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MASTER THESIS

Application of SI frames for H.264/AVC Video Streaming over UMTS Networks

by

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Abstract

In mobile video applications, frames can be encoded using temporal dependencies (to other frames) or spatial dependencies (to the same frame) to reduce the bitrate. Frames with temporal dependencies are more commonly used because of their smaller size, however, in case of error in transmition, it propagates troughout the following frames.

H.264/AVC defines two new frames, SP (with temporal dependencies) and SI (with spatial), and both of them are interchangeable if an error occurs, stopping its propagation.

This work shows the influence of the two quantization parameters of SP and SI frames (SPQP and PQP) on its size and quality, obtaining a conclusion of which values use depending on the probability of switching.

Moreover, this project presents the implementation of the switching process depending on the error information obtained from the UMTS network. The error information is detected by the Mobile Station and sent back through a feedback channel with a certain delay. The effect of the error and the delay is simulated in order to evaluate different situations.

The switching process is implemented in a Matlab program. Simulations show a save in the bitrate without decreasing the quality with respect not to use SP and SI frames in case of a low probability of switching and with a small delay in the feedback channel.

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Chapter 1

Introduction

The great growth of multimedia applications for mobiles in the last years has obliged to investigate in order to enhace the rate-distortion efficiency without increasing dramatically the complexity of the systems [1]. In video applications, this enhacement has been obtained by improving the existing codecs, adding new features. Temporal dependencies (to other frames) or spatial dependencies (to the same frame) are utilized to reduce bitrate. Temporal dependencies are more commonly used with this purpose because of their smaller size. However, in case of an error in transmition, it propagates troughout the following frames while decoding. Thus frames with spatial dependencies are sent regularly to recover good quality in these cases.

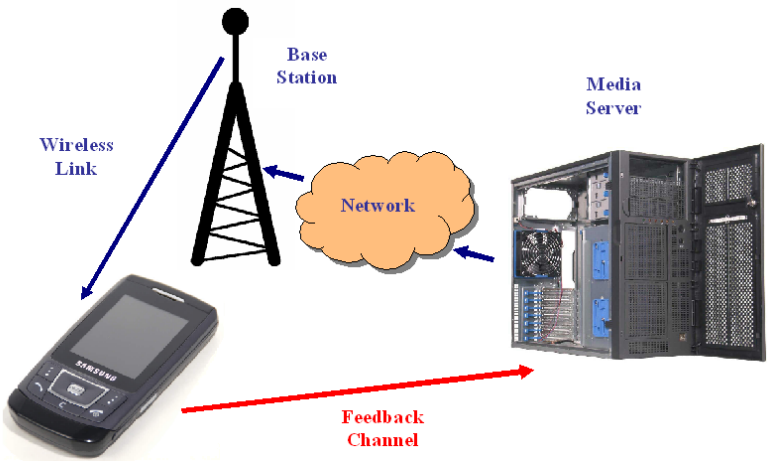


Figure 1.1: Network diagram

In this project, CIF (352x288 pixels resolution) and QCIF (176x144 pixels resolution) videos are encoded with the H.264/AVC codec [2] and transmitted over a UMTS network. H.264/AVC contains two new frame types, SP (with temporal dependencies) and SI (with spatial dependencies). The main difference with the previously existing frames is that SP and SI are interchangeable in case of having an error in the link, stopping error propagation when decoding [3].

This work studies the influence of the two quantitation parameters of SP and SI frames (SPQP and PQP) on its size and quality, and tries to understand the close

relation between these values and the probability of swtiching.

Part of this project consists of implementing the switching process as a completely independent program from the encoder. Therefore, two bitstreams, one with SP and the other with SI frames inserted, are encoded in the transmitter in order to facilitate the switching. The (switching is carried out depending on the error information obtained from the UMTS network (Figure 1.1). The Mobile Station detects the error, which occurs normally in the wireless link, transmitting this information through a feedback channel with a certain delay. Video applications make use of UDP protocol and consequently no retransmission is possible, receiving erroneous packets in reception in case of any transmission failure. This system aborts a sure error propagation in the decoder produced by the usage of temporal dependences. However the propagation of this error in the decoder is avoided with this system.

The effect of the error and the delay are simulated to evaluate different situations, and together with the switching process are implemented in a Matlab program. Simulations show a bitrate saving without decreasing the quality with regard not to use SP and SI frames in case of a low probability of switching and with a small delay in the feedback channel.

Chapter 2

H.264/AVC

H.264/AVC is the newest video coding Standard resulted from the cooperation of the ITU-T Video Coding Experts Group (VCEG) and the ISO/IEC Moving Picture Experts Group (MPEG) in a common project known as the Joint Video Team (JVT) [4] [5]. The aim of this project is to enhace the rate-distorition efficiency increasing the compression and the objective and subjective quality of the encoded video compared to previous standards (MPEG-2, MPEG-4, H.263) without increasing dramatically the complexity.

H.264/AVC is designed to work with different types of networks and to support a wide range of applications (video telephony, storage, broadcast or streaming).

2.1 Origing of a new standard

The MPEG-2 video coding standard was developed about fifteen years ago as an extension of MPEG-1, supporting interlaced video coding and at those days this technology was enable to distribute digital TV over satellite (DVB-S), cable (DVB-C) and terrestrial (DVB-T) platforms for transmision of standard definition (SD) and high definitions [6].

However, nowadays the number of services offered has increased and also the demand of high definition TV. Moreover, new transmision media like xDSL, Cable Modem or UMTS offer much smaller data rates and thus a more efficient data compresion has been required. During the development of the different standards, continued efforts have been made to deal with the diversification of the network types and therefore their formatting and robustness needs.

In 1998 the Video Coding Experts Group (VCEG - ITU-T SG16 Q.6) started a project called H.264L, with the aim of double the coding efficiency of the existing video coding standard at that moment. In 2001 VCEG and the Moving Pictures Expert Group (MPEG - ISO/IEC JTC 1/SC 29/WG 11) formed the Joint Video Team, with the target to finalize the new video coding standard H.264/AVC.

2.2 General Features

All the standards from ITU-T and ISO/IEC JTC1, and also H.264/AVC follows the block-based hybrid video coding approach, consisting in dividing the picture in macroblocks, informing about luma and chroma. H.264/AVC compresses the transmitted data by exploiting temporal and spatial dependencies, using inter-picture prediction (temporal) and intra-picture prediction (spatial).

The extensions to the original standard developed by the JVT are known as the Fidelity Range Extensions (FRExt). Some of the new features supported by these extensions are: higher-resolution color information, increased sample bit depth precision, adaptive switching between 4x4 and 8x8 integer transforms, encoder-specified perceptual-based quantization weighting matrices, efficient inter-picture lossless coding, and support of additional color spaces.

The standard allows to use different sets of coding tools, known as profiles [1], depending on the application searched:

Five new profiles are supported in recent extensions and they are used for professional applications and high definition video (HD). The profile used in this work is the extended profile (Figure 2.1). Eventhough it is not recomended for mobile applications because of its computational complexity, it is the only one that allows working with SP and SI frames.

- **Baseline Profile:** For low-cost applications with limited computing resources, specially used in video conferencing and mobile applications. All features of

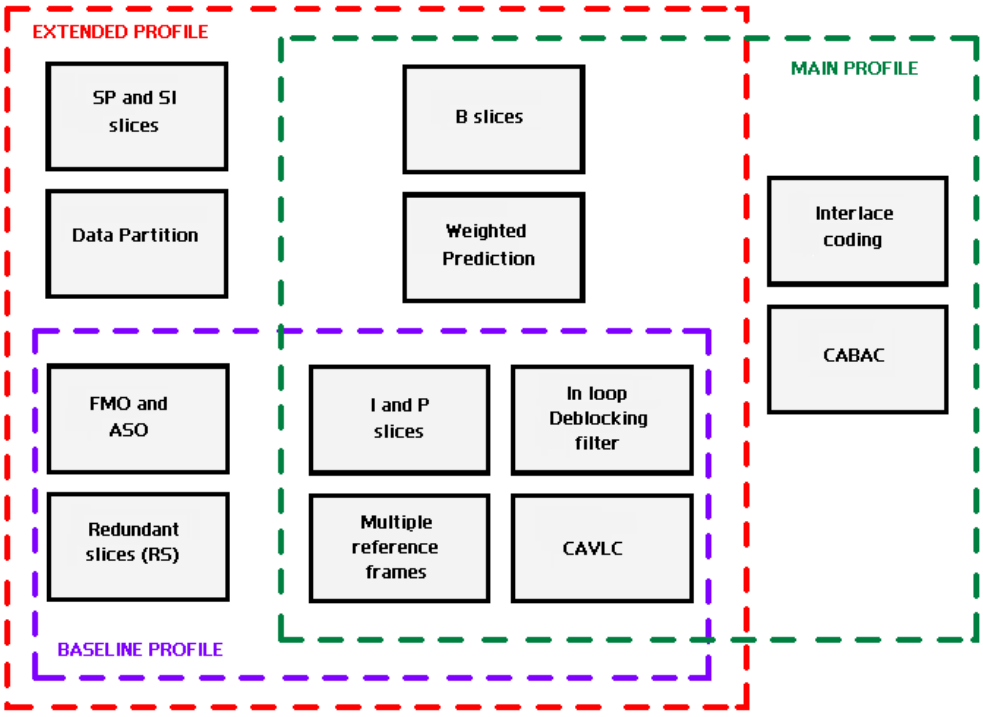


Figure 2.1: Utilities for H.264/AVC profiles

H.264/AVC are supported except B, SP and SI slices, weighted prediction, CABAC, field coding and macroblock adaptive switching between frame and field coding.

- **Main Profile:** Used for broadcast and storage applications, is the mainstream consumer profile. It does not support the FMO feature.
- **Baseline Profile:** Intended as the streaming video profile, this profile has relatively high compression capability and some extra tricks for robustness to data losses and server stream switching.

As all the previous cases of the ITU-T and ISO/IEC, only the decoder is standardized, producing similar output for an encoded bitstream in its input that conforms the constraints of the standard on bitstream and syntax. This point has its advantages, as for instance the freedom to optimize the different applications; however it does not guarantee an end-to-end reproduction quality.

The video to be codified in a raw format is the input of the encoder (Figure 2.2). The encoded sequence is the output of the encoder to be transmitted. It is also possible to see the effect that the encoding and decoding process have in the quality of the video since the reconstructed video is also obtained in the encoder. This outputs represents the received and decoded video in case of no errors in transmission. Another output is the trace file, which explains deeply the encoded sequence obtained.

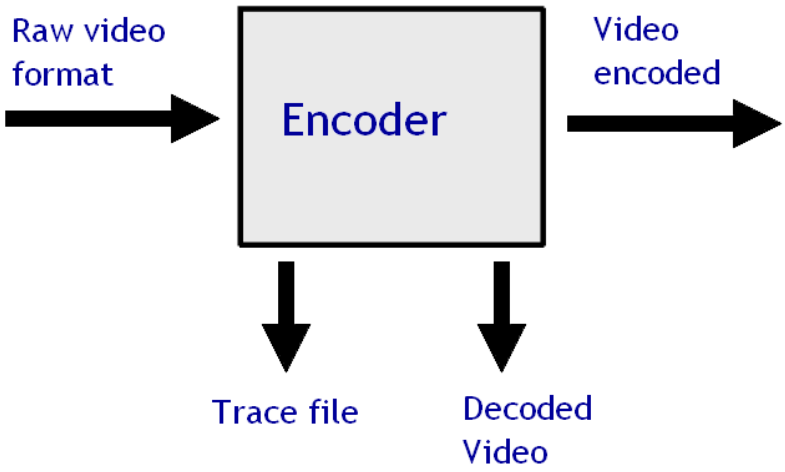


Figure 2.2: Encoder inputs and outputs

The H.264/AVC design is divided into two parts, the Video Coding Layer (VLC), which represents the video content, and the Network Abstraction Layer (NAL), which provides a format to the VLC and describes the headers added, depending on the storage media or the transport layers used.

2.3 NAL

The NAL describes the packetization process of the encoded video (VLC) and is designed to enable simple and effective mapping to differernt transport layers as RTP/IP, H.32X and MPEG-2 systems for broadcasting services. A description of the different concepts of the NAL is given below.

The encoded video is organized into packets called NAL Units, divided into a payload and a header. The header is the first byte of the packet and informs about the type of data inserted in the payload. The structure definition of the NAL Unit specifies the format for packet-oriented or bitstream-oriented transport packets. A number of NAL Units forms a NAL unit stream.

In case of using a byte-stream format, three bytes prefixe every NAL unit to define its boundaries and also some alignemt data is possible to be added if the system does not provide it. In case of using a Packte-Transport System, the boundaries of NAL units are withing the packets.

There are two types of NAL Units, VCL and non-VCL NAL Units. VCL NAL Units represent the values of the different samples of the encoded video, and non-VCL NAL Units contains supplemental information to enhace the whole process like timing information and also the parameters sets.

The parameters sets are a group of information useful for a number of NAL Units. Since this information rarely change [6], there is no need to send it with every NAL Unit, saving bitrate. There are two types :

- **Sequence parameter sets:** Information valid for several consecutive coded video pictures called a coded video sequence.
- **Picture parameter sets:** Information valid for one or more individual picture within a coded video sequence.

Every VCL NAL Unit has an identifier to refer to a picture parameter set and each Picture parameter set has another identifier to refer to a Sequence parameter set. Sequence and Picture parameter sets are sent before the VCL NAL Units to which they refer. Depending on the application, these parameters are sent withing the same channel than the VLC NAL Units or through a more reliable transport channel.

All the NAL Units of a encoded picture compose forms an Access Unit. It may also contain an Access Unit delimiter and some additional information. In the Access Unit, the VCL NAL Units contained, are known as the Primary Coded Picture. An *end of sequence* NAL Unit is sent if the coded picture is the last picture of a video sequence and an *end of stream* NAL Unit if the stream is ending.

A coded video sequence is a group of Access Unit with only one Sequence parameter set which can be decoded independently. At the begining of the Coded Video Sequence, an *instantaneous decoding refresh* (IDR) Access Unit is sent. An IDR is an intra picture that can be decoded without refering to any picture, and prohibits next Access Unit refer to previous pictures. A NAL Unit stream may contain more than one Coded Video Sequence (Figure 2.3).

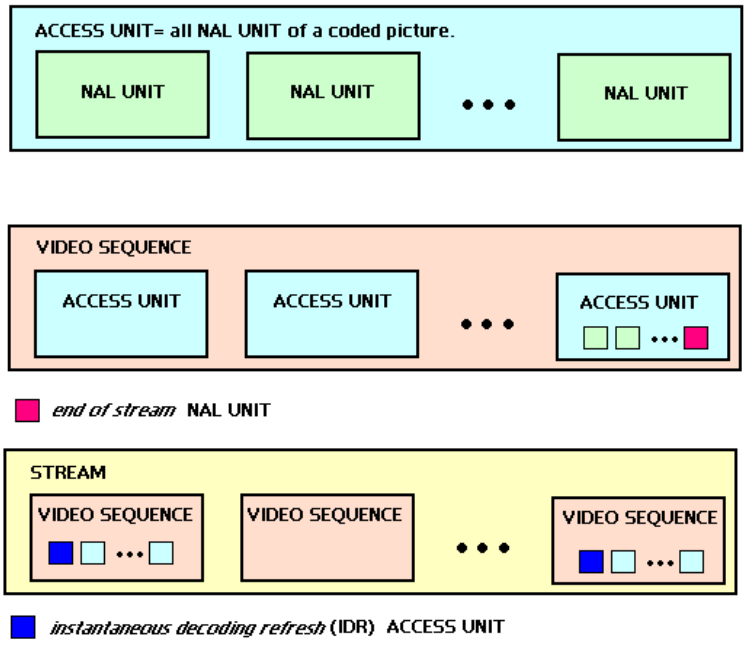


Figure 2.3: Packetization process

2.4 VLC

2.4.1 Pictures, slices and macroblockes

The VCL represents the video data and, as in all prior standards, follows the hybrid video coding approach. A coded picture is divided into samples of croma and luma called macroblocks. The final improvement depends on a big number of small steps rather than on a single coding element.

A coded picture contains two interleaved fields, one with the odd-numbered rows and the other with the eve-numbered. In case the two fields are captured in different times, the frame is known as an interlaced frame, otherwise it is a progressive frame. Chroma and luma information is extracted from the picture because humans perceive an scene in terms of brightness and color with greater sensibility to brightness. H.264/AVC separates a color representation into three components, Y, Cb and Cr. Y is luma and represents brigthness and Cb and Cr is the deviation of the color from gray toward blue and red, respectively. Chroma has one fourth of the luma samples (half of the number of samples in the horizontal dimension and the other half in the vertical dimension) because of the less sensitivity to color and thats called 4:2:0 sampling with 8 bits of precision per sample.

A picture is divided into samples and these are grouped into macroblocks. Luma macroblocks correspond to 16x16 samples of a rectangular part of the picture and the two chroma components of a picture are divided into macroblocks of 8x8 samples. Macroblocks are the basic buiding blocks of the standard.

A picture can be divided into slices. Slices are sequences of macroblocks which can be decoded independently from other slices even though they belong to the same

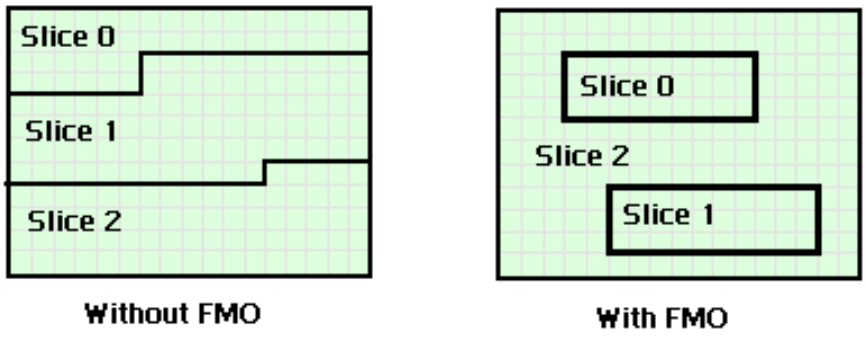


Figure 2.4: Subdivision of a picture into slices.

picture. The transmission order of macroblocks in the bitstream depends on the Macroblock Allocation Map. FMO is a feature of H.264/AVC which allows the division of a picture into slices depending on the content of the image (Figure 2.4). This concept is called *slice groups*. The specifications of the slices are defined in the Picture parameters set and in the slice headers. Each macroblock have a slice group identification number to inform to which slice it belongs. In this work FMO is not used and macroblocks are allocated in the slices in raster order.

Five different types of slice-coding [6]are supported by H.264/AVC:

- **I slice:** All macroblocks from the are encoded using intra prediction, what means that do not refer to other pictures.
- **P slice:** Macroblocks from the slice can be encoded as intra but also using inter prediction, refering to other pictures, with at most one motion-compensated prediction signal per prediction block.
- **B slice:** All coding types of a P slice are allowed and also biprediction, using inter prediction with two motion-compensated prediction signals per prediction block.
- **SP slice:** Similar to a P slice but allows to be switched by an SI frame.
- **SI slice:** Similar to an I slice but allows an exact mach with an SP frame for random access and error recoery applications.

The last two types of slice have been introduced by H.264/AVC. As a general scheme, all macroblocks are predicted spatially or temporally, calculating its residual. This residual is subdivided into smaller blocks of 4x4 and these blocks are transformed. The coefficients resulting of this transformation are quantized and encoded using entropy coding methods.

