network. The Indirect Metrics take account of the properties that affect the multimedia experience but are not directly related to the quality of the multimedia material.

In another study, the authors [06][07] discuss tests for evaluating the impact of different time periods on the loss of packets. In practice, the loss can occur because of a temporary loss of connectivity following link failures or among those of us in the network, where the best and the worst scenarios regarding the MPEG transmission flow can be identified. The tests that were conducted show how each scenario can be used to verify the kind of artifacts displayed and the impact of QoE on the spectator. These tests were carried out in consecutive periods when there was a loss of IP packets, which vary from 10ms to 500ms for the movement of the high-resolution video and for the Standard-definition (SD) and high-definition (HD) resolutions. It was found that in the tests carried out, the same conclusions that were reached for SD, apply to HD and that there were no significant differences between SD and HD during a period equivalent to that of the packet losses.

3. QoE Metrics

At present, there is an increase in the number of studies on the Quality of Experience (QoE) although not many are centred on subjective and objective technical indicators.
The relationship between the technical quality parameters and subjective indicators can be regarded as a concept that comprises both the objective side (such as, for example, the parameters related to the network) and the subjective (for example, contextual parameters concerning the user). Most of the definitions and empirical studies with regard to QoE tend to stick closely to the logic of technology and disregard the subjective experience of the user[8]. As a result, they do not interact with concepts from other areas, such as for example, Human Computer Interaction, in which the “User Experience” and “Usability” are bound up with QoE and are of great importance.

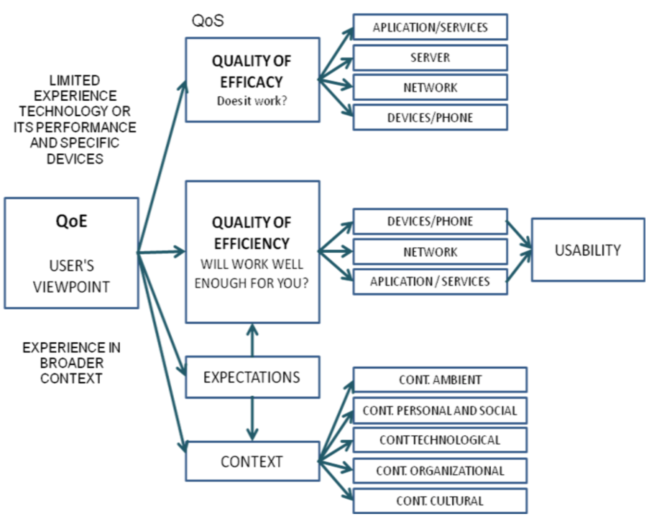

Figure 1: Diagram of the aspects and metrics of QoE[9]

The existing metrics of QoS, such as the rate of packet loss and delay, are usually used to indicate the impact of the performance of the network on the delivery of requests. However, the conventional QoS metrics only supply information about the state of the network (the level of the packet). Metrics such as the perception and awareness of the user of video traffic must be taken into account to increase the control over the information in the network. New metrics, known as QoE metrics, have appeared. The QoE metrics enable the control and evaluation systems to know how the user is selling the service and are divided into subjective and objective metrics [10][11]. In the case of subjective methods, a group of assessors who are in particular visualizing conditions, evaluate the “quality” material of video streaming. The results are handled in a statistical way to obtain a full picture of the perception of the user [12]. However, these measures are relatively slow and expensive, and cannot be used for evaluations in real time. The objective metrics are based on mathematical models which seek to approximate to the results of the subjective metrics. Owing to overloading caused by the objective metrics in the system, some methods cannot be used in real scenarios and are mainly used for purposes of simulation with the PSNR and SSIM metrics.
The PSNR [13] is a complete set of references for metrics used in the evaluation of the video. To calculate the final quality, this metric makes use of a mean value for the difference between the values of luminosity of each pixel of the original processed moulding. Apart from the fact that it is fully utilized, owing to its low level of complexity, the PSNR metric only provides an indication of the difference between the received frame and a reference signal during the evaluation of factors such as the Human Visual System (HVS). To make a comparison that takes account of the structure of the objects and provides a better evaluation, the SSIM metric breaks down the images that have been sent and received while taking note of the three HVS components – luminosity, contrast and structural distortions [14].

