Improving IPTV QoE taking the suitable MPEG-2/MPEG-4 Quantizer based on Jitter, Delay and lost packets measurements

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***Abstract* — The main issue in Digital Terrestrial Television and in IPTV networks is the Quality of Experience received by the end users. For this reason, mechanisms to automatically measure the video quality of the images received by the user are needed. In this paper, we analyze video quantization in order to determine an optimal quantizer_scale factor value for its transmission. Then, it is used as an automatic measure to improve the video quality received by the end user. The paper shows the measurements taken for Objective and subjective Video Quality, Video Quality Metric and the bandwidth consumed for several types of video quality. We use the jitter, delay and lost packets measurements in order to take the appropriate quantizer. Finally, we show the visual comparison between a high quantizer_scale factor and a reference video. Our work shows that an optimal quantizer_scale factor can be used to save bandwidth in an IPTV network or to improve the Video Quality for the same bandwidth consumption. Finally, we present some discussions of the measurements gathered and some comments of other authors.**

Keywords – MPEG-2/MPEG-4 Quantizer, Video Quality, IPTV.

I. INTRODUCTION

MPEG-2 and MPEG-4 encoding are standards that are widely used by the digital television industry [1]. Although, nowadays, MPEG-2 is the most used, MPEG-4 is gaining ground because it provides good quality image with lower bandwidth consumption [2]. The application fields where MPEG-4 can be applied are Digital Television, interactive graphical applications and interactive multimedia, while providing high audiovisual data compression to store or stream video and, at the same time, audio and video quality. MPEG-4 reduces the data rate in half, with the same image quality than MPEG-2. This will increase the offer and the plurality of channels and, at the same time, the scalability of network services. MPEG-4 compression is based on visual-objects coding [3] and uses further coding tools, like System Decoder Model, Sync Layer, Flexible Multiplex, etc. [4], to achieve higher compression factors than MPEG-2, thus it needs less bandwidth for its transmission, but MPEG-2 is the most used codec in Digital Terrestrial Television and in IPTV networks, because it has lower complexity and hardware requirements at the end user.

MPEG-2 compression format is quite used for video storage in hardware devices (DVD, SVCD, etc) and to transmit real time video in several Digital Video Broadcasting (DVB) standards [5]:

- DVB-T: This system transmits compressed digital audio, video and other data in a MPEG-2 transport stream, using COFDM modulation.
- DVB-S: This system increases the data transmission capacity and digital television via a satellite UH11 using the MPEG-2 format, and QPSK modulation.
- DVB-C: This system transmits MPEG-2 or MPEG-4 family digital audio/video stream, using several QAM modulations with channel coding.

DVB system has been used as a basis to standardize the Internet Protocol Television (IPTV). It includes MPEG-2 DVB services [6] encoded with MPEG-2 technology and encapsulated in MPEG-2 Transport Stream (MPEG-2 TS) [7]. But, MPEG-4 can be also added in this transport stream. Moreover, it covers Live Media Broadcast services (i.e TV or radio styles), Media Broadcast with Trick Modes and Content on Demand services (CoD). The goal of DVB-IP is to specify technologies on the interface between an IP based network and a DVB-IP Set-top-Box, which uses a protocol stack for DVB IP services. A diagram of the protocol stack is given in Figure 1.

Once DVB services are encoded, the video content is packaged and encapsulated. This involves inserting and organizing video data into individual packets. The encapsulation of the video content is done using MPEG-2 TS, where all MPEG-2 TS will be encapsulated in Real time Transport Protocol (RTP) according to RFC 1889 [8] in conjunction with RFC 2250 [9], and RFC 768 [10] as the transport layer protocol.

Initially, MPEG-4 doesn't define a transport layer [11]. There are only two adjustments on the MPEG-2 TS to transport MPEG-4 streams. The first one is defined in RFC 3016, which is based in RTP packets [12]. The second one is DMIF, or Delivery Multimedia Integration Framework, which is an interface between the application and the transport. It allows the MPEG-4 application developer to stop worrying about MPEG-4 transport. A single application can run on different transport layers.

The message fields used in the transport stream of the MPEG-2 based DVB content over IP are the following ones: a standard IP header, an UDP header, a RTP header and an integer number of 188 bytes MPEG-2 TS packets, see Figure 2. The maximum size of IP datagram (65535 octets for IPv4) is limited. In the case of an Ethernet-based network, with a Maximum Transfer Unit (MTU) of 1492 or 1500 bytes, the number of MPEG-2 TS packets is 7.

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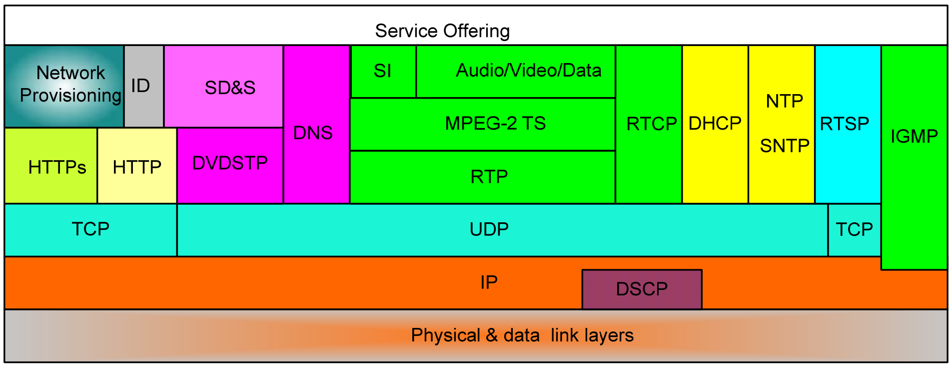

Figure 1. Protocol stack for DVB-IP services

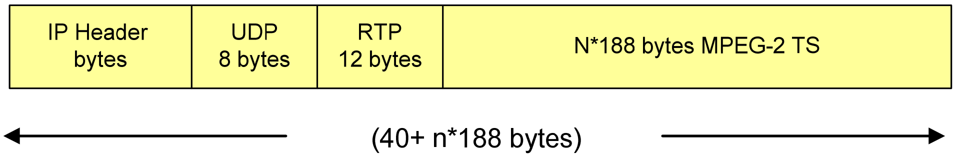

Figure 2. Message to transport MPEG-2 based DVB content

The main challenge in current IPTV systems is to provide high Quality of Experience (QoE) to the user, but using noninvasive methods while the network is being monitored.