2.4.2 Intra-Frame prediction

In blocks or macroblocks encoded using intra modes, the predicted block is formed based on previously encoded and reconstructed blocks from the same picture [7]. There are three intra coding types, Intra_4x4, Intra_16x16 and I_PCM, and all of them are permitted by any slice-coding type. Using Intra_4x4 means dividing a macroblock in 16 blocks to carry out the prediction. Due to the small size of these macroblocks there is a more accurate prediction, perfect for those parts of the picture that wants to be encoded in detail. Since macroblocks for prediction of Intra_16x16 type are bigger, they are used to predict smoth areas of the picture. Finally, the I_PCM coding type transmits directly the values of the encoded process.

Intra prediction uses spatial dependencies, what implies the usage of neighboring samples already encoded for prediction. A constraint intra coding mode can be activated to allow prediction only from intra macroblocks of the same slice, avoiding propagation of errors from motion compatation.

Intra_4x4 type permits for each 4x4 block one out of nine prediction modes, DC mode and eight directional modes. In DC prediction, the predicted value is obtained by an average of the adjacent samples. In directional modes the values of the blocks selected by the direction are directly copied. The directional values are horizontal, vertical and diagonal and they are suited to predict textures with structures in the specified direction.

Intra_16x16 only allows 4 prediction modes, DC, vertical, horizontal and plane prediction. Luma samples can make use of both Intra_4x4 and Intra_16x16 type while chroma samples are predicted using Intra_16x16 since chroma is usually smooth over large areas. Intra prediction mantains slices independent from others because the prediction is not used across slice boundaries (Figure 2.5).

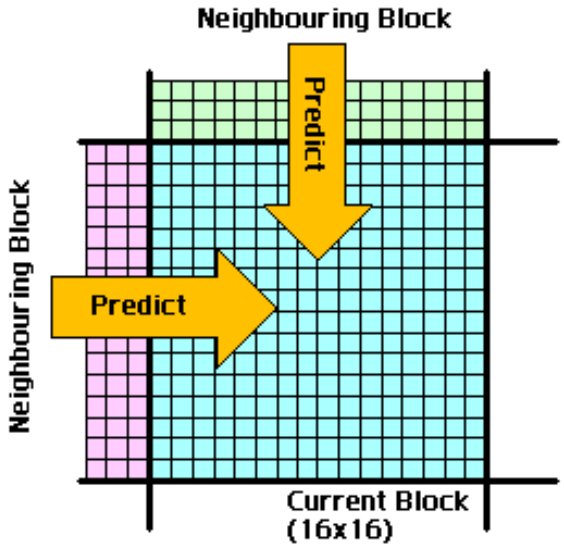


Figure 2.5: Intra Prediction

2.4.3 Inter-Frame prediction

Inter prediction uses previosly encoded video frames to create a prediction model [8]. Predictive or motion-compensated coding types are known as P macroblock types. They specify the subdivision of the macroblock into blocks of different number of samples. The partitions of luma macroblocks supported are of 16x16,16x8,8x16 and 8x8 samples. In case of choosing a partition of 8x8 sample, the standard allows an additional partition in blocks of 8x4, 4x8 or 4x4 and this information is also transmitted by adding some syntax elements. The partition of the 8x8 partitions is also avalaible for chroma samples.

The predicted macroblock is obtained by displacing an area of a reference picture. The dispacement is called motion vector and that is what is transmitted. The maximum number of motion vectors transmitted for a macroblock is 16, case of a macroblock divided into 4 8x8 partitions and each of them divided also into 4 4x4 partitions.

The values of luma and chroma samples are finite, and they represent the values of certain points of the pictures. Not only these values can be selected as a reference for motion compensation, but also the values in between. The acuracy of the system is a quarter of the distance in case of luma samples. In case the reference value corresponds to an integer-sample position, this value is copied to the predicted signal, but in other cases these values do not exist. Therefore they are calculated using interpolation. For values at half-sample positions, the prediction is obtained by applying one-dimensional 6-tap FIR filter horizontally and vertically. Values at quarter-sample positions are calculated by averaging samples at integer- and half-sample positions.

Bilinear interpolations are used to obtain the predicted values for the chroma component, and because of its lower resolution, the accuracy is of one-eight sample position.

This accuracity on motion prediction is one of the most important improvements

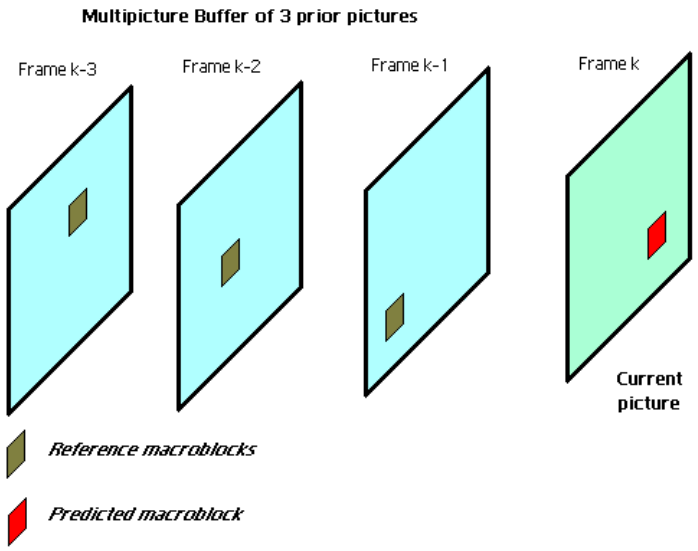


Figure 2.6: Inter Prediction

added by this standard since it allows a more precise motion representation and also a more flexible filtering in prediction.

In case the motion vector points outside the image area, the edge samples are taken as reference. However, samples from other slices can not be used as reference for motion-compensated prediction.

Once motion vectors of a picture are calculated, they are predicted using spatial prediction from neighboring motion vectors in a simmilar way intra macroblocks do. With the original motion vector and its prediction, a residual is calculated.

In H.264/AVC more than a picture can be used as a reference. That is known as multipicture motion-compensated prediction and requires a buffer to store all previous reference pictures in both, the encoder and the decoder (Figure 2.6). Every block from a macroblock transmits a reference index to indicate the position of the reference picture for this block in the multipicture buffer. Smaller regions than 8x8 share the same reference index.

Finally, another type called P-Skip is defined to represent large areas with no change or constant motion with very few bits. This type is similar to a P_16x16 macroblock type with the motion vector referred to the picture located at index 0 in the multipicture buffer.

2.4.4 Transform, quantization and entropy coding

After obtaining the residual information, it is transformed as in previous standards [9]. However, in H.264/AVC the transformation is done by separable integer values instead of a 4x4 distrete cosine transform (DCT). Since the transformation values are integer, no mismatches while decoding are introdoced. At the encoder the process includes a forward transform, zig-zag scanning, scaling, rounding and entropy coding. At the decoder the process is the inverse except for rounding. In chroma Intra_16x16 prediction mode, DC coefficients are transformed again as well as DC coefficients of the four 4x4 blocks of the chroma component. Cases of using a second transformation are utilized in very smooth areas because in that situation the reconstruction accuracy is proportional to the inverse of the one-dimensional size of the transform. But there are more reasons for using smaller size transforms. The first one is that as in H.264/AVC the predicted image is less correlated, transformation has less improvement to offer concerning decorrelation. Tranformation increase the noise in the edges of the image, and with smaller transforms it is reduced. Moreover the smaller the transform, the less comptations are required.

After transformation, the coefficients obtained are quantized. The quantization parameter used can take 52 values and the reduction of bitrate increases with its value. However, this relation is not linear. All quantized transform coefficients of a block are scanned in a zig-zag fashion but the 2x2 DC coefficients of the chroma components, which use a raster-scan order.

Before being transmitted, entropy coding is applied. Above the slice layer, syntax elements are encoded as fixed- or variable-length binary codes [10]. At the slice layer or below two entropy coding methods are supported by H.264/AVC. One of this methods consists of using a single infinite-extent codeword table for all syntax elements except the quantized transform coefficients. This table is an exp-Golomb code with

simple properties. The mapping to this table depends on data statistics. Quantized transform coefficients use the scheme called Context-Adaptive Variable Length Coding (CAVLC). In CAVLC, the different VLC tables are switched depending on the conditioned statistics, improving coding performance.

The other entropy coding method is the Context-Adaptive Binary Aithmetic Coding (CABAC). This method improves the efficiency because it uses adaptative codes to the statistics of the symbols. In this project, CAVLC is used. The whole coding process is shown in (Figure 2.7) [1].

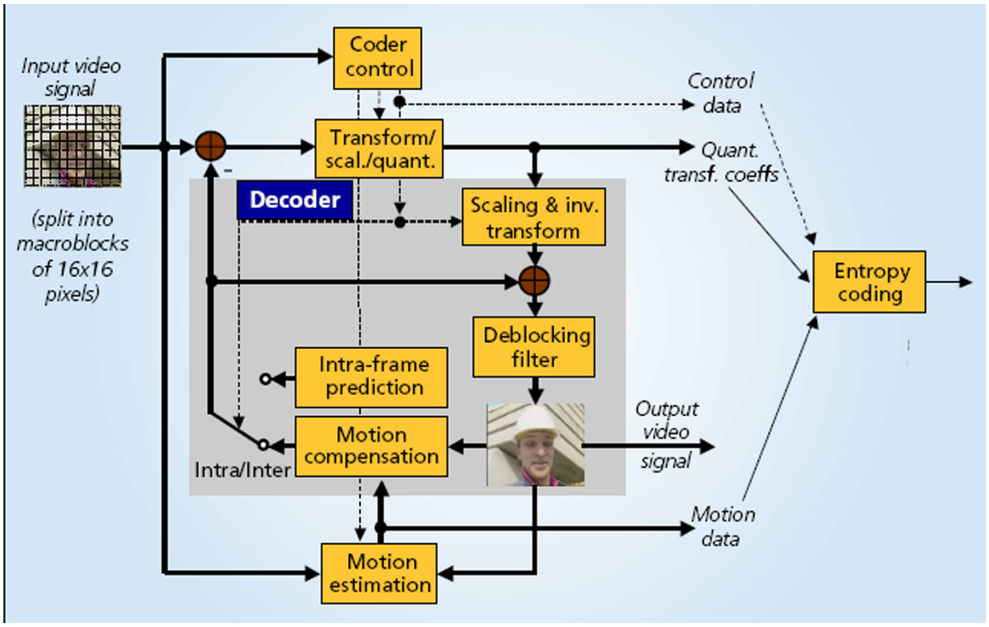


Figure 2.7: Basic encoding process for a macroblock

Chapter 3

SP and SI frames

3.1 Motivation

The new video coding standard, H.264/AVC, exploits in its extended profile the use of two new frame types, SI and SP. SP and SI picture coding was first introduced by Karczewics and Kurceren [11]. They have been created to try to avoid error propagation while decoding. Eventhough this was already possible in previous standards with the existing frame types and other techniques, they may fail to completely stop or effectively reduce error propagation and some of them may not be suitable for video streaming applications [12].

The most common frame type used and already existing in previous standard, is the P frame which exploits temporal prediction, that means that previous pictures are referenced to calculate the actual value. This method reduce significativaly the bitrate. However, in case of an error in transmision on a previously recieved picture, the decoded image with the not appropiate values will be stored in the decoder buffer and this frame will be an erroneous reference for next pictures. If only P frames are sent, this error will propagate without any stop, resulting a not desirable situation for the application users.

This error propagation can be solved by using I frames in H.264/AVC and also in previous versions. They exploit spatial prediction, that means that they depend on previously encoded macroblocks from the same frame, but not on previous frames. Thus in case of an error, the erroneous frames will not be referenced if the decoded frame is an I frame.

The reason why not all the frames sent are I frames is that its size is much bigger than the size of a P frame for a comparable quality as it is shown in Figure 3.1 and Figure 3.2. This is because the correlation among consecutive pictures is higher than the correlation among different parts of a same picture. These two pictures represents Foreman video with a QCIF resolution and size and quality are depicted for different values of the quantization parameter. The bigger the quantization parameter, the less bits are sent and the worst quality is obtained.

As a compromise some P frames are sent between two I frames, and the set of frames from an I frame up to the P frame preceding the next I frame is definded as Group of Pictures (GOP). The smaller the GOP, the more prepared is the transmision

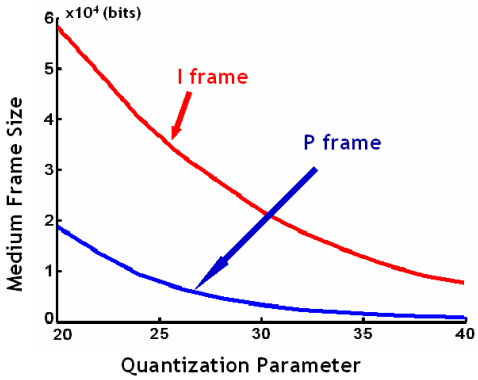


Figure 3.1: Comparison of size of I and P frames

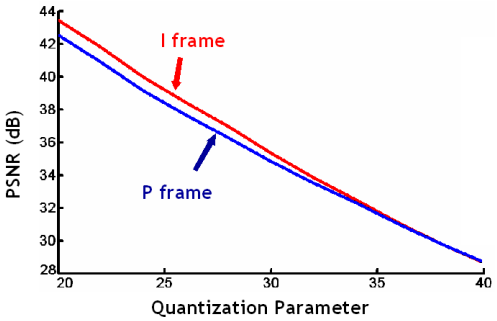


Figure 3.2: Comparison of quality of I and P frames

to an error in propagation but also the more bits are sent. In this project, a picture buffer of one frame has been used, what means the decoder uses only the previous frame for inter-prediction. Thus sending one I frame cuts the temporal referencing and it is equivalent to clearing the buffer.

The two new frame types of H.264/AVC standard, SI and SP, make also use of spatial and temporal prediction respectively. Thus SI frames are perfect to be applied to switch between different video sequence [13]. The difference with the previously commented method is that SP and SI are interchangeable, that means that both of them are able to reconstruct the same decoded picture. In case of a no error situation, SP frames are sent, and only if an error occurs, an SI frame is transmitted.

SI frames are able to stop error propagation as I frames do. No previously recieved pictures are referenced in the decoding process, only previously decoded macroblocks from the same frame because of using spatial prediction. This can lead to believe in the benefits of sending always SI frames. However, as it is depicted in Figure 3.3, its size is much bigger than the size of an SP frame. This is similar to the comparison of I and P frames using spatial and temporal prediction respectively. The main difference is that SP and SI frames obtain exactly the same reconstructed image in the decoding process as it has been mentioned before.

In that case it is possible to think about the idea of transmiting always SP frames. In that situation and if a notification of an error is received, an SI frame can be sent instantaneously. The reason is that an SI can only be transmitted instead of an SP

frame and never instead of a P frame. A look in the referenced figure shows that the size of an SP frame is even smaller than the size of a P frame. However, the graph that depicts the quality for the same encoded sequence (Figure 3.4) shows that the quality of the SP frame is smaller than the quality of the P frame.

As a consequence, some P frames are sent between two SP frames, reaching a compromise between the more quality offered by a P frame, and the possibility of recovering the good quality after an error offered by an SP frame. The number of frames between an SP frame and the last P frame before the next SP frame is called from now on in this project SP Period. A task of this project is to simulate different SP Periods to obtain an optimization of this value for several situations.

Since SP and SI pictures degrades the compression efficiency of the transmision even if there is no switching, another scheme has been proposed without using SP pictures and with a flexible switching points assignment [14]. This proposal is not studied further in this project.

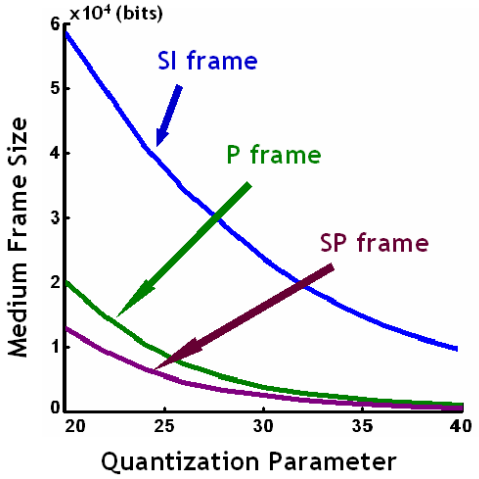


Figure 3.3: Comparison of size of SI, SP and P frames

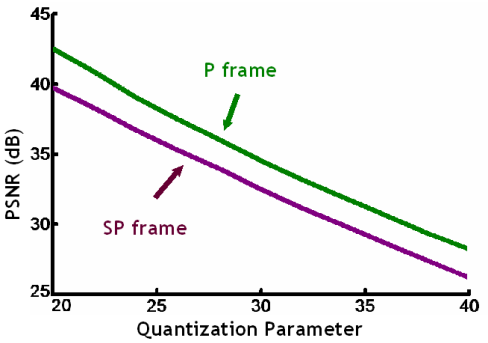


Figure 3.4: Comparison of quality of SP and P frames

3.2 Applications

The application explained before and used in this project is known as error recovery, but this is not the only for SP frames. A short resum of other uses of this frame type is explained in this section.

3.2.1 Bitstream switching

Network conditions vary continously and therefore, the effective bandwitdh avalaible for the users is not always the same. As a consequence the server should be able to modify the bitrate of the compressed video depending on these variations [15]. A possible solution is to control the quantization parameter or the frame rate in the encoding process based on the feedback information of the channel. However this method is only applicable in cases of conversational services, characterized by real-time encoding and point-to-point delivery.

In typical cases the bitstream to be sent is complitely encoded before being transmitted and in those cases there is a need to look for another solution. The solution offered is to encode the sequence several times with different bitrate and quality using different quantization parameters and frame rates resulting independent bitstreams. The server switches between the different bitstreams depending on the information of the bandwidth avalaible in the channel, adapting to it (Figure 3.5).

From now on, in this section a frame will have a subindex to identify to which bitstream it belongs, a 1 for the original bitstream and a two for the destination bitstream it will be a two. The timing information is also shown in the subindex after the bitstream.

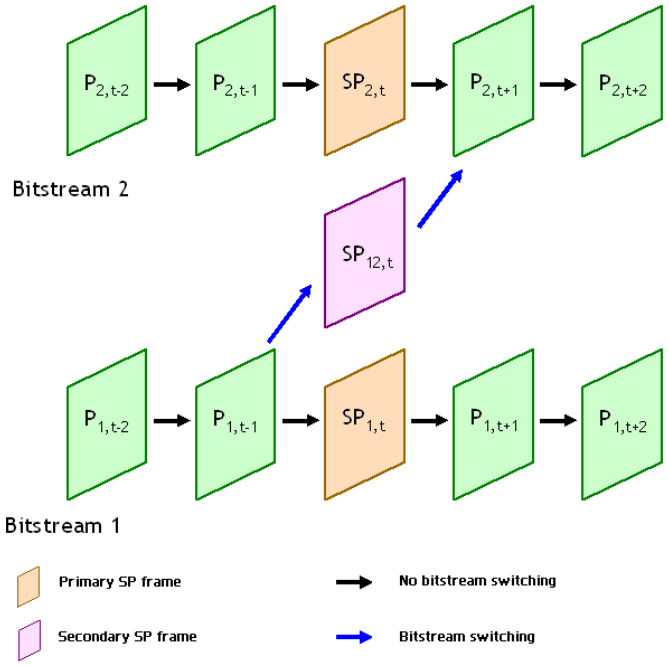


Figure 3.5: Switching bewtween bitstreams using SP frames

If the server switches between two bitstreams with different bitrates by sending a $P_{2,t+1}$ frame after a $P_{1,t}$ frame, there will be a mismatch in the decoding that will propagate in time due to motion-compensated coding. The reason is that $P_{1,t}$ is the proper reference for $P_{1,t+1}$ but is not a valid reference for $P_{2,t+1}$, wich suitable reference is $P_{2,t}$. As a consecuence the reconstructed image is not the picture expected in this time position if the second bitstream would have been transmitted from the begining and without being switched.