4. The Projected Algorithms

Currently, the standard 802.11 wireless networks offer little QoE support for video streaming applications. In the videos that are codified in the MPEG format, the frames can be divided into 3 types. These are Type 1 frames. Type P frames and Type B frames and each kind of frame has a role of a different importance that can directly influence both the ultimate quality of the video and the evaluation of the user [15]. If this feature is observed, it becomes possible to improve the regulatory elements of the traffic by means of a configuration, without overburdening the network resources to the extent of exerting any kind of influence on the flows in operation.
When the differences between the video flows and the CODEC parameters are taken into account, as well as the interdependence of the frames and the other requirements of QoS and QoE, it becomes more apparent why there is a need for algorithms that are intelligent and different from those currently being employed to improve the quality of the transmissions of video streaming and efficiency in the use of the network resources. In this scenario, it is necessary to improve support for QoE with regard to video streaming in the wireless networks, since this technology that provides access to the network can be regarded as one of the most important among the new network technologies. These algorithms should take account of the influence of the video streaming traffic on the perception of the end user in congested networks by improving the traffic regulators in accordance with the current conditions of the network, QoE, dependencies and the importance of the frames. The algorithms must be flexible with regard to

the need to make a transition from a mobile unit (UM), from one cell to another, in a way that is sufficiently transparent for the user to allow the continuity of the services and applications being undertaken, and also with regard to the need for alterations in the network devices and terminals of the users.

Two algorithms that seek to meet the requirements referred to earlier, are discussed here: they envisage an improvement in the quality of video experience from the perspective of the user and are algorithms focused on disposal and based on the control of inter-flow packets. Although they may be beyond the scope of this work, the algorithms can be configured so that they can be used in networks with different QoS models, as well as in wireless networks and mobile technologies, such as the 802.11 and 802.16 standards for the wireless network. With regard to the question of interoperability, since the framework of packetsand routers are responsible for providing the procedures required for the control of admission and congestion, they do not need to undergo extensive alterations to support the algorithms that are recommended and only minor modifications need to be carried out.

4.1 Algorithms Prioritizing Disposal (ADP)

In accordance with the MPEG codification framework, some frames are more important than others due to the fact that the number of frames depends on others that are being reconstructed. As in one GOP, all the frames are dependent on an I Frame, the I Frames are regarded as more important that the other frames. In the case of the P Frames, although their frames depend on the I Frames, they also possess frames that depend on the I Frames that will be reconstructed. Thus, these are also important although they are less important than the I Frames .

Finally, since the B Frames do not possess frames dependent on it, they become frames of a type that is less important and which have less impact on the experience of the user. The Algorithm Prioritizing Disposal (ADP) accounts for the degree of importance of each type of frame from the codification framework (MPEG) and includes a system for the disposal of packets that attains a control of the level of video streaming based on the degree of importance of the type of frame. When an I Frame is marked to be rejected, a verification procedure is carried out to check if there is a packet in the waiting file that contains a Type P or B. If one exists, it is better to carry out the disposal of a packet containing an I Frame. This is because the number of dependent frames is greater than the number of frames dependent on a P or B Frame. It also makes it more important since it reduces the loss of these types of frames and as a result, has a smaller impact on the final quality of the video with regard to user perception. Thus, an exchange occurs of the rejected packet containing a Type I frame for a packet containing Type P or B. Owing to its degree of importance, the same process takes place when a P Frame is marked to be rejected, since this time a process of verification is carried out to check if there is a packet in the waiting file containing a Type B frame. If this is the case, the B Frame is rejected instead of the P Frame. This is also due to the degree of importance and the packet containing the P Frame, which will be rejected, is entered in the waiting file, and this also reduces the impact on the final quality of the video, as is seen in Figure 2.