In the paper with reference [13] we analyze MPEG-2/MPEG-4 quantization in MPEG-2 TS packets, as a noninvasive method to evaluate the video quality function. It is useful because it allows us to measure the QoE perceived by the IPTV customer. Several quantizer scales are used to evaluate, which of them are better according to the bandwidth, objective video quality, etc. In this paper, we have extended [13] by adding many more tests and adding more variables for our comparison. Moreover, we now have another conclusion: QS=4 is better taking into account the Bandwidth, Jitter, Delay, Lost Packets, VQM, MOS and subjective analysis, while in [13] the best one was QS=2.

The remainder of this paper is organized as follows. Section II explains some work related with video MPEG-2 and MPEG-4 encoding and user video-quality perception. Section III explains the general concept of quantization. The system architecture used to perform our test is presented in Section IV. Section V shows the measurements obtained when different quantization scales are used in order to find the optimal quantizer_scale factor. Objective and subjective video quality for IPTV is shown in Section VI. Section VII shows the jitter, delay and lost packets test. Section VIII provides some discussions of the measurements gathered and the ones taken from other authors. Finally, Section IX draws the conclusion and indicates further research.

II. RELATED WORK

There are several works published where the authors improve the efficiency of the MPEG-2 and MPEG-4 algorithms by modifying some of their parameters.

In [14], Zhenzhong Chen and King Ngi Ngan review the recent advances in rate control techniques for video coding. The video quantization is used to reduce the bit rate of the compressed video signal. It lets meet the size or bandwidth limitation properly. The rate control algorithms recommended for the video coding standards are briefly described and analyzed. Moreover, the recent advances, such as new concepts in rate-distortion modeling and quality constrained control, are presented. With these techniques, the rate control performance can be improved.

An example is given in [15], where O. Verscheure et al. analyze how the user-perceived quality is related to the average encoding bitrate for a variable bitrate (VBR) MPEG-2 video. They show why simple distortion metrics may lead to inconsistent interpretations. Furthermore, for a given coder setup, they analyze the effect of packet loss on the user-level quality. Finally, they demonstrate that, when jointly studying the impact of coding bit rate and packet loss, the reachable quality is upperbound and exhibits one optimal coding rate for a given packet loss ratio.

The authors in [16] describe a complete practical two-pass MPEG-2 encoding system that can be tuned to produce a variable bit rate (VBR) stream in a second pass. In a first pass, the video sequence is encoded with constant bit rate (CBR), while statistics concerning coding complexity are gathered. Next, the first-pass data is processed to prepare the control parameters for the second pass, which performs the actual VBR compression. They conclude their paper saying that the second-pass VBR sequences visually appear to have a higher overall quality than the ones coded with CBR. For VBR to visually outperform CBR, a mix of “easy” scenes and “difficult” scenes is always required.

Sung-Hoon Hong et al. propose a rate control scheme using a rate-distortion (R-D) estimation model, which produces a consistent picture quality between consecutive frames, in [17]. Their rate control scheme ensures that the video buffers do not underflow and overflow by satisfying the buffer constraint. Their simulation results show that their control scheme achieves 0.52-1.84 dB peak signal-to-noise

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ratio (PSNR) gain over MPEG-2 Test Model 5 (TM5) rate control and maintains very consistent quality within a frame as well as between frames.

Another paper where the authors try to improve the efficiency of the encoders is shown in [18]. The authors optimize the operational control of MPEG-2, H.263, MPEG-4, and H.264/AVC encoders respect to their rate-distortion efficiency using Lagrangian optimization techniques. The performance of the H.264/AVC compliant encoder in all experiments clearly demonstrates the potential importance of this standard in future applications of video streaming as well as interactive video coding.

In [19] Zhihai He and Sanjit K. Mitra present an adaptive estimation scheme to estimate linear relationship between the coding bit rate and the percentage of zeros among the quantized transform coefficients. Based on the linear source model and the adaptive estimation scheme, a unified rate control algorithm is proposed for various standard video coding systems (MPEG-2, H.263, and MPEG-4). This algorithm is outperformed with other algorithms providing more accurate and robust rate control with very low computational complexity and implementation cost.

If a system makes a change in the quantizer, this may be more efficient in the transcoding process. In [20], the authors present a rate control scheme for MPEG-2 to H.264 transcoder. They construct an analytic model to set a reasonable initial quantization parameter (QP) for the first frame at the beginning of transcoding. The QP for each frame and each macroblock are adjusted by QPs extracted from the incoming MPEG-2 pictures to avoid consuming bits without video quality gain. They demonstrate by the experiment results that their algorithm improves overall quality for transcoded video while retaining smooth quality.

Finally, an efficient conversion scheme in order to apply the already existent DCT-domain transcoding schemes to MPEG-2/H.264 transcoding is proposed in [21]. Their scheme consists of two conversion steps: the quantization conversion and the DCT conversion. The quantization conversion changes the MPEG-2 quantization step size (Qstep) to the new H.264/AVC Qstep. Additionally, it can improve PSNR performance by reducing the reconstruction errors caused in the pixel-domain transcoder. Their experimental results show that the proposed scheme reduces computational complexity by 5-11% and improves video quality by 0.1- 0.5 dB compared with other solutions.

In this work we analyze the quantizer scale factor (QS) based on the measurements taken of the jitter, delay and lost packets. This study allows us to improve the quality of experience (QoE) in IPTV networks.

III. QUANTIZATION

Quantization is basically a process for reducing the precision of the Discrete Cosine Transform (DCT) coefficients in an encoder. This is very important, since lower precision implies a lower bit rate in the compressed data stream. The quantization process involves the division of the integer DCT coefficients by integer quantization values. The result is an integer and fraction, and the fractional part must be rounded according to the rules defined by MPEG [22]. The result is the quantized value that is transmitted to the decoder.

For reconstruction, the decoder must first dequantize the quantized DCT coefficients in order to reproduce the DCT coefficients computed by the encoder. Essentially, the Inverse Quantization (IQ) scales every element by a unique quantized weight. Since some precision was lost quantizing, the reconstructed DCT coefficients are necessarily approximations to the values before quantization.