In prior video coding standards, a perfect switching without mismatches was obtained by sending an I frame from the destination bitstream. These frames do not depend on previous images and therefore the frames from the original bitstream are not used as a reference. The main drawback of this method is the bigger size of this frames regarding to P frames because of using spatial dependencies.

In H.264/AVC, SP frames can be used with this utility. Because of their special way of encoding, an identical reconstructed image is obtained by different SP frames using every of them different reference frames. In order to allow to switch from one bitstream to other, special SP frames, called Secondary SP frames, are encoded. They are able to reconstruct the picture of the destination bitstream from the references of the previous pictures from the original bitstream. The SP frames transmitted in case of no switching are known as Primary SP frames and they use the previous pictures from the same bitstream to predict the image.

3.2.2 Splicing and Random Access

In the previous subsection, the switching is implemented between bitstreams with different bitrate and quality but from the same sequence. It is also possible to need to switch between bitstreams from different sequences of images, for example between different commertials in TV broadcast. This process is known as splicing. The use of Secondary SP frames to switch to another sequence using references from the prior frames from the original sequence (completely independent to the destination sequence) is not so efficient. That is because temporal correlation between both bitstreams is low. In these cases, using spatial prediction is a better solution, that means implement the switching sending an SI frame from the second bitstream. The reconstructed image after this SI frame is the same as if the second bitstream would have been transmitted from its begining (Figure 3.6).

SI frames provides also random access. This application consists of accessing different temporal parts of the bitstream with an SI frame that reconstructs the image for the new temporal point accessed without referencing the previously transmitted frames.

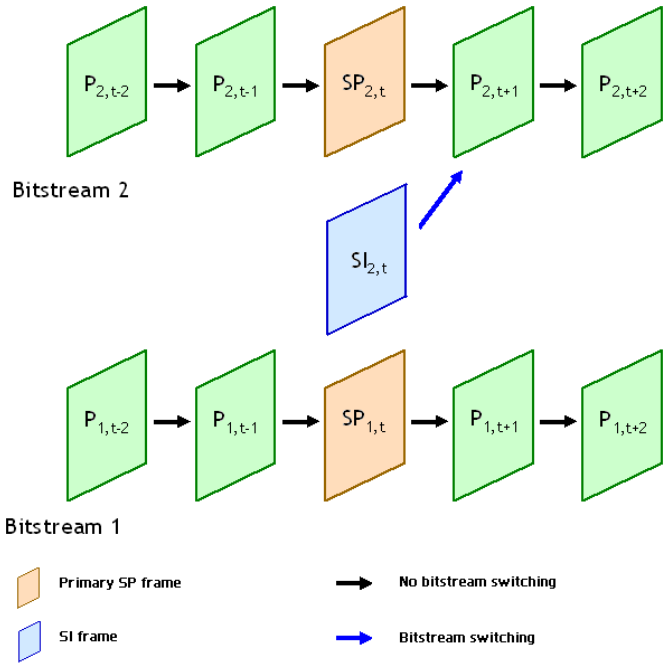


Figure 3.6: Splicing and random acccess using SI frames

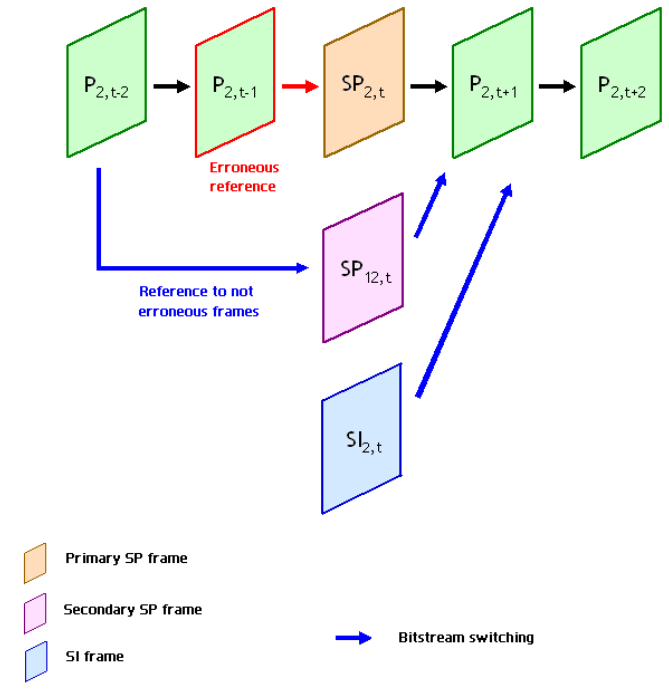


Figure 3.7: Error recovery

3.2.3 Error Recovery and Error Resilinecy

In error recovery strategy, the server is informed by a feedback channel about the loss of packets by the client. Multiple secondary representations has been encoded

before, and after this notification , instead of sending the next Primary SP frame, a Secondary SP frame with no reference to the erroneously recieved frames is sent. It is also possible to send an SI frame, without using any reference to previous frames to stop error propagation (Figure 3.7).

In error resiliency strategy the server has no information about the loss of packets or the channel conditions. In this case, either the sequence needs to be encoded with the worst-case expected network conditions or additional error resiliency/recovery mechanisms are required.

3.2.4 Video Redundancy Coding

This application consists of dividing the sequence into several threads, each of them encoded independently. In regular intervals, synchronization frames are inserted. These frames allow to start a new thread. In case of an error, the next synchronization frame will reference the no damaged threads. If this sync frames are P frames, there will be a mismatch since the reconstructed image is not the same. To solve this problem SP frames are used as synchronization frames.

3.3 The SP and SI design

The main feature of SP and SI frames is that both of them are able to reconstruct the same image. This characteristic allows them to be interchangeable without any mismatch.

In Figure 3.8 the same sequence has been encoded twice, one with SP frames inserted and the other with SI frames. The SP period is the same in both, as well as all the encoding parameters. It is possible to observe that the size is much bigger for an SI frame because of using spatial prediction instead of temporal prediction. However,

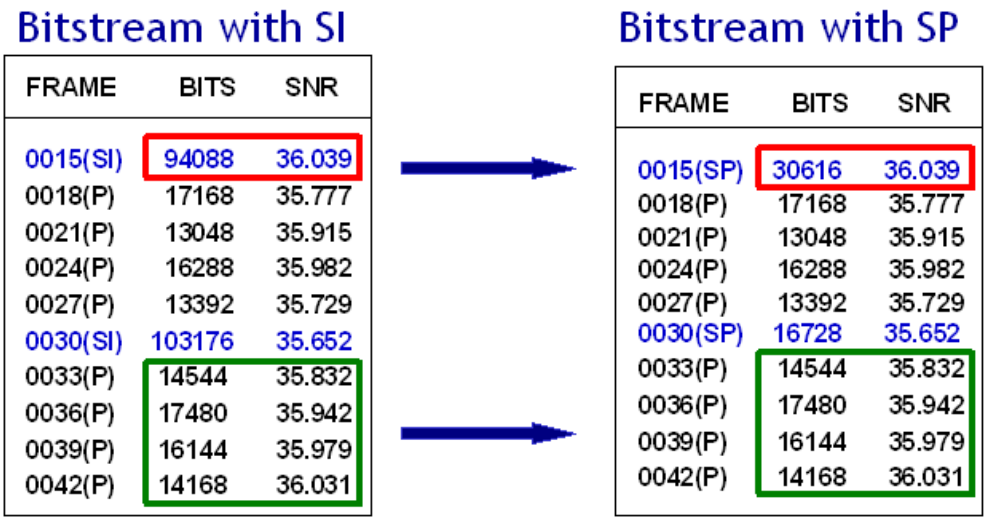


Figure 3.8: Comparison of a sequence encoded with SP/SI frames inserted

the quality of the reconstructed image is exactly the same as it has been mentioned before, and in the decoder, this image is put into the decoder buffer.

In this project, the decoding buffer has a size of one, and therefore, the P frame after an SP or an SI uses the same reconstructed image as a reference. That is the reason why the size and the quality of the P frames following an SP or an SI are exactly the same. These results are possible because of the special way of encoding and decoding of SP and SI frames and the whole process is explained below.

3.3.1 Encoding and decoding process in primary SP frames

The encoding and the decoding processes of a Primary SP block are different to those of an SI or a secondary SP block. All of them are described in the next subsections.

The encoding process of a Primary SP block is depicted in Figure 3.9. A predicted block is calculated from the previously reconstructed images stored in a buffer and the original block, using motion estimation. Both, the original block and the predicted block are transformed and the transformed coefficients of the predicted block are quantized and dequantized using SPQP as the quantization parameter. This step may seem to have no sense, but it is essential to reconstruct the same image than an SI or a Secondary SP frame.

The quantized and dequantized predicted coefficients are substracted from the transformed coefficients of the original image. The result of this substraction is quantized using the PQP parameter and it is transmitted together with the motion vector

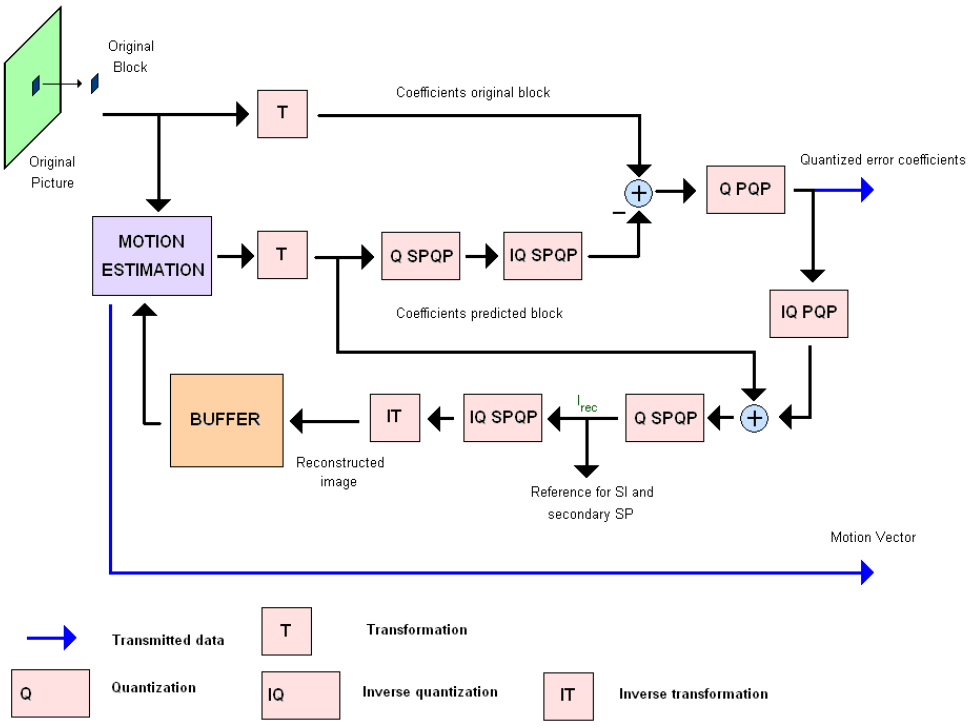


Figure 3.9: Encoding process for a block in a primary SP frame

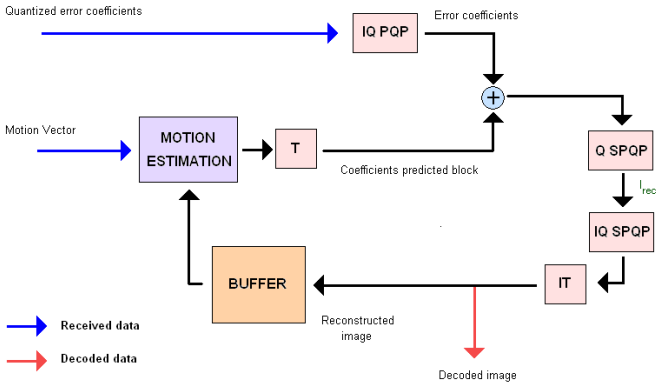


Figure 3.10: Decoding process for a block in a primary SP frame

information.

The quantized residual coefficients are used to reconstruct the image in the encoder. First this residual is dequantized and the predicted coefficients are added. This value is quantized by QPSP parameter obtaining lrec. Then a dequantization is applied in order to obtain the reconstructed blocks which are stored in a buffer to use them as a reference for next preditions. The importance of QPSP quantization and dequantization is the obtantion of lrec, which is used instead of the original block as a reference for SI and secondary SP frames. This is the main point that allows to reconstruct the same image for SP and SI frames. The quantized residual coefficients are decoded in the encoder following the same steps as in the decoder, to obtain the same reconstructed images.

In the decoder (Figure 3.10), the recieved error coefficients are dequantized by PQP parameter, the same used before the transmision. Moreover, a predicted block is formed and transformed. The coefficients obtained from both calculations are added, and the resulting values are known as the reconstruction coefficients. These coefficients are quantized and dequantized by SPQP parameter, following the same process as in the encoder, and therefore both have the same degradation. PQP and SPQP can have different values. SPQP affects to the predicted blocks and a finer value introduces smaller distortion and smaller reconstructed error.

3.3.2 Encoding and decoding process for secondary SP and SI frames

Figure 3.11 discribes the encoding process for secondary SP and SI frames. A predicted block is calculated from the previously reconstructed images stored in a buffer and the original block, using motion estimation in case of secondary SP frames. For SI frames, the spatial prediction is used, and therefore only already coded blocks from the same picture are utilized as a reference. This predicted block is transformed and quantized by SPQP parameter. These quantized coefficients of the predicted block are substracted to lrec, the reconstructed image after having been encoded as primary SP. The quantized error coefficients are transmitted together with the motion vector information in case of secondary SP frames or other information about the intra modes

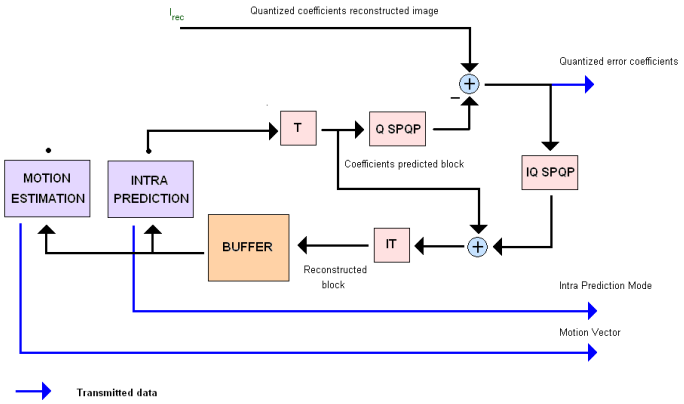


Figure 3.11: Encoding process for a block in secondary SP and SI frames

in case of SI frames.

In order to obtain the references for the next blocks, the error coefficients are dequantized by SPQP parameter and added to the coefficients of the predicted block, resulting the reconstucting coeficients. The inverse transform is applied to these coefficients and the reconstructed image is stored in a buffer to be used as a reference.

The decoding process is illustrated in Figure 3.12. A predicted block is formed by motion compensation from the already decoded frames for secondary SP frames. For SI frames, it is performed by spatial prediction from the already decoded neighbouring pixels within the same frame using the intra prediction mode information recieved from the encoder. The predicted block is transformed and quantized by SPQP parameter. The obtained quantized coefficients are added to the quantized error coefficients recieved to calculate the quantized reconstructed coeffcients. These coefficients are dequantized and inverse transformed is applied to obtain the reconstructed image. This image is also stored in a buffer to be used as a reference.

The use of the reconstructed image as a reference is the main point of this description since it makes the system independent from the prediction mode chosen and now identical frames are reconstructed even when different reference frames are used for their prediction. SI frames utilizes the own picture as a reference, not the previous; secondary SP uses the previous frames from the other bitstream; primary SP uses the previous frames from the same bitstream. With a different prediction mode, a different residual is obtained, but since the same prediction is performed in the decoder, the reconstructed images are identical.

Quantization is applied for both, the reconstructed image and the predicted image, before the substraction is calculated. For the reconstructed frame the reason is clear, the value can be obtained from its corresponding primary SP. For the predicted - image, the quantification error is exactly the same in the encoder and in the decoder. Moreover, in case of quantizing the residual, this process would add more quantification error to this value (quantification error for the reconstructed image and for the residual), and there would be a mismatch with primary SP frames. Transformation and quantization applied directly to the predicted blocks is necessary for identical reconstruction of SP and SI frames, but the coding efficiency decrease regarding to that of

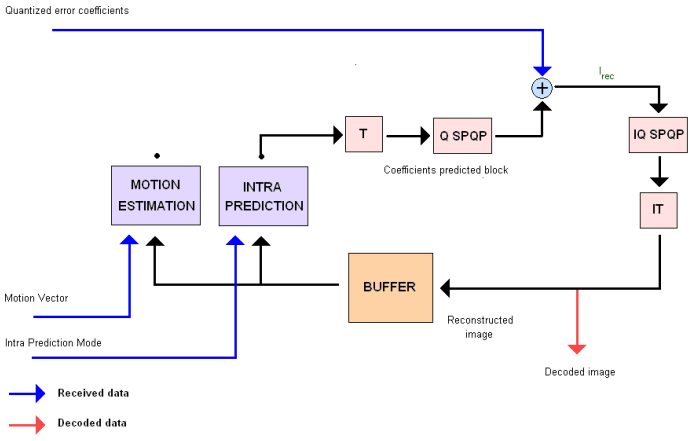


Figure 3.12: Decoding process for a block in secondary SP and SI frames

P or I frames.

As a conclusion, these frames are first encoded as primary SP in order to obtain the reconstructed image (lrec). Afterwards another encoding process is performed using the reconstructed image as a reference.

The effect of the different parameters is explained in another section, but a first contact with this process lead to think that PQP has an important effect in size and quality whereas SPQP can affect mainly the distortion of the frame.

3.4 Study of the influence of the different quantization parameters

As it has been explained before, the H.264/AVC standard introduces SI and SP frames with new encoding structures that utilize different quantization parameters. In order to see the influence of these parameters, simulations have been performed with two different videos, Mother and Daugther and Foreman, both of them with QCIF resolution (176x144). For every parameter value, two bitstreams are encoded, one with SP frames and the other with SI frames and the SP period used is 5. In this section only some results with the video Mother and Daughter are depicted, and the rest are shown in the Appendix B, since its results are similar or not so relevant.

Three different quantization parameters are used to quantize or dequantize data in the encoder and the decoder. In previous standards only one, known as QP parameter was used. In H.264/AVC it is still used to codify I and P frames. The other two quantization parameters are specific from SP and SI frames, and are known as PQP and SPQP parameters. PQP is only used directly in SP frames, but as an SI frame is first encoded as a SP frame, to use the reconstructed image as a reference, there is also some influence between its value and these frames. SPQP is used in both of them, affecting the predicted image and also its reconstruction in SP and SI frames.