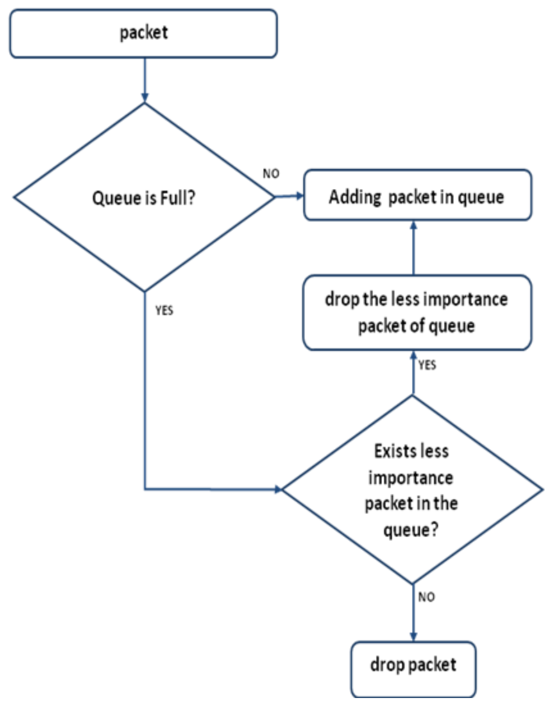

Figure 2: Flowchart of the ADP functions

Table 1 below, shows the logical pattern of the algorithm in a simplified form to highlight its functions.

TABELA 1. DISCARD ALGORITHM FOR PRIORITY (ADP)

```
if queue.isNotFull():
  queue.add(packet);
else:
  if packet.containsFrameType('B'):
    drop(packet);
  else:
    packetToDrop :=
queue.getFrameType('B');
    if packetToDrop <> null:
      queue.drop(packetToDrop);
      queue.add(packet);
    else
      if packet.containsFrameType('P'):
```

```
            drop(packet);
        else:
            packetToDrop :=
queue.getFrameType('P');
            if packetToDrop <> null:
                queue.drop(packetToDrop);
                queue.add(packet);
            else:
                drop(packet);
```

4.2 Algorithm for Breaking Disposal (ADQ)

The purpose of this rejected algorithm is to improve the management of the network resources and thus make it possible for the quality of the video streaming applications to be stabilized to a satisfactory level. The ADQ is based on the interdependence between the frames and the perception of the user. This algorithm checks when the file is full and allows the packet that is to be rejected to be retrieved by checking whether there are packets containing frames in the file with a broken dependence, or rather, ensuring that if the frames that will be reconstructed, whether they have failings or not, can be reconstructed at their terminal point, in accordance with what can be seen in Figure 3.

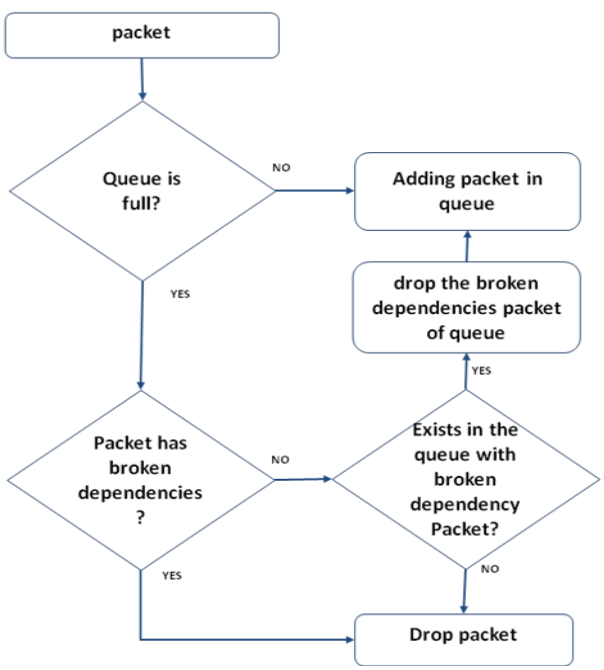

Figure 3: Flowchart of the Functions of ADQ

As shown below in Table 2, the logical pattern of the ADQ algorithm, in the case where a packet is marked to be rejected, is first checked to establish if it contains a frame with a broken dependence in the file. If a packet is rejected, but does not have a broken dependence in the file, it is checked to determine if there are packets with a frame with a broken dependence in the file. If this is the case, the packet in the file is rejected and the entry packet is included in the file.