After entropy decoding, the two-dimensional array of coefficients, QF[v][u], is inverse quantized to produce the reconstructed DCT coefficients, F[v][u]. In MPEG, IQ consists of three stages: Inverse Quantization Arithmetic, Saturation, and Mismatch Control. The inverse quantization arithmetic produces F’’[v][u] coefficients [22]. For DCT coefficients matrix expression 1 is used:

$$F''(v,u) = \sum_{v=0}^{7} \sum_{u=0}^{7} \left[\left(2^n \times Q_F(v,u)\right) \times k\right] \times a + \left[\left(2 \times Q_F(v,u) + \left(\frac{d|Q_F(v,u)|}{d(u,v)} \times (1-k)\right)\right) \times W(w,v,u) \times Q_S/32\right] \times (1-a) \quad (1)$$

Where, 2^n represents the multiplier intra DC, the intra_dc_mult factor, for $n \in \{0, 1, 2, 3\}$. It is derived from the data element intra_dc_ precision (in case of MPEG-2, it is estimated according to Table 7-4 of the ITU-T Recommendation H.262 [22], in case of MPEG-4, it is estimated according to [3]). The a variable depends of the intrablocks (see expression 2), when the three conditions are complied it doesn't get a null value, otherwise, only operates with the second one. The second part of expression 1 depends of the value of k (seen in expression 3), the luminance and chrominance.

$$a = \begin{cases} 1 & v = 0, \quad u = 0, \quad k = 1 \\ 0 & others \end{cases} \quad (2)$$

$$k = \begin{cases} 1 & \text{intrablocks} \\ 0 & \text{non-intrablocks} \end{cases} \quad (3)$$

The quantizer_scale factor (QS) is an integer and is encoded with a 5-bit fixed-length code. Thus, it has values in the range {1,....,31}. 0 value is not allowed. Each weighting coefficient, W[w][v][u]; w = 0 ... 3; v =0...7; u = 0... 7, is represented on an 8-bit integer. The operator /’ represents the integer division with truncation of the result towards zero.

One of the main uses of QS is for bit rate adaptation. The higher the QS value, the lower the bit rate, but a lower bit rate means a less picture quality, therefore the QS value must be chosen so as to minimize perceived distortion in the reconstructed picture.

In an encoder, the QS can be changed at the start of coding of each macroblock. Each time it is changed, the new value must be coded in the bitstream and there is coding overhead in doing this. In the case of IPTV this is done using MPEG-2 TS packets. Therefore, the QS can be retrieved very simply in the IQ block of the MPEG decoder, without adding extra devices, in an IPTV receiver (Set-Top-Box).

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IV. SYSTEM ARCHITECTURE

The system is divided into two main parts: the server level and the user level. Both are linked by a communication network based on TCP / IP.

At the user level, the system is based almost in the preprocessing and transmission of uncompressed images in PNG format with a resolution of 1920x1080. The videos were generated from 14315 uncompressed PNG images [23] and encoded and quantized using ffmpeg software [24]. We made 8 video sequences with the following characteristics:

- MPEG-2 and MPEG-4 coding
- 720x576 of resolution
- 25 fps
- 120 seconds long
- QS {1, 2, 4, 6, 8, 10, 16, 31}, see [13].

Figure 3 shows the steps from the preprocessing of encoding of the uncompressed PNG images until they are transported to the end user. In the preprocessing stage, after the PNG images are encoded, we obtain a standard definition television (SDTV) MPEG1 video with a resolution of 720x576 dpi. This video will be the reference video used for the different tests. This video is later compressed in MPEG-2 and MPEG-4 format. Then, they are quantized to different QS for each format. In the transport stage, the videos are packaged in MPEG-2 TS and transmitted by the server in broadcasting to the network. We used VLC Media Player [25] as IPTV video server to send the SDTV MPEG-2 Transport Streams for both compressions types.

The IPTV network (shown in Figure 3) has been simulated using a PC placed in a Fast Ethernet network. The final user has a commercial set-top box to watch the video from the TV. It can also be observed directly from PC.

At the user level, we have also installed the VLC Media player. There, we can see the captured video and measure it subjectively. On the other hand, we installed the Clearsight Analyzer Software [26] in user's PC. It lets us capture and analyze the MPEG-2 TS packets. We measured the video quality at the receiver, and estimated the Mean Opinion Score (MOS). There is also installed a sniffer at the user level, which allows us to obtain the QoS parameter values of the video transmission (jitter, delay and lost packets). These measures will serve to evaluate the QS.

Our network architecture uses IPv4 because nowadays the most of commercial set-top boxes only use IPv4, but it can be implemented in an IPv6 network with IPv6 set top boxes easily.

V. QS ANALISYS

In order to analyze the optimal QS, we encoded several videos with the different QS values shown in the previous section. The Video Quality (VQ) results, obtained by the IPTV analyzer software of each video transmitted through our test bench, is shown in Figures 4 and 5. The VQ is measured in MOS values. In order to obtain the MOS, we compare the captured video with the video reference. MOS is an ITU (International Telecommunication Union) standardized term [27], used as a methodology for the subjective assessment of the quality of television pictures. MOS scores are rated on a scale from 1 to 5, where 5 is the best possible score, and indicates the degree of the user's satisfaction. In Figure 4 and 5, the best scores are obtained for QS {1, 2}, which provide a medium-high VQ value (4 at the MOS scale), while the rest of QSs have VQ values below 4. So, we can say that optimal QS, is 1 or 2 according our needs. If we want the best video quality without considerer the file size we will choose 1, but if we take into account the size of the video, we will select QS equal 2. However the tests were conducted using fixed QS in the video encoder. But in practice, a commercial MPEG encoder often uses variable QS, which assigns different values to intraframe or interframe, even to macro-blocks contained in each of them. For that reason, an average QS of 4 could be the highest value used by a video encoder. It seems that, according to the values of Figure 4 and 5, QS values upper than 4 will give fair or poor VQ. To check the results, we used an objective video quality method [28]; which is used traditionally to determine the video quality in presence of impairments. This will be discussed at the next section. Furthermore, in order to analyze, which is the best QS for video streaming, we used new testing measures as jitter, delay and packet lost, which will be explained in next sections.