3.4.1 Influence of QP parameter

To observe the influence of this parameter, some simulations have been performed fixing PQP and SPQP to 26 and varying QP value from 20 to 50.

The first two graphs (Figure 3.13 and Figure 3.14) show the size and the quality of P, SP and SI frames depending on QP parameter. It is possible to observe the high dependence that this parameter has on P frames. There is a logarithmic dependency (linear in quality because it is depicted in dBs and they are already logarithmic). The bigger is QP, the more quantization distortion is introduced and the more reduced is the picture in size. Consequently, the bitrate needed to transmit the picture decrease dramatically as well as the quality.

SP frames are indirectly influenced by QP parameter because they use the previous frame as a reference, a P frame. The higher the QP parameter, the worse quality the previous P frame has and therefore, the worst is the prediction of the SP frame. As a consequence, the residual is bigger and the size of the SP frame increase. However

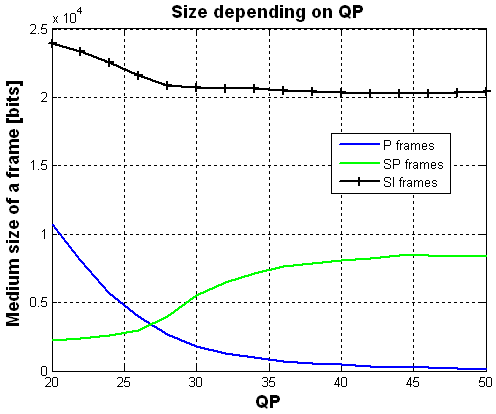


Figure 3.13: Comparison of size depending on QP

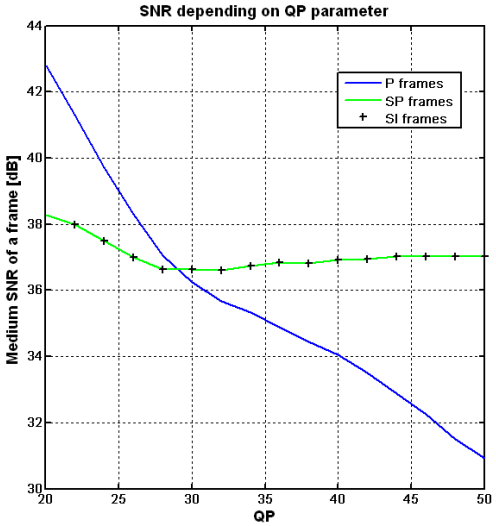


Figure 3.14: Comparison of SNR depending on QP

this parameter has no influence on the quality of the SP frame because it is not a quantization parameter of these frames, and thus they do not introduce quantization distortion. SI frames, since they do not use previous P frames as a reference for prediction, are not affected by QP parameter.

P frames use inter prediction, and therefore they utilize the previous frames as a reference. In our video application, only the previous frame is utilized because the buffer size is set to one. The behaviour of P frames has also been studied depending on whether they are just after an SP or an SI frame, and the rest of P frames (Figure 3.15 and Figure 3.16). In general terms, SP and SI frames are worse in quality than P and

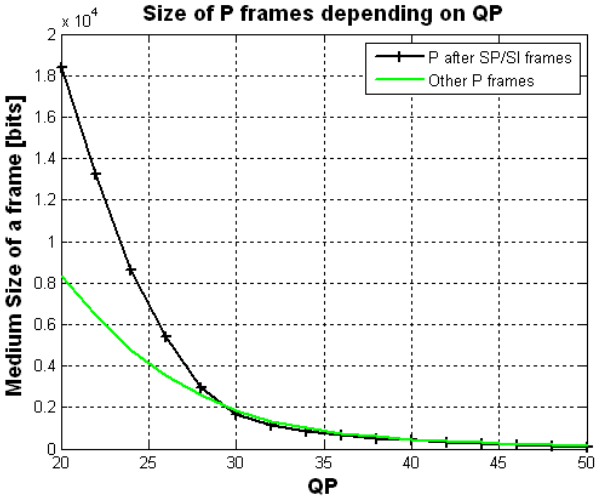


Figure 3.15: Comparison of size of P frames depending on QP

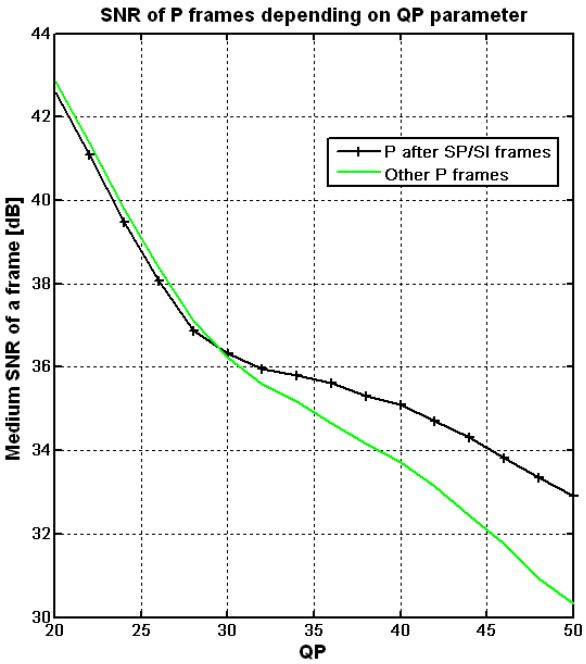


Figure 3.16: Comparison of SNR of P frames depending on QP

I frames. That is the reason of the bigger size of P frames just after SP frames for low QP parameters. Their reference is worse and therefore their prediction so more residiual is transmitted. However, the higher the QP, the more the quality of P frames decrease whereas the SP and SI quality is maintained stable. In these circumstances, SP frames are a better reference than P frames and the behaviour of the P frames after the SP or SI frames improve, outperforming the other P frames.

3.4.2 Influence of PQP parameter

Simulations have been performed fixing QP and SPQP parameters to 26 and varying PQP from 20 to 50. The two figures depicted next (Figure 3.17 and Figure 3.18) show the influence that this parameter has in the size and in the quality of SP, SI and P frames. The effect of this parameter is clear in SP frames. The residual obtained to be transmitted is quantized by this parameter and therefore the size of SP frames decrease logarithmically with PQP. The higher the PQP means also a bigger quantization distortion and a worse quality for the reconstructed image.

The reconstructed image is the same for SP and SI and thus SI quality also decreases with the increase of PQP. On the contrary, this parameter has no relevant influence on the size of SI frames.

The behaviour of P frames regarding this parameter is better shown in Figure 3.19 where P frames are divided between the frame just after an SP or an SI frame, and the rest. As it has been mentioned before, P frames uses only the last frame as a reference. P frames just after an SP or an SI frame use those as a reference whose quality decrease for high PQP. In those cases, the prediction is worse and consequently more bits are sent. However the other P frames use other P frames, with good quality even for high PQP, as a reference, resulting a proper prediction and a small residual. The size of these frames do not increase with high PQP.

On the other hand, this parameter has no influence on the quality of these two frames because they are not quantized by this value and therefore no quantization distortion is introduced.

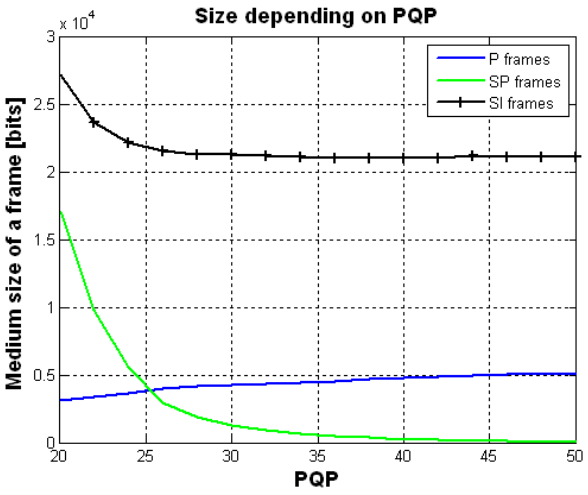


Figure 3.17: Comparison of size depending on PQP

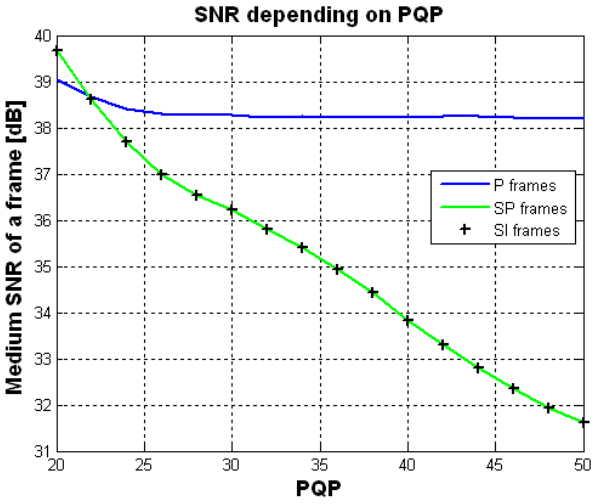


Figure 3.18: Comparison of SNR depending on PQP

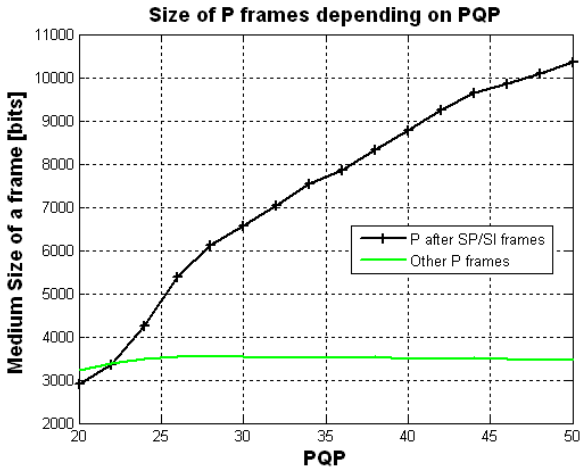


Figure 3.19: Comparison of size of P frames depending on PQP

3.4.3 Influence of SPQP parameter

Simulations has been performed fixing QP and PQP parameters to 26 and varying SPQP from 20 to 50.

The two figures depicted next (Figure 3.20 and Figure 3.21) show the influence of this value on the size and the quality of P, SP and SI frames. In SI frames, the residual is quantized by this parameter and thus the size of the transmitted data is reduced by increasing SPQP. Moreover, there will be more quantization distortion affecting the quality of the reconstructed picture. SPQP does not quantize the residual of SP frames, and that is the reason why their size do not depend on this parameter. However, in order to have identical reconstructed images in both, SP and SI frames, the decoded image is quantized and dequantized by SPQP. consequently, SP quality decrease when this parameter becomes higher.

The behaviour of P frames is better explained in Figure 3.22 where there is a

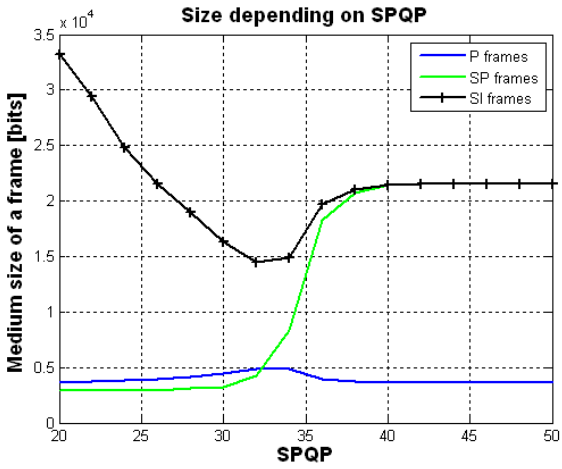


Figure 3.20: Comparison of size depending on SPQP (PQP=26)

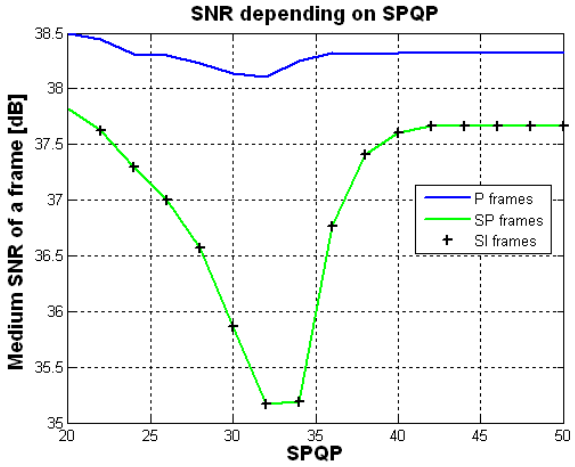


Figure 3.21: Comparison of size depending on SPQP (PQP=26)

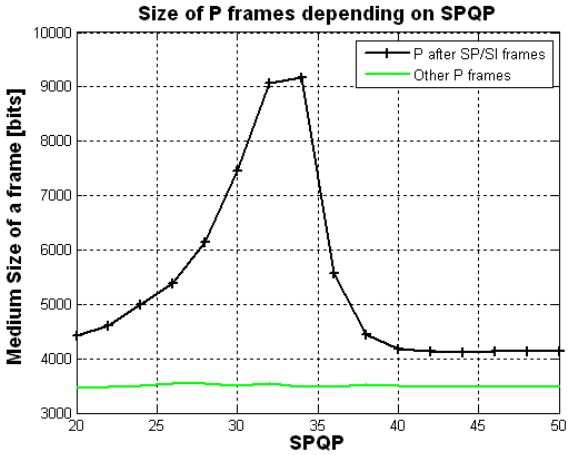


Figure 3.22: Comparison of size of P frames depending on SPQP (PQP=26)

distinction between P frames just after an SP or an SI frame and the rest of P frames. As in previous cases, P frames use only the previous picture as a reference. If quality of SP or SI frame decrease, the prediction of the P frame just after those, becomes worse. Consequently, the residual size increases and therefore the amount of transmitted data. However, its quality is maintained stable, since this parameter is not a quantization parameter for these frames. The other P frames that use other P frames with good quality as a reference do not depend on SPQP, neither in size nor in quality.

All these conclusions are true until a certain SPQP value. SI macroblocks can only be codified as I macroblocks whereas SP can be codified as P or I macroblocks. The decision of the type of prediction is made in the encoder in the function that encodes one macroblock [16], and the simplified algorithm is explained in Figure 3.23.

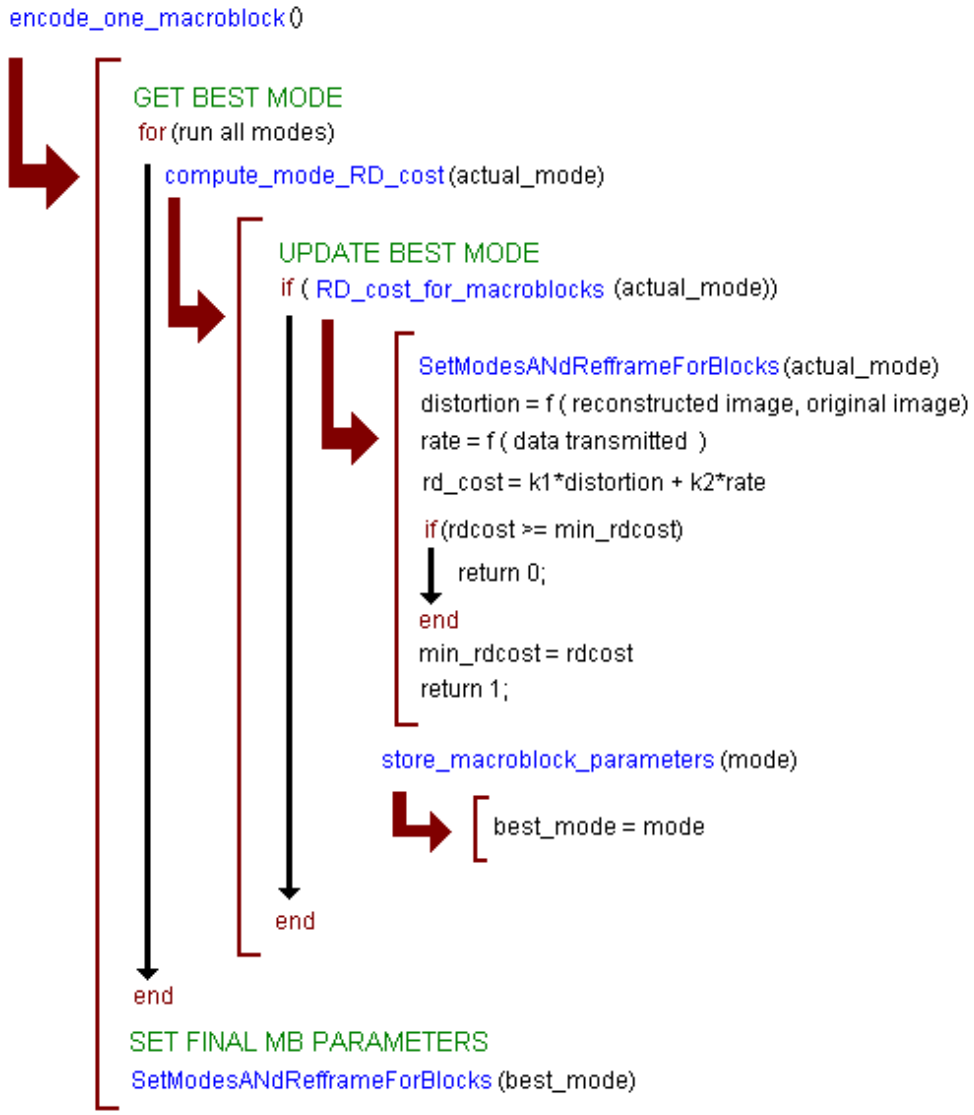


Figure 3.23: Algorithm to decide the encoding mode of a MB

The information about the type of macroblock has been extracted from the trace file of the encoder (Figure 3.24). The trace file explains deeply the different fields of the encoded bitstream. At the beginning of the information that describes every macroblock there is a section called *mb_type* with a number. If this number is smaller than five it is a P macroblock and if it is five or more it is a I macroblock. *mb_type*

```
@69     PPS: redundant_pic_cnt_present_flag                          0 (  0)
@70     SH: first_mb_in_slice                                        1 (  0)
@71     SH: slice_type                                         0001000 (  7)
@78     SH: pic_parameter_set_id                                     1 (  0)
@79     SH: frame_num                                             0000 (  0)
@83     SH: idr_pic_id                                               1 (  0)
@84     SH: pic_order_cnt_lsb                                0000000000 (  0)
@94     SH: no_output_of_prior_pics_flag                             0 (  0)
@95     SH: long_term_reference_flag                                 0 (  0)
@96     SH: slice_qp_delta                                           1 (  0)

************ Pic: 0 (I/P) MB: 0 Slice: 0 ************

************ Pic: 0 (I/P) MB: 0 Slice: 0 ************

@97     mb_type (I_SLICE) ( 0, 0) =   9                              1 (  0)
@98     Intra 4x4 mode  = predicted (context: 0)                     1 ( -1)
@99     Intra 4x4 mode  = predicted (context: 1)                     1 ( -1)
@100    Intra 4x4 mode  = predicted (context: 2)                     1 ( -1)
@101    Intra 4x4 mode  =   0 (context: 3)                        0000 (  0)
@105    Intra 4x4 mode  =   1 (context: 4)                        0001 (  1)
@109    Intra 4x4 mode  =   1 (context: 5)                        0001 (  1)
@113    Intra 4x4 mode  = predicted (context: 6)                     1 ( -1)
@114    Intra 4x4 mode  = predicted (context: 7)                     1 ( -1)
@115    Intra 4x4 mode  = predicted (context: 8)                     1 ( -1)
@116    Intra 4x4 mode  = predicted (context: 9)                     1 ( -1)
@117    Intra 4x4 mode  = predicted (context: 10)                    1 ( -1)
@118    Intra 4x4 mode  = predicted (context: 11)                    1 ( -1)
@119    Intra 4x4 mode  = predicted (context: 12)                    1 ( -1)
@120    Intra 4x4 mode  =   7 (context: 13)                       0111 (  7)
@124    Intra 4x4 mode  = predicted (context: 14)                    1 ( -1)
@125    Intra 4x4 mode  = predicted (context: 15)                    1 ( -1)
@126    Chroma intra pred mode                                       1 (  0)
```

Figure 3.24: Extract of the trace file.

mb_type	Name of mb_type	NumMbPart (mb_type)	MbPartPredMode (mb_type, 0)	MbPartPredMode (mb_type, 1)	MbPartWidth (mb_type)	MbPartHeight (mb_type)
0	P_L0_16x16	1	Pred_L0	na	16	16
1	P_L0_L0_16x8	2	Pred_L0	Pred_L0	16	8
2	P_L0_L0_8x16	2	Pred_L0	Pred_L0	8	16
3	P_8x8	4	na	na	8	8
4	P_8x8ref0	4	na	na	8	8
inferred	P_Skip	1	Pred_L0	na	16	16

Figure 3.25: P macroblocks types.

equal to five corresponds to I4x4 macroblocks and greater to 5 corresponds to I16x16.