TABELA 2. ALGORITHM FOR DISPOSAL DUE TO BREAKAGE (ADQ)

```
if queue.isNotFull():
  queue.add(packet);
else:
  if
queue.getFrameBrokenDependencies(packe
t):
    drop(packet);
  else:
    packetToDrop :=
queue.getFrameBrokenDependencies();
    if packetToDrop <> null:
      queue.drop(packetToDrop);
      queue.add(packet);
    else:
      drop(packet);
```

5. Results From The Adopted Algorithms

It can be observed that in our previous items, the differences have already been established between the objective and subjective QoE metric objectives and those that concern the algorithms with rejected packets in the wireless video streaming networks. It should be stressed that the factors being considered are restricted to the performance of the rejected packets and not the QoE metrics. In the light of this, this article compares the algorithms affecting the disposal of ADP and ADQ with a wireless network without any control of rejection.

5.1 Methodology

The purpose of the experiment was to simulate the performance of the algorithms rejected from the packets while including important factors in the wireless networks such as the PSNR evaluation, SSIM, the sizes of GOP and the rates of congestion in the network. To achieve this, a series of measures were undertaken by means of a simulation tool for networks called the NS-2 Network Simulator with modules for the 802.11 standard for the wireless network and multimedia traffic with the aid of the evaluation tool of the Evalvid video [17].

5.2 The Simulation Scenario

The scenario is formed of a fixed node source and a mobile receptor which is displaced from a home network to a foreign network (RE) and was used to compare the algorithms outlined in this study by analysing the performance for different values of the size of the GOP. Video flows and VoIP flows were used for all the scenarios with a better strength and density of traffic, since each flow was mapped out in a category with a different 802.11 access standard. The flows were divided into access categories as follows: VoIP for AC_VO, the video flow for AC_VI, the better strength traffic for AC_BE and the dense traffic for AC_BK. The foreman video from the Common Intermediate Format Sequences (CIF), was used with 30 frames per second (fps), between each P frame, two B frames and a bit rate of 320 kbps and different sizes of GOP.

5.3 Results

The following section shows the results of the simulations which compare the algorithms for the disposal of the packets. The algorithms were submitted to a scenario with different GOP sizes (10,15 and 20) and with different rates of congestion in the network (from 50% to 150%). The results found with regard to the video flows are shown in Figures 4 – 6 for the PSNR metrics and Figures 7-9 for the SSIM metrics for each rejected algorithm in this study and for situations when the rejection algorithms were not made use of. Figure 4 shows when the GOP possessed a frame with 10 frames; with regard to the PSNR metric, it was observed that they all equalled up to 85% of the congestion. In a range of 85-115% and of 125-135% of congestion, the ADP obtained a better evaluation. The ADQ obtained a better evaluation in a range of 115-125% of congestion.

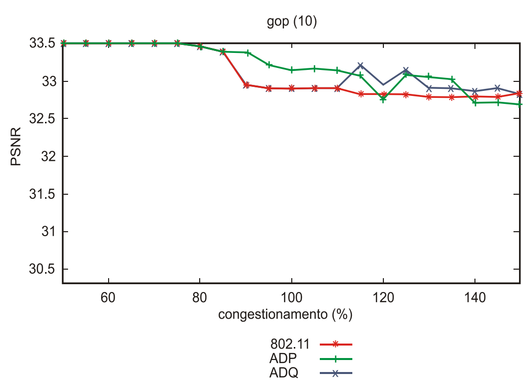

Figure 4: The PSNR Metric with GOP 10

Figure 5 shows that when the GOP has a size of 15 frames with regard to the PSNR metric, it was observed that practically all of the frames have the same evaluation – up to 100% of congestion.
In a range of 100-135% and approximately 148-150% of congestion, the ADQ has a better evaluation. The ADP obtained the best evaluation in a range of 135-145% of congestion.