Another important aspect in the QS analysis is the used instantaneous bandwidth. In Figures 6 and 7, the bandwidth used by MPEG-2 and MPEG-4 videos, and their respective QS, is shown. The appearance of traffic peaks on all graphs, with the same behavior, indicates that the tests have been conducted with the same sequence of images. Analyzing MOS and BW results we see that Q1 and Q2 have practically the same quality and features. According to [29], a MOS between 3.1 and 4.0 is acceptable. These videos are high quality, but the bandwidth needed for transmission is very high. Quantification values higher than Q8 are below the acceptable MOS. This reflects will serve for further analysis.

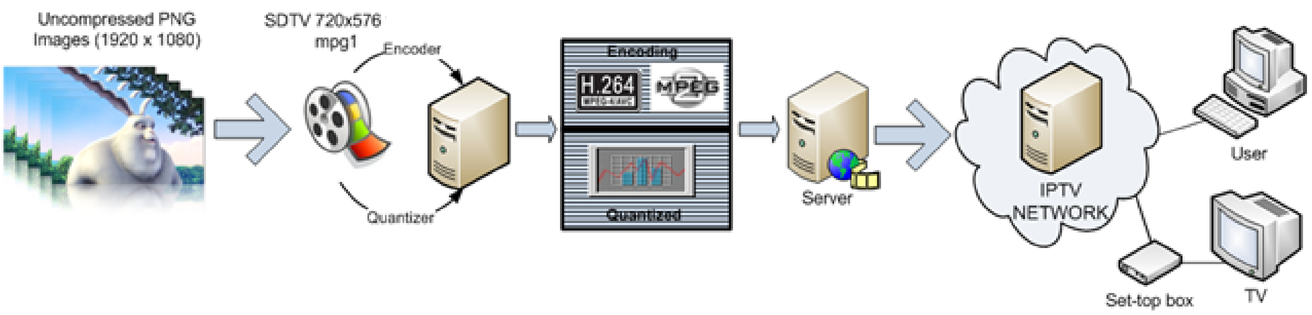

Figure 3. Encoding process and location of IPTV devices

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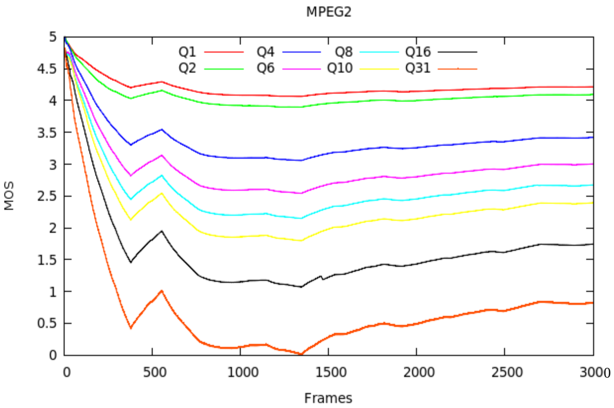

Figure 4. Video Quality in MOS value using the MPEG-2 codec quantizer

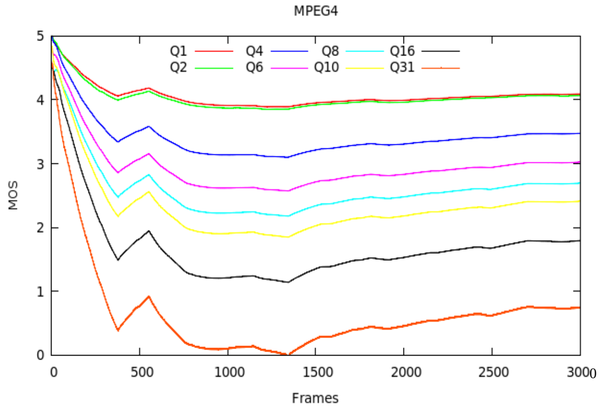

Figure 5. Video Quality in MOS value using the MPEG-4 codec quantizer

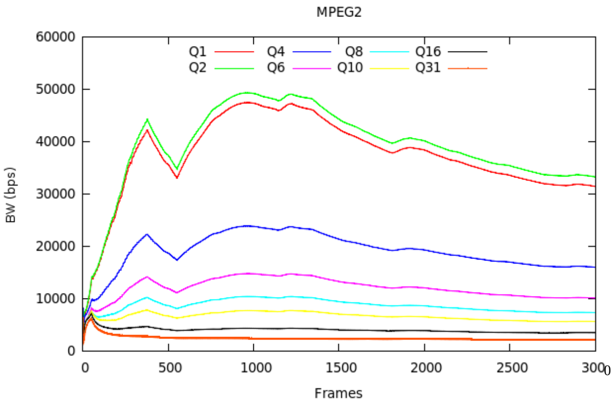

Figure 6. Bandwidth used by MPEG-2 video with different QS

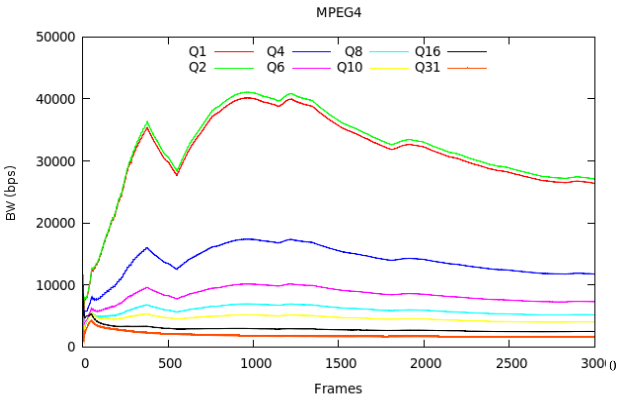

Figure 7. Bandwidth used by MPEG-4 video with different QS

VI. OBJECTIVE AND SUBJECTIVE VIDEO QUALITY

The goal of the objective video quality assessment is to design quality metrics in order to predict the perceived video quality automatically. The subjective video quality is a tool for the evaluation of videos from the point of view of the observer.