The types of macroblocks are described in Figure 3.25 and Figure 3.26 [16]. I macroblocks come after P macroblocks and therefore 4 must be added to the *mb_type* number in Figure 3.26.

When SPQP increases, the distortion of P macroblocks in SP becomes so high that the encoder finds better to encode using intra prediction in blocks known as I16x16. Consequently, the size increase as well as the quality of the reconstructed image.

I macroblocks in SI frames also change from I4x4 to I16x16. This bigger blocks are more difficult to be predicted using spatial prediction since spatial correlation

mb_type	Name of mb_type	MbPartPredMode (mb_type, 0)	Intra16x16PredMode	CodedBlockPatternChroma	CodedBlockPatternLuma
0	I_4x4	Intra_4x4	na	na	na
1	I_16x16_0_0_0	Intra_16x16	0	0	0
2	I_16x16_1_0_0	Intra_16x16	1	0	0
3	I_16x16_2_0_0	Intra_16x16	2	0	0
4	I_16x16_3_0_0	Intra_16x16	3	0	0
5	I_16x16_0_1_0	Intra_16x16	0	1	0
6	I_16x16_1_1_0	Intra_16x16	1	1	0
7	I_16x16_2_1_0	Intra_16x16	2	1	0
8	I_16x16_3_1_0	Intra_16x16	3	1	0
9	I_16x16_0_2_0	Intra_16x16	0	2	0
10	I_16x16_1_2_0	Intra_16x16	1	2	0
11	I_16x16_2_2_0	Intra_16x16	2	2	0
12	I_16x16_3_2_0	Intra_16x16	3	2	0
13	I_16x16_0_0_1	Intra_16x16	0	0	15
14	I_16x16_1_0_1	Intra_16x16	1	0	15
15	I_16x16_2_0_1	Intra_16x16	2	0	15
16	I_16x16_3_0_1	Intra_16x16	3	0	15
17	I_16x16_0_1_1	Intra_16x16	0	1	15
18	I_16x16_1_1_1	Intra_16x16	1	1	15
19	I_16x16_2_1_1	Intra_16x16	2	1	15
20	I_16x16_3_1_1	Intra_16x16	3	1	15
21	I_16x16_0_2_1	Intra_16x16	0	2	15
22	I_16x16_1_2_1	Intra_16x16	1	2	15
23	I_16x16_2_2_1	Intra_16x16	2	2	15
24	I_16x16_3_2_1	Intra_16x16	3	2	15
25	I_PCM	na	na	na	na

Figure 3.26: I macroblocks types.

decreases between big portions of a picture. The residual becomes bigger and more bits are transmitted. The quality of the reconstructed image improve because the same is used for both, SP and SI frames. For high SPQP values, both SP and SI frames are formed by I16x16 blocks, and thus they become very similar. That is the reason why SP and SI size converge in these cases.

The effect of the improvement in the quality of the reconstructed image of an SP or an SI frame affects clearly the first P frame just after them. A better reconstructed image means a better reference, a better prediction and a smaller residual. Therefore, less bits are transmitted.

In the graphs shown below (Figure 3.27 and Figure 3.28), it is possible to observe that the SPQP value where the two types of macroblocks cross, coincide with the SPQP value where a peak in size and quality was observed, the point where the behaviour change.

The value of the SPQP parameter from which P macroblocks in SP frames become I macroblocks depends on the value of PQP. In previous simulations PQP parameter was

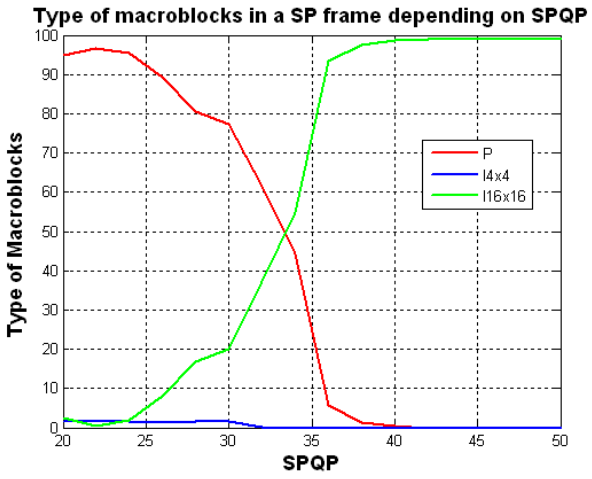


Figure 3.27: Type of macroblocks in an SP frame

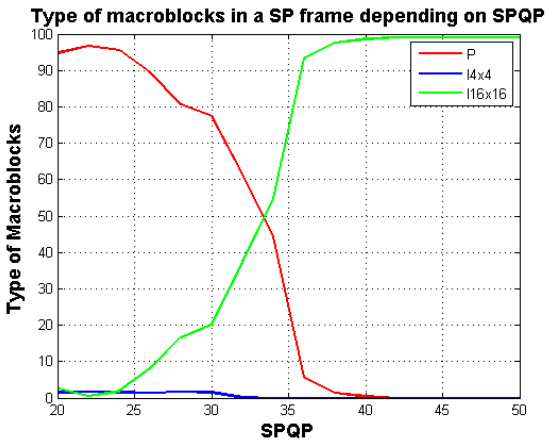


Figure 3.28: Type of macroblocks in an SI frame

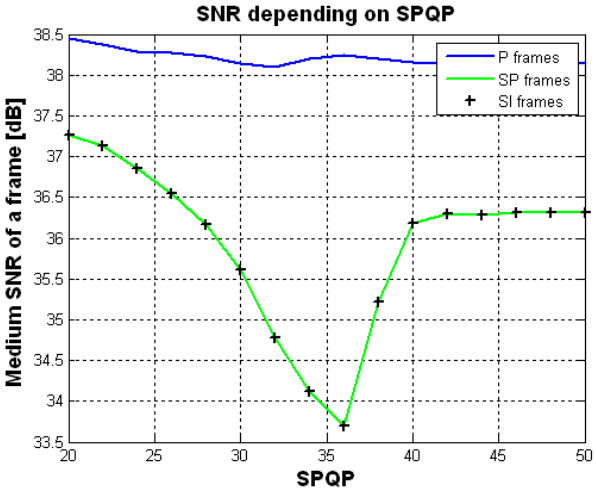


Figure 3.29: Comparison of SNR depending on SPQP (PQP=28)

fixed to 26 and in the figure depicted below a PQP of 28 has been used (Figure 3.29). QP value is fixed to 26 and SPQP parameter varies from 20 to 50. The peak in size has displaced from SPQP 34 to 36. At this point there are more I macroblocks than P macroblock, improving the behaviour in quality and increasing the bitrate.

A higher PQP value means more quantization distortion for the reconstructed image in SP and SI, and that means a worse quality. In those cases, the effect of the distortion introduced by SPQP is in some way concealed until a higher value of this parameter.

QP parameter does not displace the SPQP value where P macroblocks becomes I macroblocks, that produces this peak in quality.

3.5 Description of the optimized values for SPQP and PQP

In the previous section, the effect in rate and distortion of the different quantization parameters have been described. General conclusions are that PQP influence size of SP and quality of SP and SI frames whereas SPQP afects size of SI and quality of SP and SI frames. Figure 7.3 [17] depicts how the increase of SPQP parameter decrease the size of SI frames and increase the distortion of SP frames.

These conclusions explains the behaviour, however optimized values were seek for this project. The choice of the values of these parameters have followed the recommendation described in [18].

In the considered scenario, SP frame positions are spaced regularly in the transmitted video stream and SI frames can be sent in these positions in case of packet losses to stop error propagation. The proportion of SI frames transmitted instead of SP frames depends on the packet error rate and on the SP period. The probability of transmitting an SI frame at an SP frame position is denoted as x. In this optimization, the quality is considered as a constraint and the bit-rate is minimized what is equivalent to minimizing the expected size of a frame sent at an SP position:

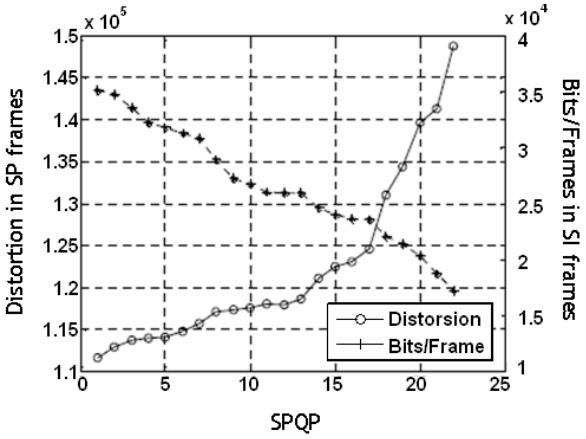


Figure 3.30: Distortion of SP frames and size of SI frames depending on SPQP

$$\text{Min. } \mathcal{R} = xR_{SI} + (1-x)R_{SP_1} \tag{3.1}$$

$$\text{s.t. } D_{SP_1} = D_{SI} = D \tag{3.2}$$

where R_{SI}, R_{SP_1}, D_{SI} and D_{SP_1} denotes the rate and distortion of SI and primary SP frames respectively.

The optimization is carried out by minimizing the bitrate of the transmission for a given distortion, that means that a minimum value of quality must be exceeded. This optimization method is described with more detail in Appendix A, and all the mathematical explanations are written in [18].

This method concludes that the only possible optimum values are those shown in Table 7.1 , as well as the optimal values derived empirically. Empirical values have been used in these project.

x model	≤ 0.2	≥ 0.2 and ≤ 0.5	≥ 0.5
x empirical	≤ 0.1	≥ 0.1 and ≤ 0.2	≥ 0.2
PQP	QP - 1	QP - 2	QP - 3
$SPQP$	QP - 10	QP - 5	QP

Table 3.1: Optimal values for PQP and SPQP

Chapter 4

Modifications in the encoder

This project is thought for mobile applications and it has been decided to work with bitstreams encoded in RTP packets, every one forced to a maximum of 700 bytes. H.264/AVC allows this encapsulation, however it introduces some problems in that case while working with SP and SI frames.

The main feature of SI and SP frames is that both of them reconstruct identical images and therefore they are interchageable with no mismatches. This is true when the RTP packets are as big as the size of the whole frame, that means every packet corresponds to one frame. When it is set in the configuration file of the encoder that every packet must have as maximum 700 bytes, some frames, specially I and SI frames, which are bigger, are divided in several RTP packets.

The actual version of the encoder is not prepared to divide SI and SP frames into several packets and mantain the identical reconstruction. This is depicted in Figure 4.1. All the I and P macroblocks before the first SP/SI frame are equal in both bitstreams, in size and quality. The problem begin in the SP and the SI frame. It is

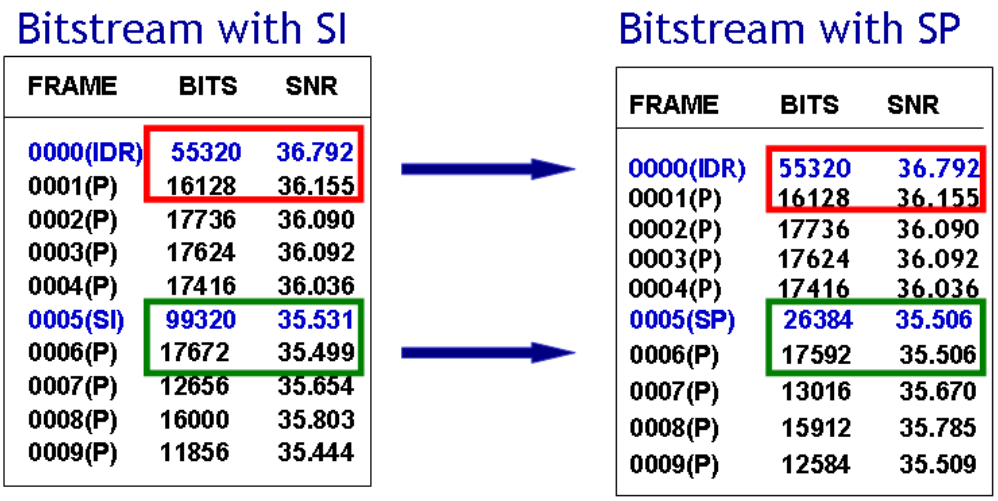


Figure 4.1: Encoding of the same sequence (1 RTP packet = 700 bytes)

normal that every of them have a different size because spatial prediction used in SI frames is not so accurate as inter prediction and therefore the number of residual sent is bigger. However the quality of the reconstructed image is also different. Consequently, the next P frames use different references for the prediction and that is the reason why they do not coincide in size and quality.

At first sight the diffence between the two bitstreams is very little. However, after applying the switching program depending on the errors in transmision (explained later) the results obtained are far away the optimal. Figure 4.2 is a picture that shows the SNR of the encoded bitstream with SI frames and the decoded one. The SP period is 5. In frame number 8 there is an error in transmision and the quality decreases dramatically 10 dBs. As soon as the routines that implement the swithing process detects the error and the switching is possible, in frame number 20, an SI frame is sent. Consequently, the good quality is recovered. However, since the P frame after the SI is encoded using the SP frame as a reference for prediction, and in the decoder, the SI is its reference instead of the SP, the degradation in the quality of the reconstructed image corresponding to this P frame increases. From this point, every frame will be reconstructed using a not appropiate reference for prediction and therefore degradation is propageted and increases with the time. In this example, in frame number 45, the quality of the decoded bitstream is 2 dBs lower than the quality of the encoded bitstream.

This decreases the performance of this method and makes it not interesting for mobile applications. In order to avoid this behaviour, the encoder has been modified.

Figure 4.2: Comparison of the quality of the encoded and the decoded bitstream

The idea is to send I and P macroblocks divided into blocks of 700 bytes, but to send SP and SI frames as an only packet. This has been considered a proper solution to obtain realistic results of the behaviour of the method because of how the errors are simulated in the switching program, but this is explained in the next chapter.

A simplified algorithm of the encoder is depicted in Figure 4.3. The changes performed on it are remarked with an orange square. The slice mode is previously arrenged in the configuration file, where it is set:

SliceMode = 2 # Slice mode (0=off 1=fixed #mb in slice 2=fixed
# # bytes in slice 3=use callback)
SliceArgument = 700 # Slice argument (Arguments to modes 1 and 2 above)

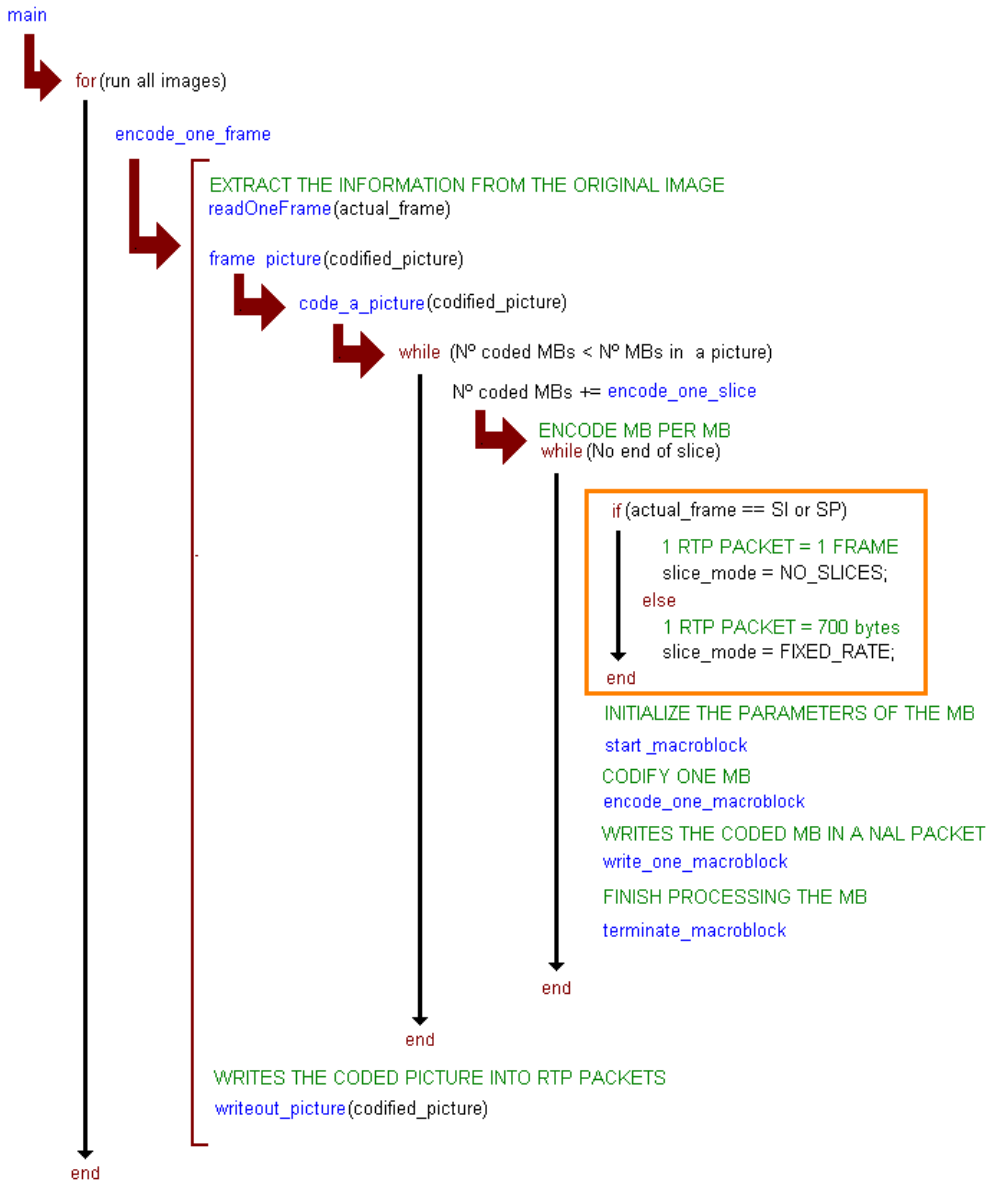


Figure 4.3: Modification of the encoder

So the *slice_mode* is set to 2, what means that the RTP packets have a fixed number of bytes and equal to 700. The changes are located in the function that encodes one macroblock, before its parameters are initialized, since this initialization depends on the *slice_mode*. If the macroblock belongs to an SI/SP the *slice_mode* changes to NO_SLICE, what means that the whole frame is encapsulated in an only RTP packet.