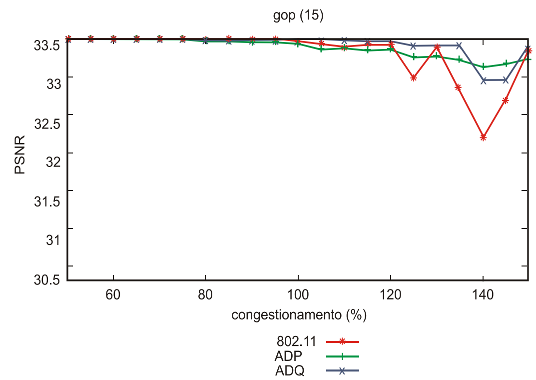

Figure 5: the PSNR metric with GOP 15

Figure 6 shows when the GOP has a size of 20 frames with regard to the PSNR metric; it was observed that all of the frames have the same evaluation of up to 105% of congestion. When the congestion is at 125%, in the range of 125-140% or when it is at 150%, the ADQ has a better evaluation. The ADP has a better evaluation when the network is at 110%, 120% and 145% of congestion.

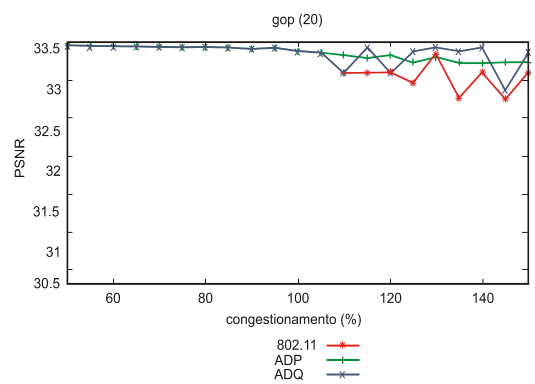

Figure 6: PSNR Metric with GOP 20

Figure 7 shows when the GOP has a size of 10 frames based on the SSIM metric; it was observed that all of the frames had the same evaluation of 50-85% of congestion and were in the range of 130-150% of congestion. When the congestion is in the range of 90-110% and when it is at 125%, the ADP has a better evaluation. The ADQ has a better evaluation when the network is at 115% of congestion.

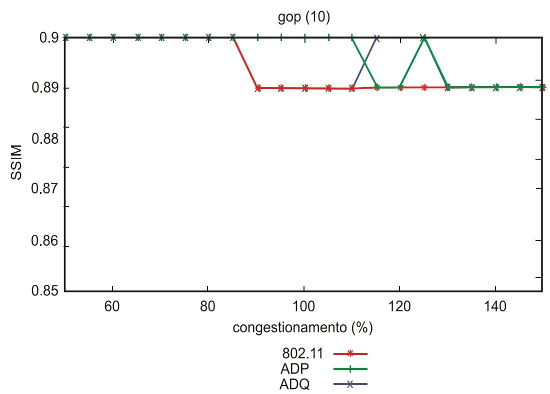

Figure 7: The SSIM Metric with GOP 10

Figure 8 shows when the GOP has a size of 15 frames based on the SSIM metric; it was observed that all of the frames had the same evaluation with almost all the rates of congestion. When the congestion is at 145% of congestion its evaluation is the same as that of standard 802.11.

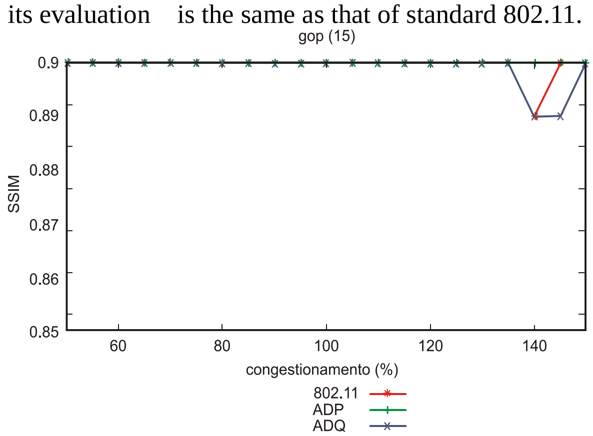

Figure 8: SSIM Metric woith GOP 15

Figure 9 shows when the GOP has a size of 20 frames based on the SSIM Metric; it was observed that all of the frames had the same evaluation with almost all the rates of congestion. The ADQ and ADP have a better evaluation when the network is between 135-145% of congestion.