A video signal, whose quality is being evaluated, can be thought of as a sum of the reference signal and an impairment signal. We may assume that the loss of quality is directly related to the strength of the impairment signal. Therefore, a natural way to assess the quality of an image is to quantify the error between the distorted signal and the reference signal, which is fully available in Full Reference (FR) quality assessment [28]. But this is a problem because these videos of reference require a large amount of storage and, in many cases, it is impossible to obtain it. Reduced-reference (RR) [30] quality assessment does not assume the complete availability of the reference signal, only a partial reference information that is available through an ancillary data channel. Partial reference information could be Packet Loss Rate (PLR), is the probability that a packet is dropped at any router, or I/B/P Frames Statistics Losses (FSL), while in our case is QS.

In our case we will quantify the error between the distorted signal and the reference signal using Video Quality Metric VQM [31]. VQM uses the zero value as the best possible value; this means that there is no error between the reference video and the impairment video. The results of MPEG-2 and MPEG-4 are shown in Figures 8 and 9 respectively. In both Figures, the best value is QS1 and the worst QS31, although the difference is lower in MPEG-4. This verify that the larger the QS, the lower image quality (as we mentioned in the previous sections). We can see that when there are errors, the MPEG-4 video is seen with higher quality image than the MPEG-2 video because lower values of VQ are obtained. Again, QS2 was the optimal value.

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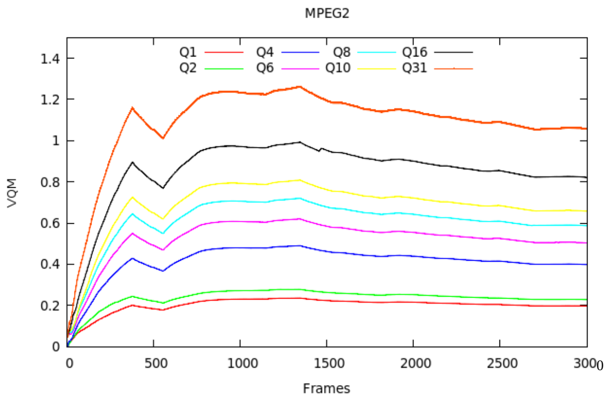

Figure 8. Objective video quality for MPEG-2

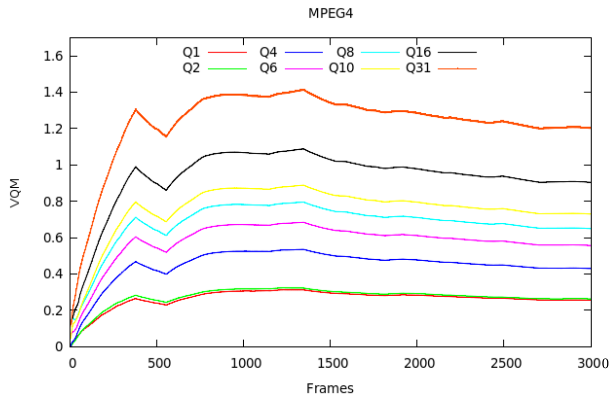

Figure 9. Objective video quality for MPEG-4

Figure 10. Visual comparison between a video with lost packets (at the bottom), QS31 video (in the middle), and the reference video (at the top)

Figure 10 shows a visual comparison between a video with lost packets (at the bottom), QS31 video (in the middle) and the reference video (at the top). In the video with loss packets we can see that when losses occur in the network, there is a lack of information in the video and pixels appears empty or with adjacent information. In the QS31 video, we can see that it has color degradation, figure pixelation, and it loses clarity. These issues are given due to the high value of QS. This information allows us to compare the video received with the reference. Videos with QS31 and QS16 do not exceed the minimum requirements of subjective assessment because the end user does not get a good picture quality.

VII. JITTER, DELAY AND LOSS PACKETS TEST

The latest set of tests conducted in this work has been to measure the jitter, delay and lost packets. The aim is to find how these parameters affect the QS. Finally, we will gather all the measurements and select an optimal value of QS. In order to perform this experiment, we added changes in the network parameters by applying different values of jitter, delay and lost packets in the video transmission in a controlled manner. It has been done by using the software NetDisturb [32]. Based on our previous experiments [13] we will only analyze QS1, QS4, QS6, QS8 and Q10. We will cause 0.1% ,1% and 3% of loss packets in the network during the a IPTV channel transmission because higher values than these ones, give very low video quality results as it is shown in Figure 10 (bottom). The results can be seen in the following test bench.

In Figure 11, the average delay when there is a loss of 0.1% for several values of quantizer (Q) in the MPEG-2 encoding can be observed. When the value of Q increases, the instant average delay is higher. The behavior of the delay is the same regardless of the Q value. If we have a low Q the file size to be transmitted is greater, so the number of packets to transmit will be bigger. For example, when Q=1 we have an average delay lower than 2 milliseconds, but the number of packets transmitted is approximately 70000. But when Q=6, the delay is lower than 5 milliseconds and the transmitted packets are approximately 24000.

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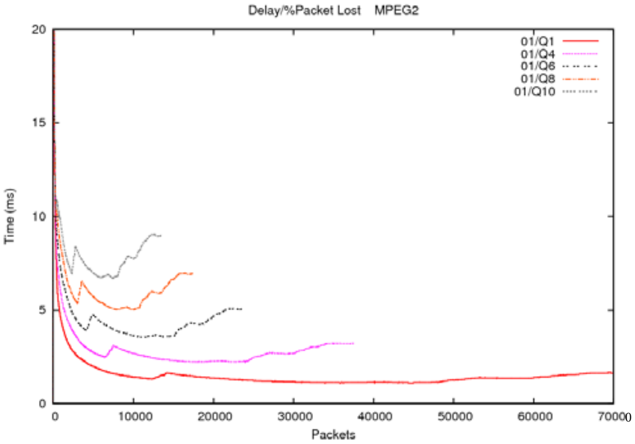

Figure 11. Delay when the loss rate is 0.1% for different Q of MPEG-2 videos.