With this method SI and SP reconstruct exactly the same image, and consequently the next P frames use the same picture as a reference and are identical in the two bitstreams. In Figure 4.4 it is shown the bitstream encoded with SI inserted and the bitstream decoded with errors. The errors are in the same positions as in Figure 4.2, however, after the SI, the decoded bitstream recovers the good quality not only in this frame, but also in the next P frames, just until there is another error in frame number 52.

All simulations discribed in chapter 3 have been performed with the encoder already modified.

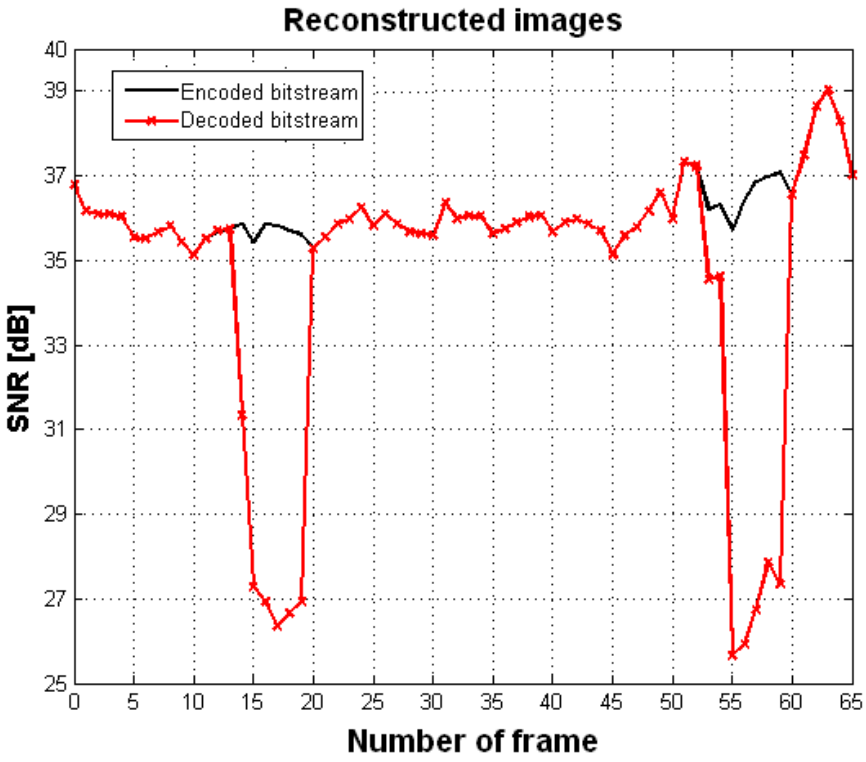


Figure 4.4: Quality of the encoded and the decoded bitstream after the modification

Chapter 5

Implementation of the switching process

5.1 General scheme

The aim of the project is to implement a switching process between the two new frames that H.264/AVC standard includes regarding to prior standards, SP and SI frames. Depending on the error information received from the decoder, it is decided to send an SI frame, formed using spatial prediction. The no abscence of previous frames helps to stop the error propagation while decoding.

The H.264/AVC encoder is able to produce a sequence with some SP or SI frames within the bitstream in periodical locations. However, the encoder does not decide whether to encode an SP or an SI depending on the error information recieved, that means that the switching process is not implemented inside the encoder. A general scheme is shown in Figure 5.1, where the switching program is depicted between the encoder and the decoder.

The switching process is implemented in a Matlab routine with some inputs and an output (Figure 5.2). The encoder codifies the sequence twice, one with SP frames and the other with SI frames inside. Both bitstreams must have the same periodicity for the switching frames and the same encoding properties, for instance the quantization parameters (QP, PQP and SPQP). An example of the bitstream with SP frames inserted with an SP periodicity of 4 is shown in Figure 5.3. The two bitstreams begin with an I frame. All frames from one bitstream reconstruct identical images to the corresponding frames in the other bitstream. Also the size is the same for I and P frames to the corresponding frames from the other bitstream. These bitstreams are

Figure 5.1: Diagram of the location of the switching process

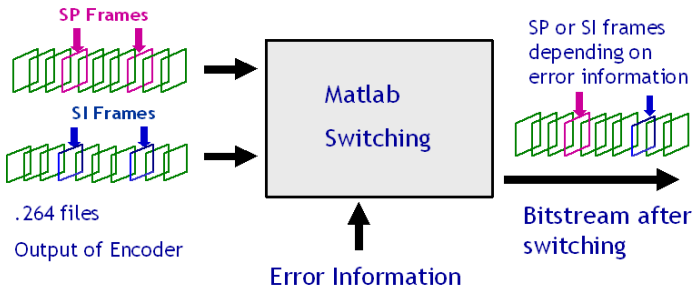


Figure 5.2: Bitstreams with SP and SI frames inserted.

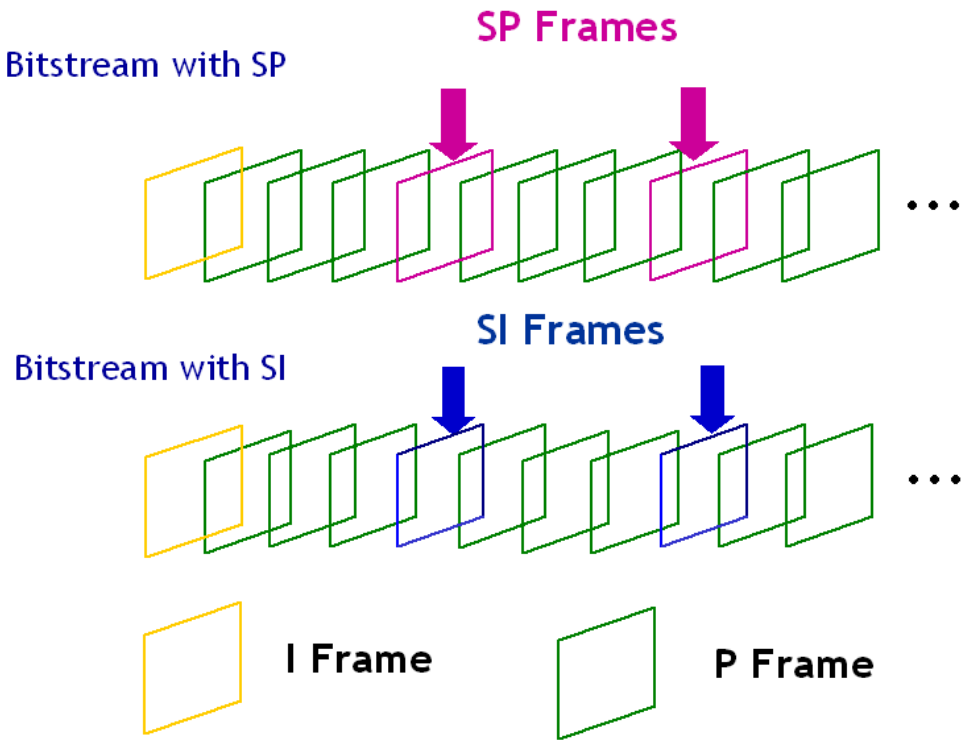


Figure 5.3: Inputs and outputs of the switching routine.

two of the inputs of the Matlab routine.

The third input is the error information that points out when the switching must be implemented. The output is one bitstream with SP and SI frames inserted periodically, depending on the error information. This output is transmitted and is what the decoder recieves in case of no error in transmision.

5.1.1 UMTS. General transmision scheme

This project is based on video applications transmitted over UMTS networks. Universal Mobile Telecomunications system (UMTS) is a technology used for mobiles of third generation, succeed of GSM. It is nowadays being developed into a fourth gene-

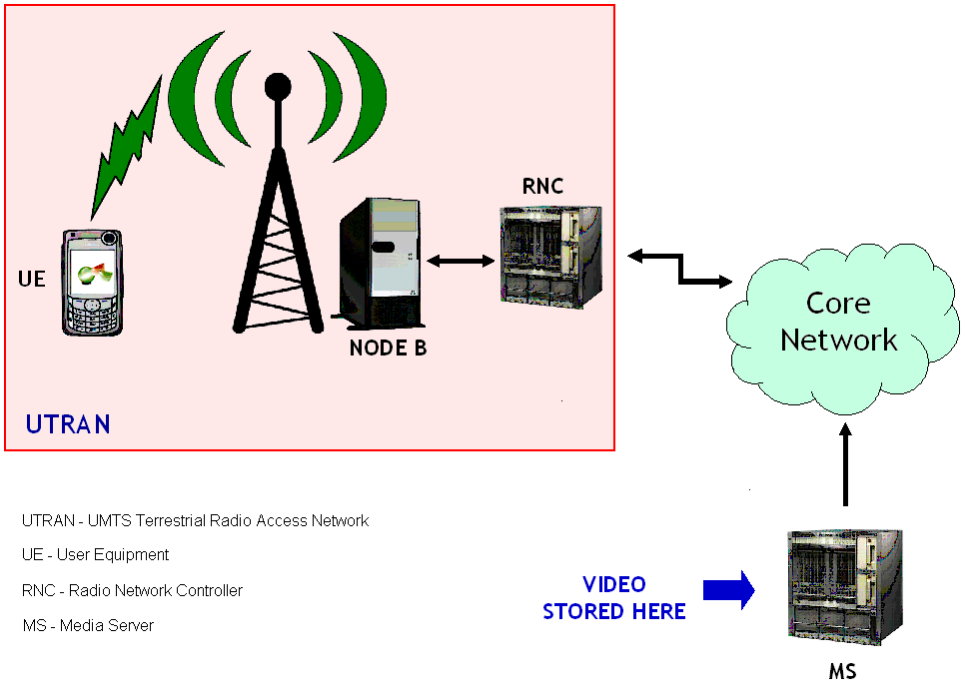


Figure 5.4: Simplified scheme of the UMTS network

ration technology. UMTS normally works over a WCDMA interface and over GSM infraestructures.

UMTS is not limited to mobile applications. Its three main features are its multimedia capabilities, a high internet access speed and the possibility of real time video. The voice quality is comparable to that of the fixed line telephone and it provides a wide variety of services like videoconferencing and live TV. Figure 5.4 depicts a simplified scheme of a UMTS network.

The UMTS network structure consits of two big subnetworks, a telecomunication network and a management network. The first one deals with the support of the transmition of information between the extrems of a conection. The second one controls the registration, definition and security of the service profiles provided as well as it has the misison of detecting and solving posible breackdowns and anomalies in the net.

The telecomunication network have the following two elements:

- **Core Network:** It includes transport and inteligence functions. The transport of the information and the signaling data and its conmutation is supported by transport functions whereas routing is considered as an intenligence function. Throughout the core network, UMTS is conected to other networks to make possible the communication not only with users from UMTS.

- **UMTS Terrestrial Radio Access Network (UTRAN):** Developed to obtain high velocities in transmition, conects mobile stations with the core network. It is formed by a series of Radio Network Subsystems (RNS), which are respon-

sible of the resources, the transmision and the reception in a set of cells.

The RNS is made up in turn of one Radio Network Controller (RNC) and one or several NodeB. Node Bs are also referred to as Base Transceiver Stations (BTS) or Base station in GSM. It has a radio frequency trasnmitter and reciever to communicate directly with the mobiles of its corresponding cell.

The RNC is responsible of controlling the UTRAN and therefore the NodeB of one or more Base Transceiver Stations (BTS).BTS are controlled by the RNC and therefore it has not a lot of functionalities. Some of the tasks that the RNC present are:

- **Admission control**
- **Packet scheduling**
- **Security functions**
- **Handover control**

- **User Equipment (UE)**: This is the new terminology in 3G of what is was known as Mobile Station in 2G. It is made up of a mobile terminal, that is the end-point of the radio interface, and a USIM (Universal Subscriber Identity Module). The USIM is roughly the SIM of GSM, and it stores information of the user for its identification in the network as well as other informations as mobile number, contacts and short messages.

In UMTS networks, the videos are stored in the Media Server. The Media Server is defined as a device that stores and shares media. The different types of media (video, photos, audio) are stored on its hard drive. With an special application it is possible to make this information accesible to users from a remote location via internet. Because of its vague definition, a media server only requieres a method of storing media and a network connection. Thus several devices can be considered Media Server, as a media center PC, or a commercial web server that hosts media. Depending on the application, a powerful processor, big RAMs for large storage or high access speed may be needed. Some Media Servers are able to capture the media using specialized hardware as TV turner cards. In case of analog video, it has to be encoded in digital format before being stored on it.

Video applications are transmitted over UDP. UDP is a transport level protocol, very useful for real time applications because it does not waste resources in controlling the parameters of the link, like flow control. Consequently the different packets can be received out of order. A previous conection before trasnmitting is not necessary because a UDP datagram incorporates enough address information on its header to allow the end-to-end communication. Since it does not make error control, there is no information of whether a packet has been recieved correctly or not, and therefore no retransmission is possible. This is the main difference with the other very important transport protocol, TCP, which is reliable. However, for real time applications, retransmissions are not allowed because of the little delays required.

5.1.2 Location of the Switching program in the UMTS network

Depending on several features of the network, different locations within the UMTS network are more interesting for the switching program. In this project two are explained:

- **Media Server:**

  This is the device where the videos are stored. The main advantage of this location is that both bitstreams, one with SP and the other with SI frames inserted, must not be transmitted, only the result of the switching process, saving bandwidth in the transmission. To make the switching possible, information about how the packets have been recieved is needed. As it has been explained before, UDP does not inform about errors in transmission and therefore a higher level is used to know whether the switching must be implemented or not.

  RTP is the higher protocol over which the UDP datagrams are transmitted (Figure 5.5). RTP is utilized together with RTCP to provide information about the quality of data distribution. The video is sent from the Media server to the client in RTP packets, transparent to other devices of the network because it is an end-to-end protocol. RTCP is used to monitor the quality of service, providing out-of-band control information. RTCP packets do not transmit data, but contains stadistics that reports over the number of packets sent, the number of packets lost, the interarrivel jitter, etc. RTCP packets are sent over an

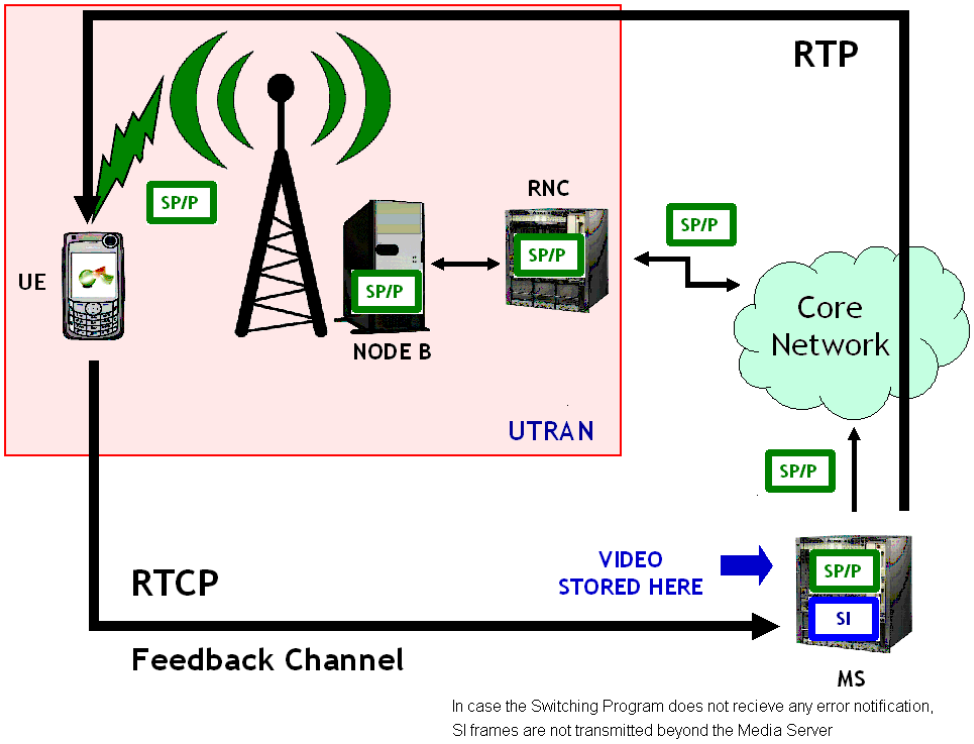


Figure 5.5: Switching program located in the Media Server

independent feedback channel.

The main drawback of this location is the delay on the reporting messages introduced by the feedback channel. This delay prevents the sender from adapting to the real state of the session. In our application, the switching could be implemented a long time after the error has started propagating, causing a great degradation of the quality in the decoded video.

- **Radio Network Controller:**

  The RNC is the device that controls the UTRAN. To make possible the implementation of the switching in the RNC, both bitstreams, the one with SI frames inserted and the other with SP, must be transmitted until this point of the UMTS Network (Figure 5.6). This suppose an increase in the bandwidth needed to transmit the video information, not only because the P frames have to be transmitted twice, but because the SI frames are much bigger than P and SP frames.

  The main advantage is that the RNC is closer to the Mobile Station than the Media Server and therefore the time needed to transmit some information from the Mobile Station is smaller. The RNC does not understand the transport protocol (RTP/RTCP), and therefore the error information must be transmitted in the link layer or a lower one.

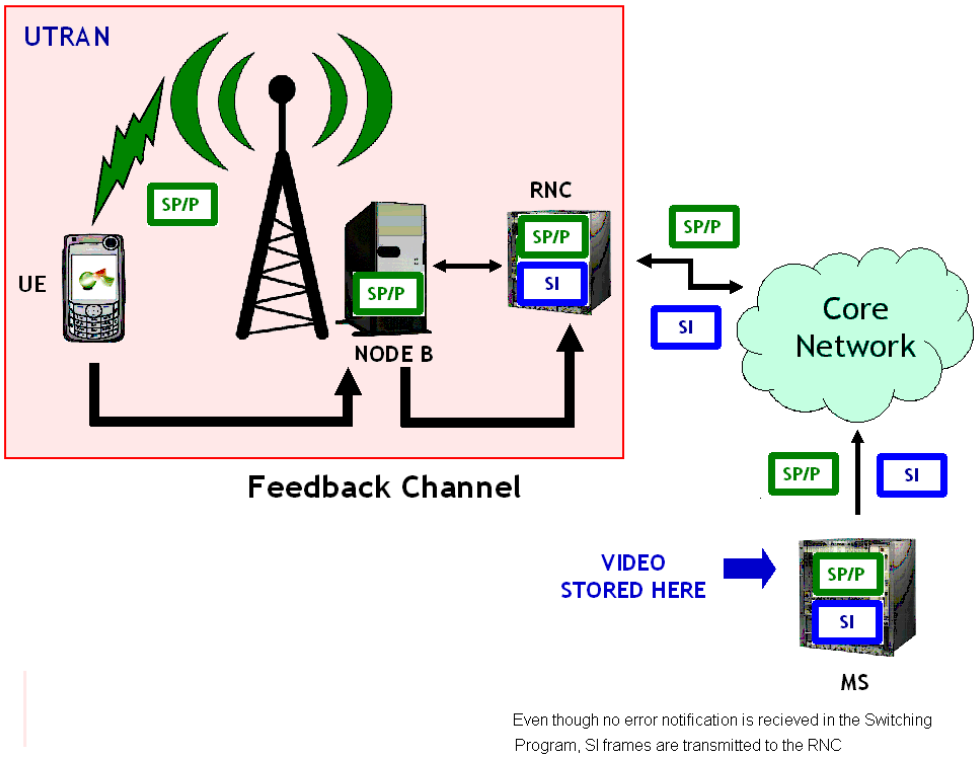


Figure 5.6: Switching program located in the Radio Link Controller

5.2 Program performance

5.2.1 Overview

The program performed implements the switching process and also simulates differents features of the UMTS network. The challenge of implementing the system in a real UMTS network could be an interesting continuation of this project. In a real situation, specially for real time applications, as soon as the RTP packets are encoded the program should recieve them and decide wether to switch or not. In this simulation the whole encoded sequence is avalaible in the program.