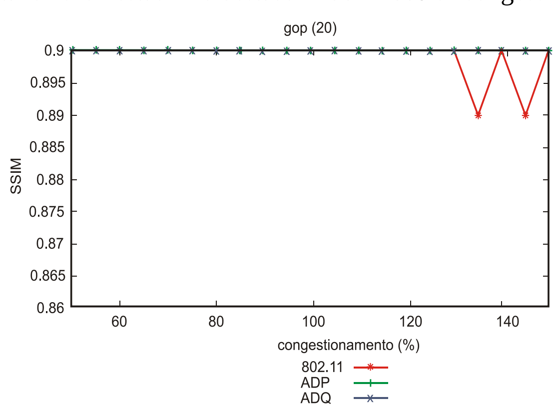

Figure 9: SSIM Metric with GOP 20

The ADP attained more stable PSNR values for all the congestion values, regardless of the size of the GOP, and better SSIM values at moderate levels of congestion when the GOP had a size of 10 and advanced levels of congestion when the GOP had a size of 15. The reason for this is that in the other levels of congestion, the ADP was equal to those of the other algorithms, where it should be noted that no packet was rejected that contained the I Frame. The ADQ showed values for the SSIM evaluation below that of ADP at some periods of the congestion and moderate PSNR values of quality although it had shown higher values for the lost frames, including the Type 1 frames, due to the fact that it had an algorithm for the disposal of packets through a breakage of dependencies.

6. Conclusion and Recommendations for further studies

The demand for new aggregated multimedia applications for mobile devices has acted as a driving-force for research to seek new wireless network standards. At the same time, it remains a challenge to improve the efficiency of the multimedia applications for both fixed and mobile users.

In this article it has been possible to analyse a hypothetical scenario through the assessed results emerging from a simulation that involves the implementation of two algorithms for the disposal of packets in an intelligent way. The ADP and ADQ, which only affect video flows, do not affect the quality of other multimedia flows and maintain a quality equal to that of standard 802.11.No rejection of the packets of the VoIP flows has occurred because it is a flow of high priority and the network possesses sufficient resources. The standard 802.11 only shows a small reduction in the level of the quality of VoIP due to the increase of delay and jitter with the congested network.

With regard to the standard 802.11 wireless network, without the use of algorithms for the disposal of packets, the rate of loss of packets is low. Moreover, it has high values of traffic congestion caused by dense traffic controls which had a powerful effect. The network envisages that it can guarantee support for video flows of quality and shows a good video quality for all the values of congestion.

However, the regulator of the service does not distinguish between the I,P and B frames and has random values of loss for each type of frame. In this context, there could be an improvement in the quality of video if, to achieve this, the importance of each type of frame was taken into account. Future studies could be undertaken into the question of the union of two algorithms (ADP and ADQ) while it is noted that the ADQ does not take account of frames of less importance without a broken dependence on the rejected packets.

References

[01] Seyedzadegan, M. e Othman, M. WLANs Fairness Controlling: Distributed TCP Fair Bandwidth Allocation. Ed. 1º.LAP LAMBERT Academic Publishing 2010.

[02] Holt, A. e Chi-Yu Huang. 802.11 Wireless Networks: Security and Analysis (Computer Communications and Networks). Ed. 1º. Springer. 2010.

[03] K. Kilkki, “Quality of experience in communications ecosystem,”Journal of Universal Computer Science, vol. 14, no. 5, pp.615–624, 2008.

[04] Kovács, A. Gódor, I. Rácz, S end Borsos, T. Cross-layer quality-based resource reservation for scalable multimedia. Computer Communications, Volume 33, Issue 3, 26 February 2010, Pages 283-292.

[05] Serral-Gracia, R., Cerqueira, E., Curado, M.;Monteiro, E.Yannuzzi, M., X. Masip-Bruin. An Overview of Quality of Experience Measurement Challenges for Video Applications in IP Networks. In: International Conference on Wired/Wireless Internet Communications (WWIC 2010), 2010, Lule. Proceedings of The International Conference on Wired/Wireless Internet Communications (WWIC 2010), 2010.