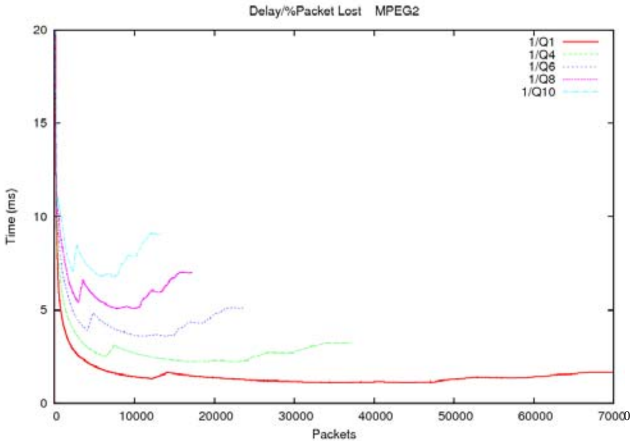

Figure 12. Delay when the loss rate is 1% for different Q of MPEG-2 videos.

In Figure 12, we can see that we obtained similar results, but in this case the losses introduced in the network were 1%. We can say that in a network with 0.1% and 1% losses, in the case of MPEG2 encoding, the behavior of the delay is very similar.

Finally, in Figure 13, we show the delay for a network with a loss of 3%. In this case, the delay follows the same behavior such as the previous figures but the average values are slightly smaller when there is a low Q. In this case, the delay increases for higher Q compared to Figures 11 and 12. This phenomenon can be seen when we have Q=1 (the delay decreases compared to the delays with a loss of 1% and 0.1%) and Q=10 (the delay increases)

We have also evaluated the delay compared to a loss rate for different Q of MPEG4 video encoding (see Figures 14, 15 and 16). With a loss rate of 0.1%, with Q=1, the observed delay is lower than 2 milliseconds, while with Q=4 this delay increases almost up to 4 milliseconds. Moreover, as we see in Figure 14, increasing the value of Q, the delay is also increased. This behavior is also obtained in the figures related to MPEG-2.For a network loss rate of 1%, the behavior of the delay is the one obtained in Figure 15. In this figure we see that for a Q=1 the delay is lower than 2 milliseconds and in the case of Q=4 this delay is lower than 4 milliseconds. With a loss rate of 3% in MPEG-4 videos, we obtained the delay shown in Figure 16. In this case, we see that the delay is slightly higher compared to loss rates of 0.1% -1%. This difference in delay is very small but exists.

As a general conclusion related to the delay measurements, we can say that, regardless of the codec used, there are Q values, which contribute with lower delays, but the number of transmitted packets is higher when we use small QS. Besides, this delay is referenced to belay between packets. For this reason when we have more packets, the intermediate devices will need more resources to give the same service. In all cases, the delay is lower than 20 milliseconds, an acceptable delay in the transmission of IPTV channels.

In the Figures 17, 18, 19, 20, 21 and 22, we can observe the instantaneous average jitter using MPEG-2 and MPEG-4 encoded videos for different loss rates. In Figure 17, we analyze the jitter for a loss rate of 0.1%. In this case, we observe that the measurements obtained have an exponential behavior. It can be seen that the video with Q=4 has the smallest jitter, something that can be taken into account for the IPTV transmission.

When we have a loss rate of 1%, the behavior of the jitter follows the graph shown Figure 18. In this case, the jitter has an exponential behavior. We have also observed that when Q increases, the jitter is reduced. For an efficient IPTV transmission, it is the better to have a small jitter, but we can't choose the highest Q value because the video quality is very poor.

Figure 19 shows the jitter measurements obtained for a rate loss of 3%. In this case, we see that the behavior is not the same as that obtained in the previous figures. This is because packet losses are already quite high and the jitter doesn't follow any specific distribution. We observed that the Q=4 value has the smallest jitter values (around 1500 packets), then this jitter value increases. We must indicate that the jitter values obtained in this figure are very small and therefore they will not affect to the transmission of IPTV channels.

In the MPEG4 encoding, the jitter obtained for different Q and a loss rate of 0.1% is seen in Figure 20. In this case, the behavior of this parameter is also exponential. In figure 20 we observe that the values of Q=1 and Q=6 are the ones that provide lowest jitter. In this figure, we can see that the value of Q=10 has the worst jitter value introduced into the network. Figure 21 shows the obtained jitter for various Q values when the loss rate increases to 1%. In this case, the Q value that introduces more jitter to the IPTV transmission is the Q=8, having the other cases much lower jitter values.

Finally, the jitter obtained for a loss rate of 3% is represented in the Figure 22. In this case, we see that the behavior doesn't follow any pattern. The Q value, which introduces lowest jitter is when we encode the video with Q=8. Otherwise, when there is higher jitter, the Q=1 is the quantizer, which provides higher jitter values to the IPTV transmission.

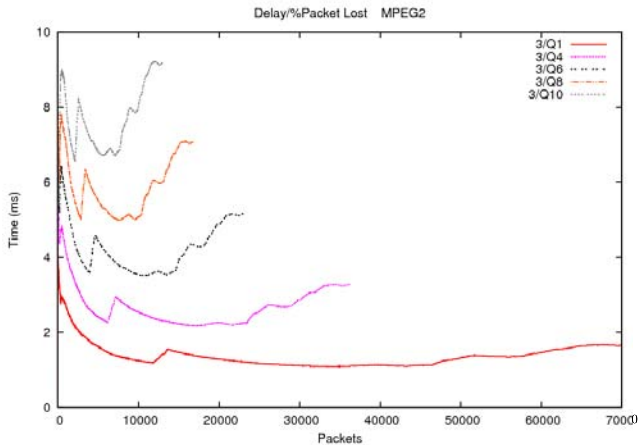

Figure 13. Delay when the loss rate is 3% for different Q of MPEG-2 videos.