The output of the Switching Program is directly decoded, and therefore no transmission channel have been used. No errors would be in the decoded bitstream and no switching would be implemented. As a conscequence, the errors in transmission have also been simulated in the program following the information from the error file.

The error information should be received a little time after the error in transmission has occured, applying the switching in case of error as soon as there is an SP frame. However, the delay of the feedback channel and the error information have to be simulated also from the error file.

Figure 5.7 shows the different routines of the Switching Program as well as its inputs and outputs. Both bitstreams, one with SP and the other with SI frames inserted, are inputs of *save_vectors*, a routine that divides its RTP packets into several fields, extracting some information about them. This data is used later to help in the implementation of the switching. These fields and information are stored in vectors saved in memory.

Simulate_Different_Delays routine has the error file and the delay as inputs. The user decides which delay the feedback channel has, depending on the desired UMTS network enviroment to be simulated. This routine also loads the vectors with the information about the RTP packets.

With the information about the errors, the RTP packets and the delay, the switching is implemented in *Main_Switching* routine. This function calls other functions that simulates the errors in transmission extracting the information from the error file, and also the delay of the feedback channel.

Since a comparison with the performance of bitstreams with only I and P frames is a part of this project, also the errors in transmission must be simulated for these bitstreams. However, no switching is implemented since I frames are always sent with the same periodicity and therefore the information about the delay in the feedback channel is useless. Consequently, routines used for bitstreams with SI frames are different to those utilized for bitstreams with SP frames inserted.

The bitstream with errors inserted and the switching implemented to be decoded is one of the outputs of the program but not the only. Some text files have been created during the implementation to inform about different aspects which will be useful next. These files includes information about the size of the bitstreams, the probability of sending an SI frame instead of an SP frame and also information about the RTP packets.

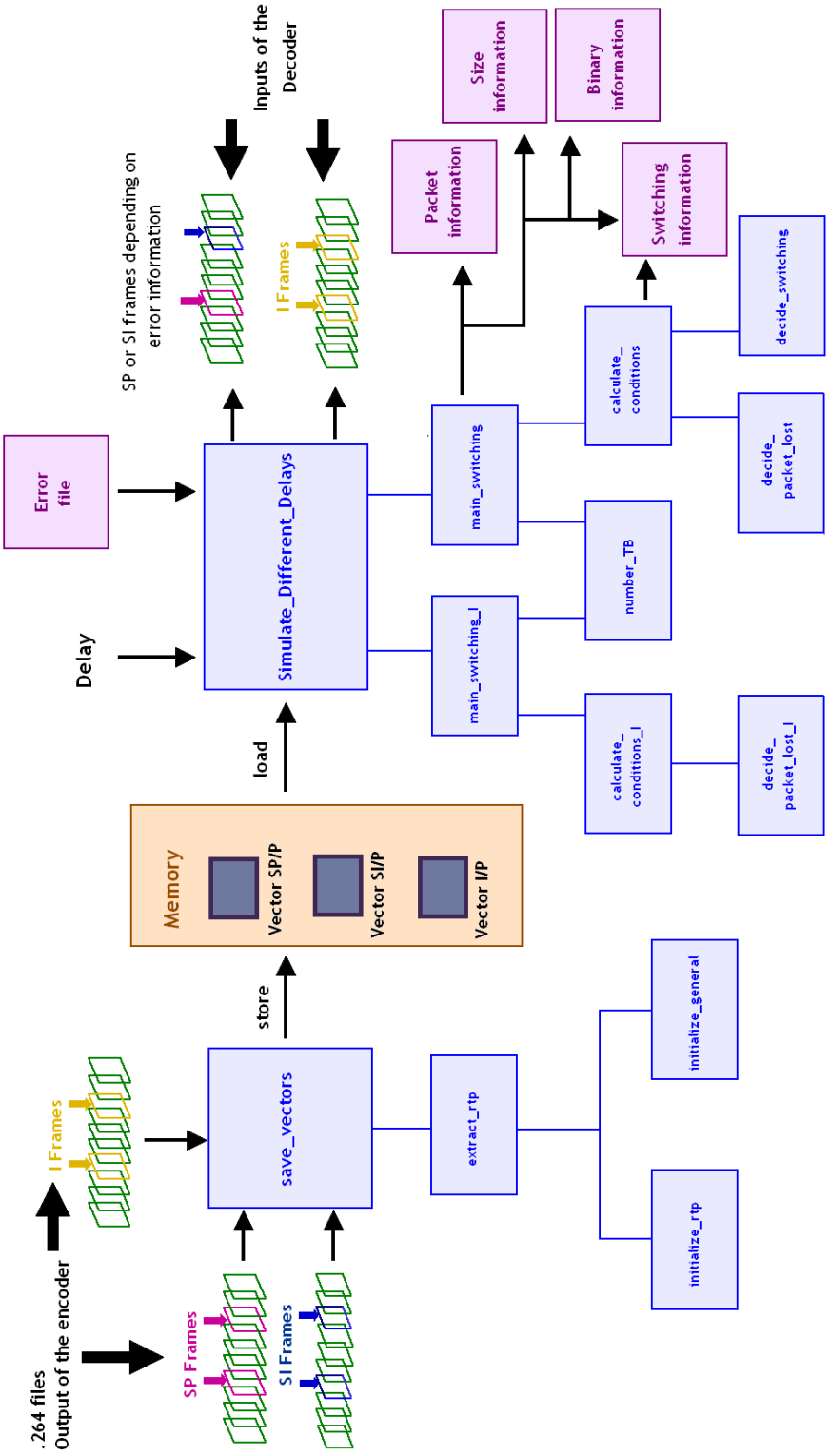


Figure 5.7: Routines of the Switching Program

5.2.2 Extraction of the RTP packets

The first part of the switching program consits of various routines that extract the different fields of the RTP packets resulted from the encoder (Figure 5.8). In a real case, not the whole bitstream would be avalaible to be extracted (obvious in real time applications).

The different parts of the bitstream are saved into two vectors, and every element correspond to one packet. In one of these vectors the information is saved in the same format as it is in the encoded bitstream in order to directly be used in the switching. Both bitstreams with SP and with SI frames inserted must be saved to implement later the switching. The other vector, stores some interesting information decodified from the RTP packet that will be needed to implement the switching.

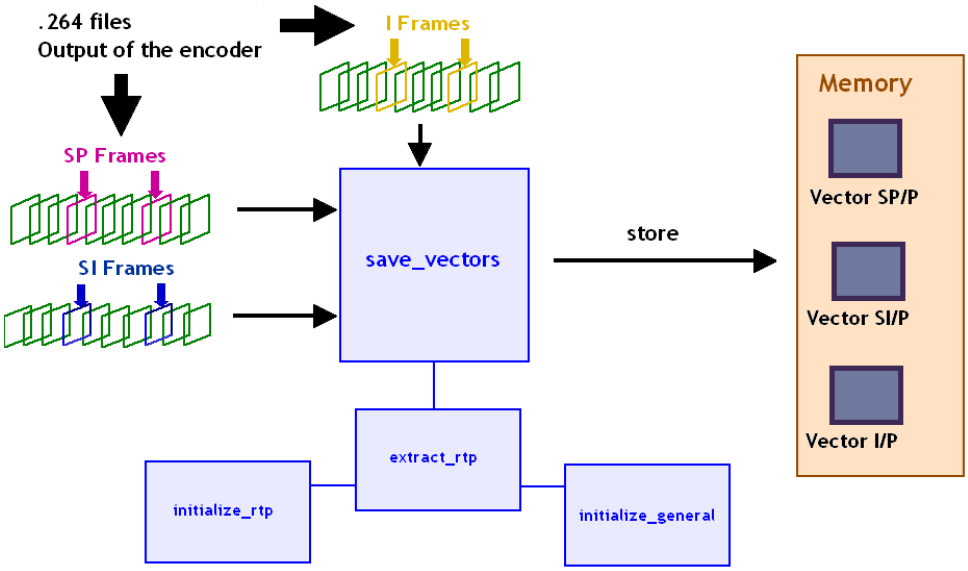


Figure 5.8: Routines to extract RTP packets

The main routine consist of a loop that runs all the packets of the bitstream. The information is extracted from the packet. But the bitstream is a string of bits with no apparent division between each packet. However the encoder has added some bytes at the begining of every packet informing about its length. The process is described in Figure 5.9 and explained next.

The first four bytes corresponds to the length of the RTP packet including its header. This information is codified in little endian. In the example depicted, the length is equal to 21. Afterwards there are 4 syncronization bytes which bits are always equal to 1. The packet header is always equal to 13 bytes, 12 corresponding to the RTP header and one to the NALU header (NALU is the term used to talk about NAL units, packets in which the coded video is organized). With this information it is very easy to calculate the length of the payload, just by substracting headers length to the length of the whole packet. In the example of the picture 21 - 13 = 8 is the length of the payload. Consequently the next 13 bytes corresponds to the header and the following 8 bytes to the payload of the packet.

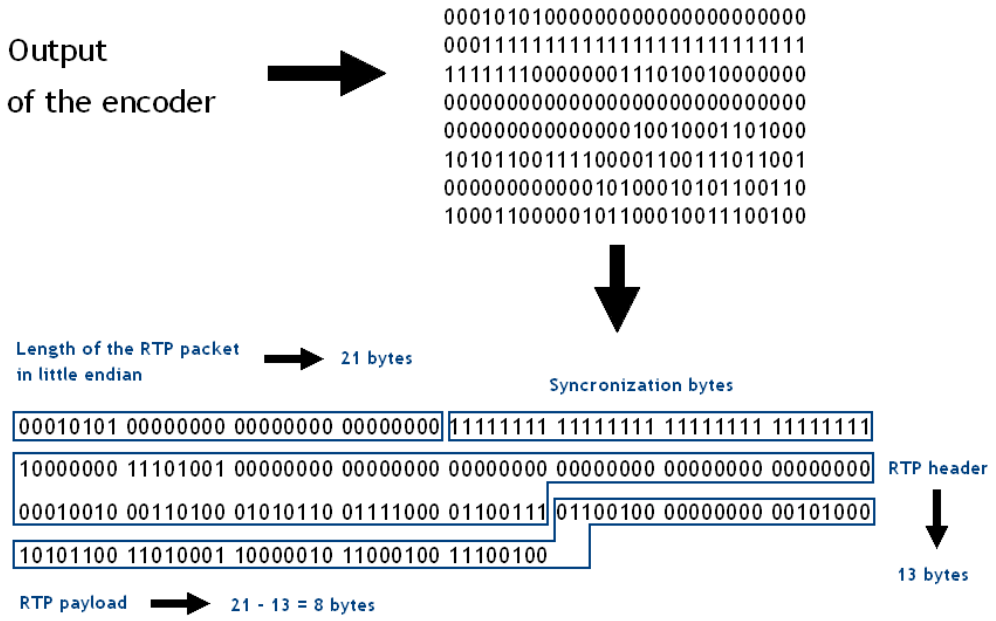


Figure 5.9: RTP packets output from the encoder

```
@79293 SH: first_mb_in_slice                    0000001100011 ( 98)
@79306 SH: slice_type                                   00110 (  5)
@79311 SH: pic_parameter_set_id                             1 (  0)
@79312 SH: frame_num                                     1001 (  9)
```

Figure 5.10: First bits of the RTP payload extracted from the trace file.

Besides, some useful information is extracted from the payload, the slice type and the frame number corresponding to this packet. This information will be used in the switching to help on its impelementation. Here above there is an example of the first bits from the payload, extracted from the trace file (Figure 5.10) :

The *pic_parameter_set_id* is always 1 bit and the *frame_num* corresponds to a fixed number of bits (four in this example). However *first_mb_in_slice* and *slice_type* are codified in such a way that the number of zeros that precede the bits of information are always one bit less that those. In this example, the *first_mb_in_slice* is equal to 0000001100011, and there are 6 zeros (000000) before the first 1, that means that the information bits (1100011) are $6 + 1 = 7$ bits.

At the end, the two vectors with the information about the RTP packets are stored in memory in order to be used later in the switching. The reason of this storage is to avoid this operation when it is wanted to simulate the same bitstream but with several delays (no extraction is needed but only to load this vectores from memory).

5.2.3 Switching main routines

The routines involved in the switching process are depicted in Figure 5.11. This figure is a simplified diagram of the process. The previously extracted and stored RTP packets are loaded from memory to two vectors. One of these vectors stores the bitstream information in the same format as the output of the encoder, the same format to be decoded. The other vector, stores some information decoded from the packet and used in the switching process, like the number of frame and the slice type. Every element of both vectors corresponds to one RTP packet.

The other two inputs of this process is the delay and the error file. The delay is introduced by the user and simulates the transmission time of the feedback channel. Depending on the characteristics of the simulated UMTS network, the delay would be bigger or smaller. The error file is used to simulate errors in transmission and also as the information received through the feedback channel to inform about these errors.

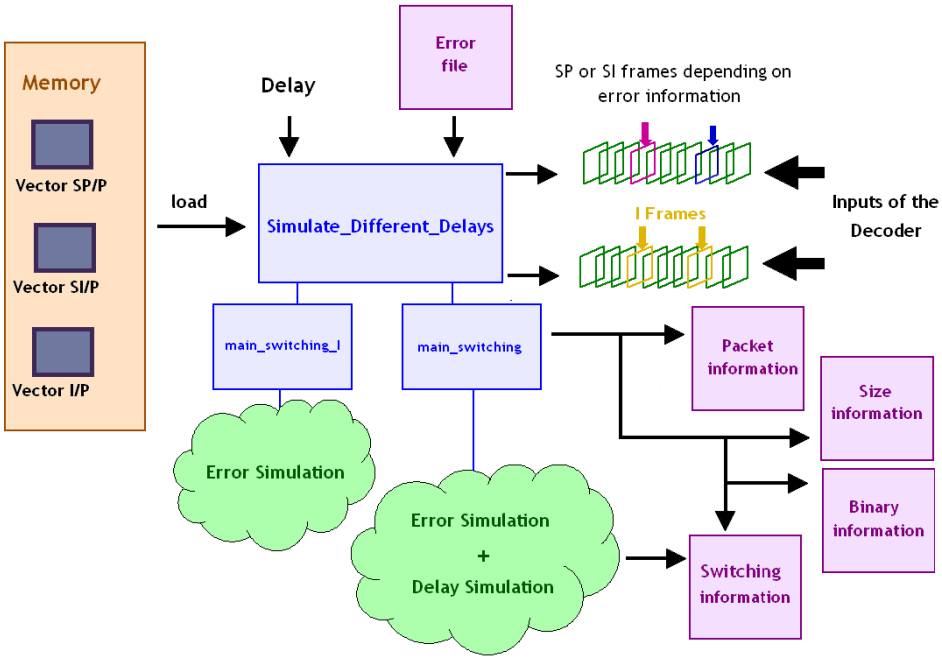


Figure 5.11: Routines to implement the switching process.

The output of this process is the bitstream with the switching implemented, in case of error, in the same format as the output of the encoder to be decodified by the decoder. In case of simulating a bitstream with I and P frames, no switching is implemented but only the effect of errors in transmission.

Most of the aspects related with the simulation of errors in trasmission and the delay of the feedback channel are explained in next subsections. During the switching process some interesting information is written in several files created by these routines.

The first routine involved in the switching process is called *Simulate_Different_Delays* and a simplified diagrame is depicted in Figure 5.12. At first both bitstreams, one with SP and the other with SI frames inserted, are loaded from memory. The

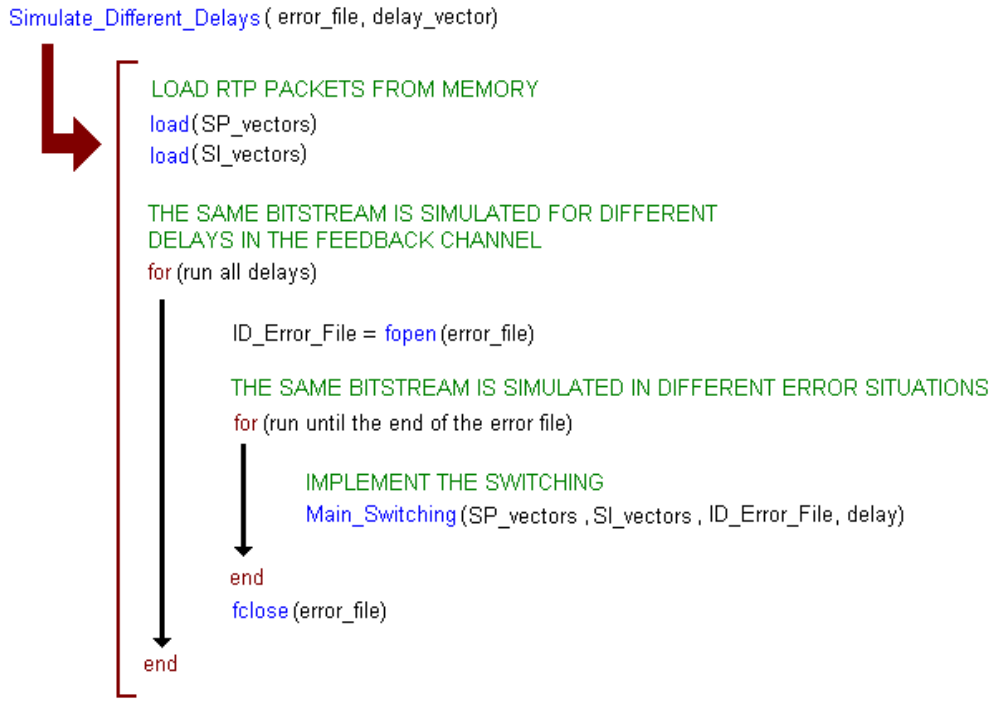


Figure 5.12: Simplified algorithm of *Simulate_Different_Delays* routine.

different delays introduced by the user are simulated one after the other. For every delay the error file is utilized from its beginning in order to simulate with the same error environment for all of them. The error file have much more information that the needed for the transmission of only one bitstream. To make it more realistic, the same bitstream is simulated several times, using different parts of the error file, to show the effect of errors in differet points of the bitstream.

In case of simulating a bitstream with only I and P frames inserted, the delay is not needed because no switching is implemented (only the effect of errors in transmission affects the bitstream).