[06] Greengrass, J., Evans, J. e Begen, A, C. "Not all packets are equal, part I: streaming video coding and SLA requirements," IEEE Internet Comput., vol. 13/1, pp. 70-75, Jan./Feb. 2009

[07] Greengrass, J., Evans, J. e Begen, A, C. "Not all packets are equal, part II: the impact of network packet loss on video quality," IEEE Internet Comput., vol. 13/2, pp. 74-82, Mar./Apr. 2009

[08] K. De Moor, W. Joseph, I. Ketyko, E. Tanghe, T. Deryckere, L. Martens and L. De Marez, “Linking Users’ Subjective QoE Evaluation to Signal Strength in an IEEE 802.11b/gWireless LAN Environment,” EURASIP Journal on Wireless Communications and Networking. pp. 01–12, 2010.

[09] K. De Moor and L. De Marez, “The challenge of user- and QoE-centric research and product development in today’s ICT-environment,” in Innovating for and by Users, J. Pierson, E.Mante-Meijer, E. Loos, et al.,Eds., pp. 77–90,Office forOfficial Publications of the European Communities, Luxembourg, UK, 2008.

[10] A.E. Mahdi. Voice quality measurement in modern telecommunication networks. In Systems, Signals and Image Processing, 2007 and 6th EURASIP Conference focused on Speech and Image Processing, Multimedia Communications and Services. 14th International Workshop on, pages 25–32, Maribor, June 2007.

[11] Marie Guéguin, Régine Le Bouquin-Jeannès, Valérie Gautier-Turbin, Gérard Faucon, and Vincent Barriac. On the evaluation of the conversational speech quality in telecommunications. EURASIP J. Adv. Signal Process, 8(2):1–15, 2008.

[12] ITU-T j.140 : Subjective picture quality assessment for digital cable television systems, March 1998.

[13] K. Nishikawa, K. Munadi, and H. Kiya. No - Reference PSNR estimation for quality monitoring of motion JPEG2000 video over lossy packet networks. Multimedia, IEEE Transactions on, 10(4):637–645, June 2008.

[14] David M. Rouse and Sheila S. Hemami. Understanding and simplifying the structural similarity metric. In Image Processing, 2008. ICIP 2008. 15th IEEE International Conference on, pages 1188–1191, San Diego, CA, USA, October 2008.

[15] Gomide. J, V, B. and Araújo. A. A. Visual effects, an approach from digital image processing, Magazine of theoretical and Applied Informatics, RITA. Vol: XVI. Nº 1. 2009.

[16] The network simulator - ns-2. http://www.isi.edu/nsnam/ns/, June 2010.

[17] EvalVid - a video quality evaluation tool-set. http://www.tkn.tu-berlin.de/research/evalvid/, June 2010.

Paulo Henrique Gonçalves Bezerra. Electrical Engineer PhD. Student from Federal University of Para (UFPA). Master in Electrical Engineering from Federal University of Para (UFPA).

Adalberto Cavalcante de Melo. Student Master Program in Computer Science (UFPA). Has degree in Systems Analysis and Development with a specialization in Computer Networks and Computer Science.

Allan Douglas Bento da Costa. Electrical Engineer PhD. Student from Federal University of Para (UFPA). Master in Computer Science from Federal University of Para (UFPA).

Carlos Jean Ferreira de Quadros. Student Master in progress in Electrical Engineering with emphasis on Applied Computing at the Program in Electrical Engineering (UFPA). Graduated in Computer Engineering from the Federal University of Para (UFPA).

Antonio Jorge Gomes Abelém. Received Ph.D. degree from Catholic University of Rio de Janeiro (2003). Professor and researcher at the Federal University of Para (UFPA).Works in Computer Science, with emphasis on Computer Networks, acting on the following topics: the future of the Internet, optical networks, wireless ad-hoc multicast communication, TCP / IP architecture, Quality of Service (QoS) quality of experience (QoE) and security.

Eduardo Coelho Cerqueira. Received Ph.D. degree from Computer Engineering from the University of Coimbra (2008). postdoctoral fellow in Computer Engineering from the University of Coimbra (2009). Professor and researcher at the Federal University of Para (UFPA). Interests are Mobility, Multimedia Multicast, next generation networks, sensor networks, resource reservation, Quality of Service and Quality of Experience.