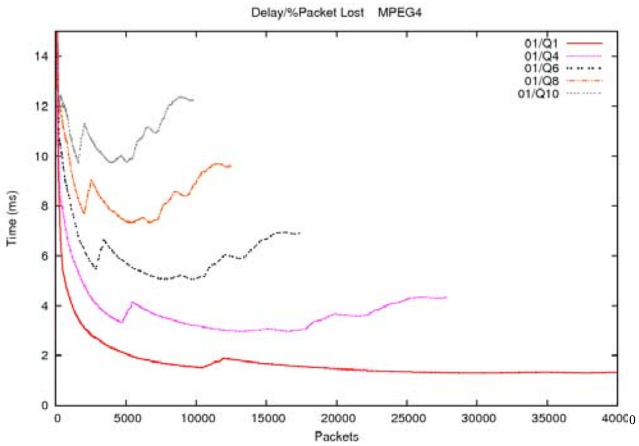

Figure 14. Delay when the loss rate is 0.1% for different Q of MPEG-4 videos

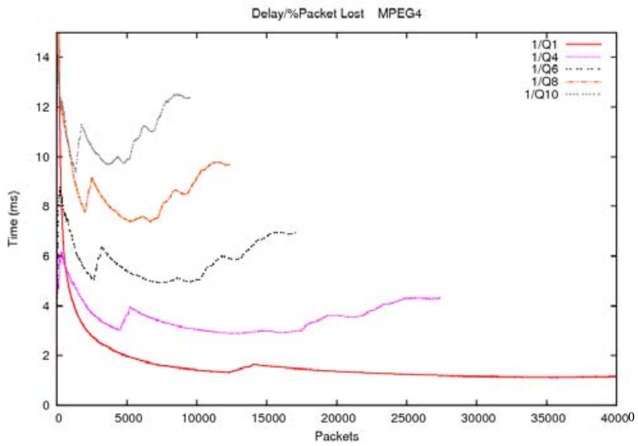

Figure 15. Delay when the loss rate is 1% for different Q of MPEG-4 videos

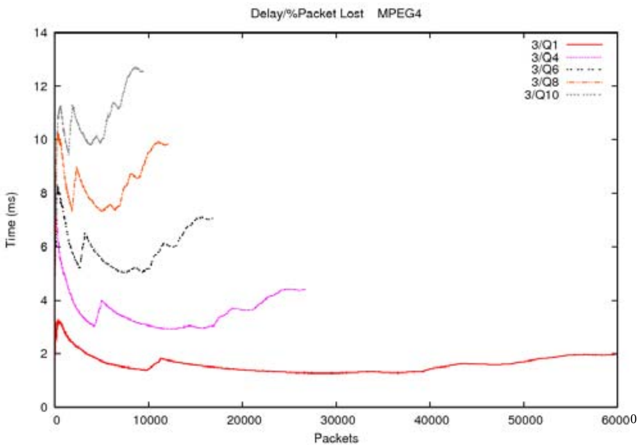

Figure 16. Delay when the loss rate is 3% for different Q of MPEG-4 videos

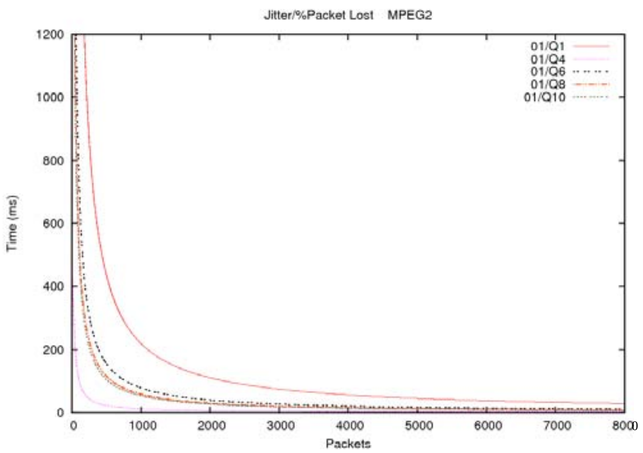

Figure 17. Jitter when the loss rate is 0.1% for different Q of MPEG-2 videos.

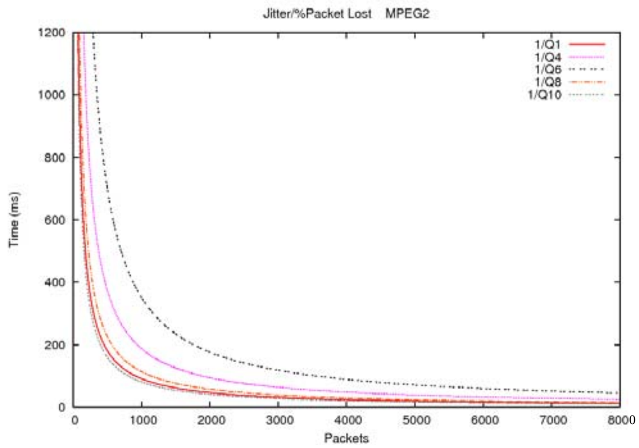

Figure 18. Jitter when the loss rate is 1% for different Q of MPEG-2 videos.

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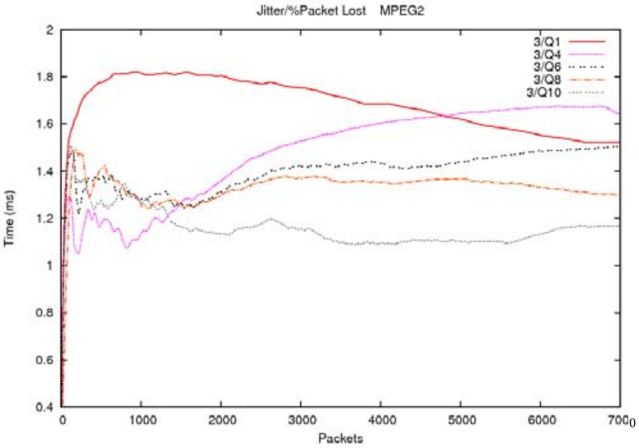

Figure 19. Jitter when the loss rate is 3% for different Q of MPEG-2 videos.

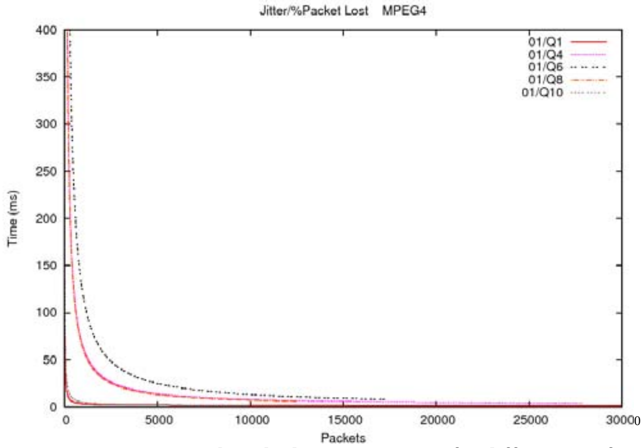

Figure 20. Jitter when the loss rate is 0.1% for different Q of MPEG-4 videos

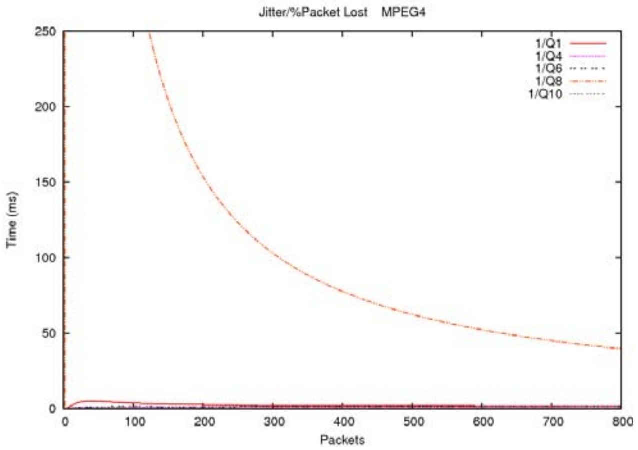

Figure 21. Jitter when the loss rate is 1% for different Q of MPEG-4 videos

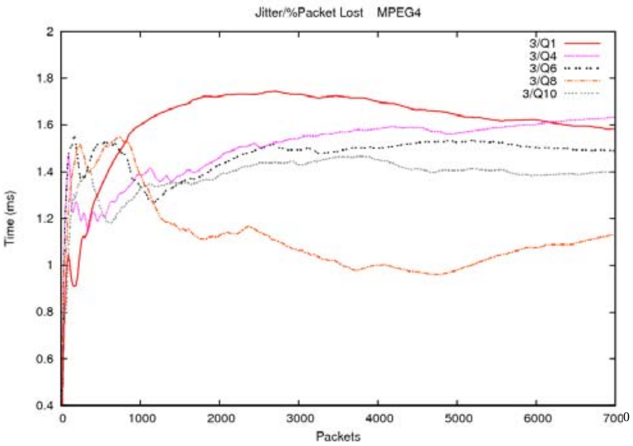

Figure 22. Jitter when the loss rate is 3% for different Q of MPEG-4 videos

VIII. DISCUSSION

Authors of reference [33] recommend a maximum loss of 5 consecutive IP packets every 30 minutes in SDTV. They say that between 0.5% and 1.5% of packet loss is acceptable. We have taken this into account when we made our test. For this reason we don't presented loss packet values higher than 3%.

Authors of reference [34] recommend that delay bounds for the various grades of perceived performance in terms of human interaction can be defined as: Good (0ms-150ms), Acceptable (150ms-300ms), Poor (> 300ms). Moreover, the authors of reference [29] suggest the following jitter values to be reasonably reliable to determine the grade of perceived performance: Good (0ms-20ms), Acceptable (20ms-50ms), Poor (> 50ms).

The jitter values of our test bench ranges between 200 ms and 2 ms and, for delay values between 2 and 30 ms. Therefore, we see that there are videos that don't satisfy this consideration, like the ones p given by QS31.

We made a subjective analysis of the video quality. It is shown in Figure 23. In the first line of the figure we see an image of the reference video (first in the left), an image of the QS4 video (second) and the difference between them (third). In the second line we see an image of the reference video (first in the left), an image of the QS6 video and the difference between them.

With this information and the information taken from the previous sections, we can deduce that the optimum values of QS are both QS4 and QS6 for MPEG-2 and MPEG-4 videos. Taking into account the extreme values of QS31 and QS1, we built the comparative shown in table I. In that table the average values for the different measures taken respect to the optimal values of QS31 and QS1is shown.

IX. CONCLUSION

In this paper, we have analyzed the QS as visual quality parameter. QS can be calculated at the decoder by extracting the information encoded in MPEG-2 TS packets, and, then, it could be used in VQ in order to create a reduced reference model to be used in the estimation of the QoE of the user .

Figure 23. Subjective Video Quality for QS4 (first line) and QS6 (second line).

TABLE I. AVERAGE VALUES WITH SEVERAL QS FOR MPEG2 AND MPEG4 CODECS

		Jitter_01% (ms)	Jitter_1% (ms)	Jitter_3% (ms)	Delay_0.1% (ms)	Delay_1% (ms)	Delay_3% (ms)	BW (bps)	MOS
QS1	**2**	74.98	10.41	1.4	1.86	1.57	1.37	37355.7	4.20
	4	23.98	10.41	1.48	1.75	1.57	1.6	31452.7	4.06
QS4	**2**	3.95	41.87	1.65	2.98	3.01	2.68	19000.93	3.36
	4	18.96	1.58	1.6	3.88	3.56	3.61	13898.6	3.40
QS6	**2**	25.99	48.57	1.37	4.65	6.42	13.17	12029.7	2.92
	4	50.23	1.39	1.42	6.31	5.87	5.99	8406.34	2.94
QS31	**2**	91.84	143.69	1.29	21.84	22.11	20.92	2377.25	0.7
	4	204.39	136.53	1.39	26.73	26.98	26.99	1833.02	0.64

We reached the conclusion that QS4 and QS6 are the optimum values when we include changes in the network and the network conditions are not optimum. Moreover, QS4 provide better video quality. IPTV service providers will require less bandwidth when QS4 and QS6 MPEG-4 videos are stream to the network. Therefore, we have shown that an optimal quantizer_scale factor can be used to save bandwidth in an IPTV network or to improve the Video Quality for the same bandwidth consumption.

In [13] we have demonstrated that the best QS value was equal to 2. However, in this paper we have added the bandwidth, jitter, delay and packet loss measurements in order to test the IPTV channel performance and the VQM and MOS. Now, we have demonstrated that the best QS values have been QS4 and QS6 when these parameters are included between the parameters taken into account. Moreover, in [13] we used a DVD video as a reference and then we codded it to MPEG-2 and MPEG-4, in this paper we have coded the raw video to MPEG-2 and MPEG-2 directly.

Reduced reference models are the next challenges for perceptual visual quality measurement techniques in multimedia services over digital television networks. Our future work will be focused on adding these results to the VQ algorithms in order to produce an efficient QoE to the user.

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