The routine that includes the main loop of the switching process is called *Main_Switching* and its simplified algorithm is shown in Figure 5.13. Its inputs are the vectors with SP and the SI bitstreams, the error file and the delay introduced by the user. The main loop runs all the packets of the bitstream.

First, the type of frame of the current packet is controlled. Only in case it corresponds to an SP frame the *switching_condition* is checked. If it is set to one, it is a packet from an SI frame, that means that the switching is going to be implemented. The way to perform it, is assigning all the packets corresponding to the SI frame with the same frame number from the SI bitstream to the vector that stores the packets of the final bitstream. If this condition is equal to zero or the packets corresponds to an I or a P frame, the switching is not implemented and the packet from the SP bitstream is assigned to the vector with the final bitstream.

The switching condition is obtained from the routine called *Calculate_Conditions* and explained later. The packets are assinged to the final bitstream only in case the

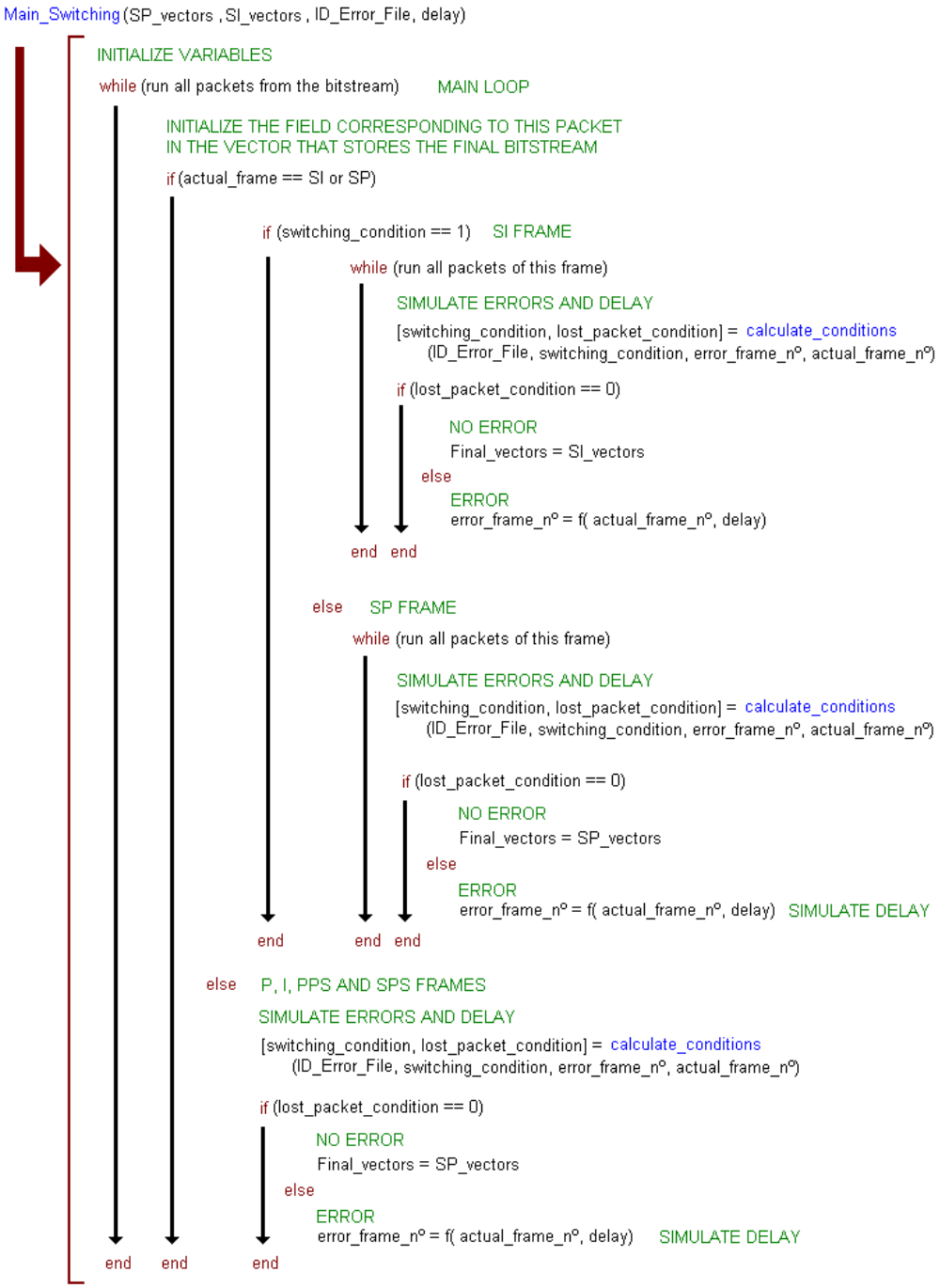


Figure 5.13: Simplified algorithm of *Main_Switching* routine.

lost_packet_condition is set to zero but this is part of the error simulation and is also described later.

The main routine of the process for a bitstream with only I and P frames inserted is called *Main_Switching_I* and its simplified algorithm is shown in Figure 5.14. The principal difference with *Main_Switching* routine is that since no switching is implemented,

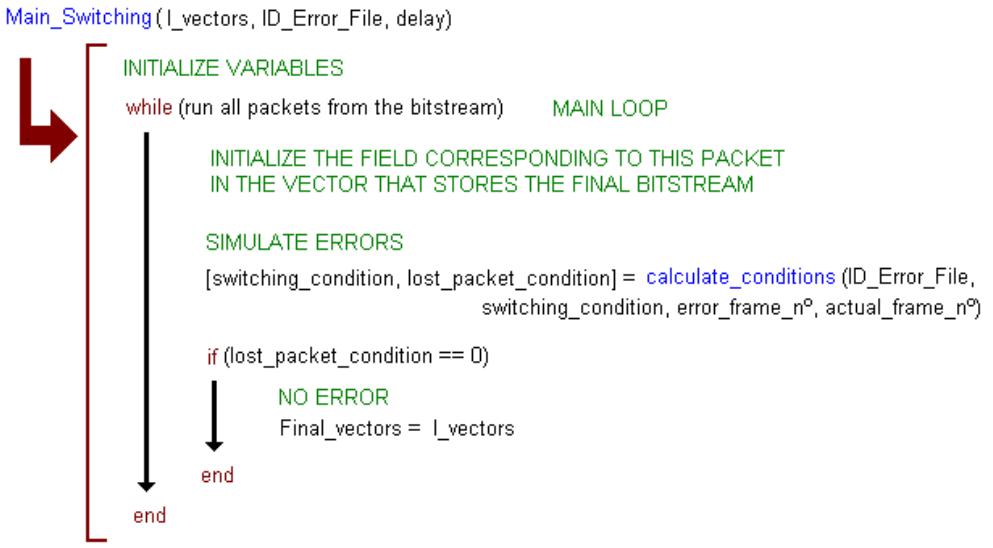


Figure 5.14: Simplified algorithm of *Main_Switching_I* routine.

it is not needed to control the type of frame and there is no switching condition. Only the effect of the errors is simulated by assigning or not the packets from the orignal bitstream to the vector that stores the final bitstream.

The different files created during the execution of these routines try to inform about several aspects of this process. In the so-called *Packet information* file, the different bitstreams are divided into packets, informing about the frame they belong to, the number of bits of the header and the payload and the number of packet. The *Size information* file contains the total number of bits and the bitrate of the final bitstream, previously calculated in the *Main_Switching* routine. There is another file with the information about the bitstreams in bynary data. The last one, the *Switching information* file describes deeply the switching process, informing about when the transmission is affected by an error and when the error is detected by the system.

The final bitstream resulted after this process is stored also in a file with proper format to be decoded by the decoder.

5.2.4 Error simulation

Avoid propagation of the errors occured during data transmission is the aim of SP and SI frames. In order to obtain good results these errors must be simulated in a realistic form.

Most of the errors occur in the wireless link between Node B and the Mobile Station. The error file uses information about the errors detected in this link, measured using a mecanism shown in Figure 5.15. In the UMTS Terrestrial Radio Access Network (UTRAN), where the errors have been detected, the IP packets are segmented into Transport Blocks (TB) [19]. Every Transport Block has a CRC field attached, which size can be configurable to 0,8,12,16 or 24 bits. This field is checked in the Mobile

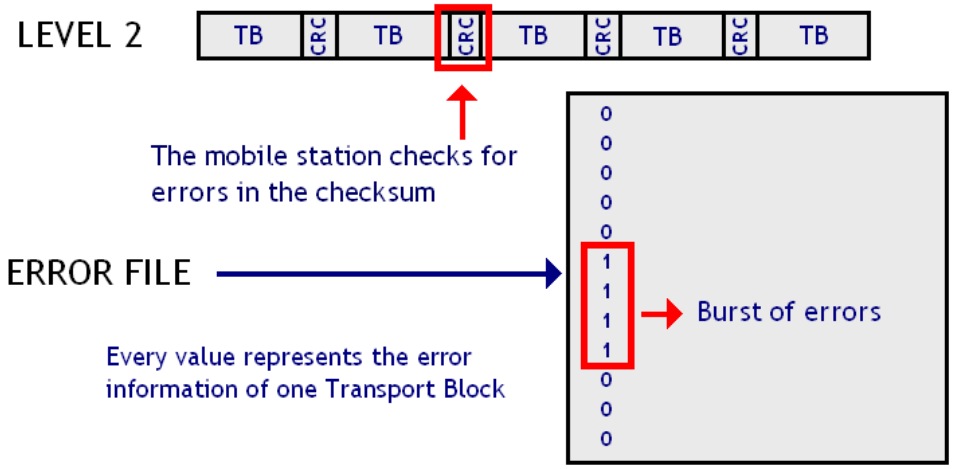


Figure 5.15: Error file: origin and format.

Station to detect errors in transmission. In case of no error in a TB, a 0 is written in the error file and if an error is found, a 1 is set. Normaly errors occur in bursts, described by several 1s together in the error file. This would be the information transmitted through the feedback channel.

Routines involved in error simulation are depicted in Figure 5.16. It can be observed that this process affects both, bitstreams with SP frames inserted and bitstreams with only I and P frames.

The number of values to read from the error file are calculated in *number_TB* and this routine is called by *Main_Switching*. To be realistic in the simulation, only the

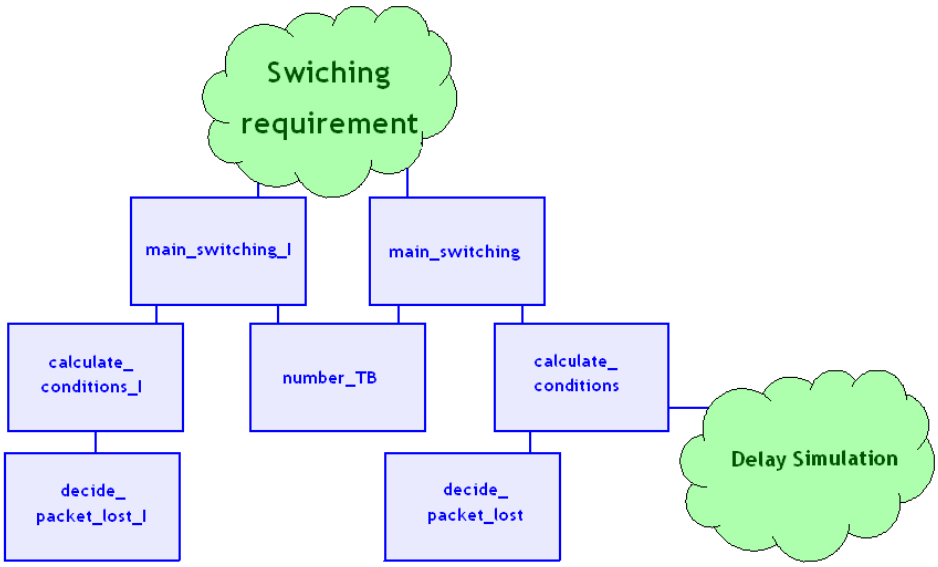


Figure 5.16: Routines involved in the error simulation.

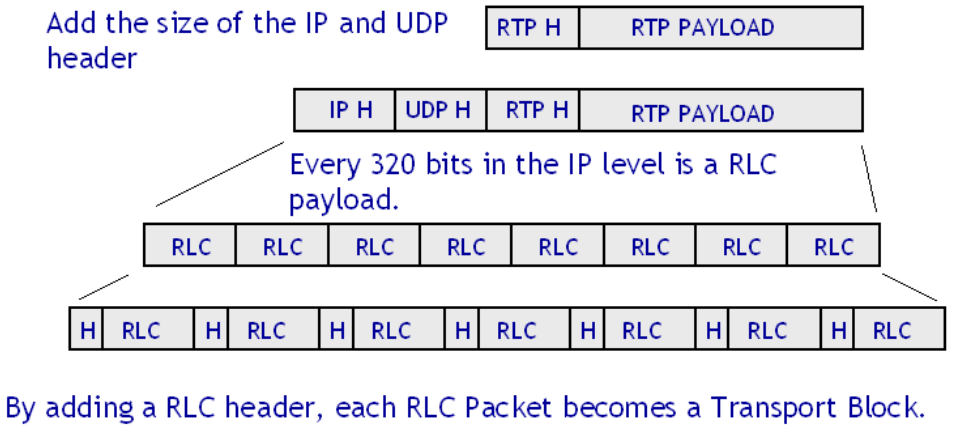


Figure 5.17: Fragmentation of an RTP packet into Transport Blocks.

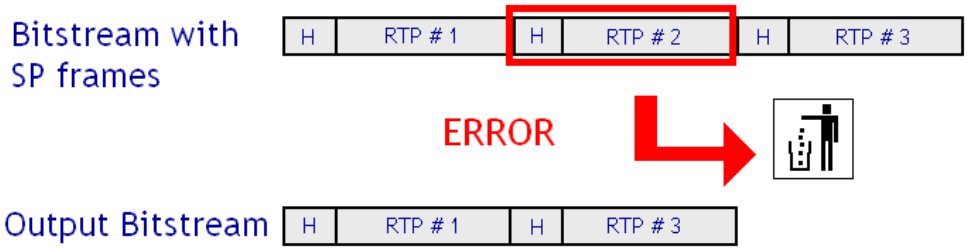


Figure 5.18: Discard process.

TBs of the packet we are working with at that moment, are read. However, the packets of our bitstreams are RTP packets before being segmented by the UTRAN into TBs. In order to know in how many TBs every RTP packet would be divided, fragmentation is performed (Figure 5.17).

It is an IP packet what is fragented into TBs and therefore, the UDP and IP headers must be added [19]. The UDP header are 8 bytes and the IP header are 20. For a data rate below 384kbits/s, the payload of a Radio Link Controller (RLC) is 320 bits and 640 bits for higher data rates. A TB corresponds to a RLC payload and a header, which size depends on whether they work in acknowledged mode (AM), allowing RLC packet retransmission or in unacknowledged mode (UM) where the error is only detected and no feedback is allowed.

However the size of this header is not important to calculate the number of TBs, since it is the same as the number of RLC payloads. In our simulation we work with data rate below 384kbits/s and consecuently the number of TBs is resulted from dividing the number of bits of the IP packet to 320:

$$Number\_TBs = \frac{RTP\_Packet + UDP\_Header + IP\_Header \text{ [bits]}}{320} \tag{5.1}$$

As it has been mentioned before, this number of TBs informs us about the number of values to check in the error file. The reading of these values is performed in *decide_packet_lost* routine. If among these values there is one or more 1s, then an error must be simulated, setting the *switching_condition* to 1. This condition is utilize in *Main_Switching* to simulate the error by copying the packet from the original bitstream to the final vector only if this value is equal to zero. In case of no errors, the output is the original bitstream with SP frames inserted (Figure 5.18).

5.2.5 Delay simulation

The decoder detects that there is an error because the timing of decoding the frames becomes not normal. So the decoder knows at the moment of decoding, that there has been an error, but not the Radio Network Controller or the Media server, where the switching is located. This information about the errors occured is sent back through a feedback channel which have a certain delay. This delay must be taking into account in order to be realistic in the simulation.

The delay is introduced in the program by calculating the following equation:

$$delay[\text{number of frames}] = fps \text{ x } delay[\text{seconds}] \tag{5.2}$$

where *fps* is the number of frames per second of the video, what is a configurable feature of the encoder. For an introduced delay value of the feeback channel it is possible to calculate the number of frames that the system needs, to know about the error.

Figure 5.19 depicts how the program works. In case of no errors, SP and P frames are sent. If an error is read from the error file, a packet is extracted from the bitstream in order to simulate the error. From this point, and after the number of frames of delay calculated by the equation, the system detects the loss of the packet, simulating the delay of the feedback channel. But the SI frame is sent only instead of an SP frame, not replacing a P frame, and the error follows propagating until the switching point. After the SI frame, good quality is recovered.

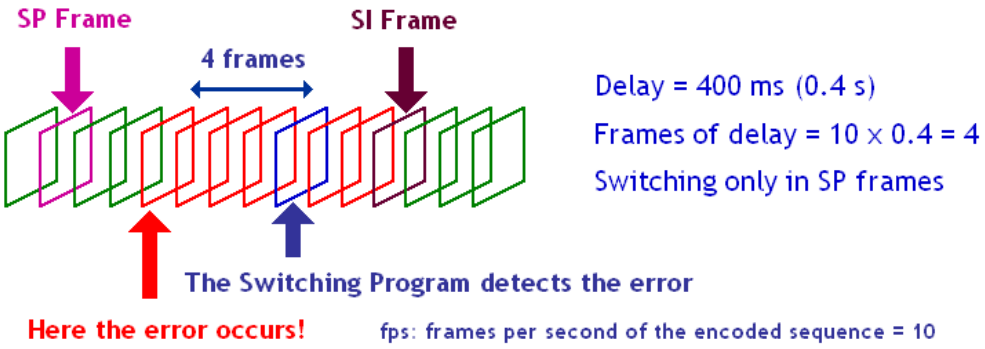


Figure 5.19: Effect of the feedback channel delay.

As it has been mentioned before in *Main_Switching* routine, if *lost_packet_condition* is set to one, the current packet will not be copied to the final bitstream in order to simulate the packet loss, and at this point, the frame where the system will detect the error is calculated by:

$$error\_frame\_n^o = actual\_frame + delay[\text{number of frames}] \tag{5.3}$$

The variable *switching_condition* is controlled by *decide_switching* routine, whose simplified algorithm is shown in Figure 5.20. Its value is only set to one in case $error\_frame\_n^o = actual\_frame$, what means that in a real case the data informing about the packet loss would have been recieved.

The variable $error\_frame\_n^o$ is only different to 0 when an error have been simulated and the problem is not solved yet. When an SI frame is sent to solve the problem, both, $error\_frame\_n^o$ and *switching_condition* are initialized.

The effect of simulating the errors in transmission and the delay of the feedback channel is depicted in Figure 5.21. In this example there are two errors, one in frame number 16 and the other in frame number 60. The difference in quality regarding to the original bitstream is 15 dBs. The delay simulated is big, 1.5 seconds. As $fps = 10$, the systems needs 15 frames to know about the packet loss. The good quality is recovered as soon as there is a switching point, when an SI is sent. In these case, the SP period used is 5, and the system knows about the first error in frame number 31, therefore the problem is solved in frame number 35, the next multiple of 5. The second error is noticed in frame number 75 and the SI is sent in frame number 80.

The feedback delay is only simulated for bitstreams with SP frames inserted and not for those with only I and P frames because they do not implement the switching, only the packet loss to compare their quality for a similar error environment.

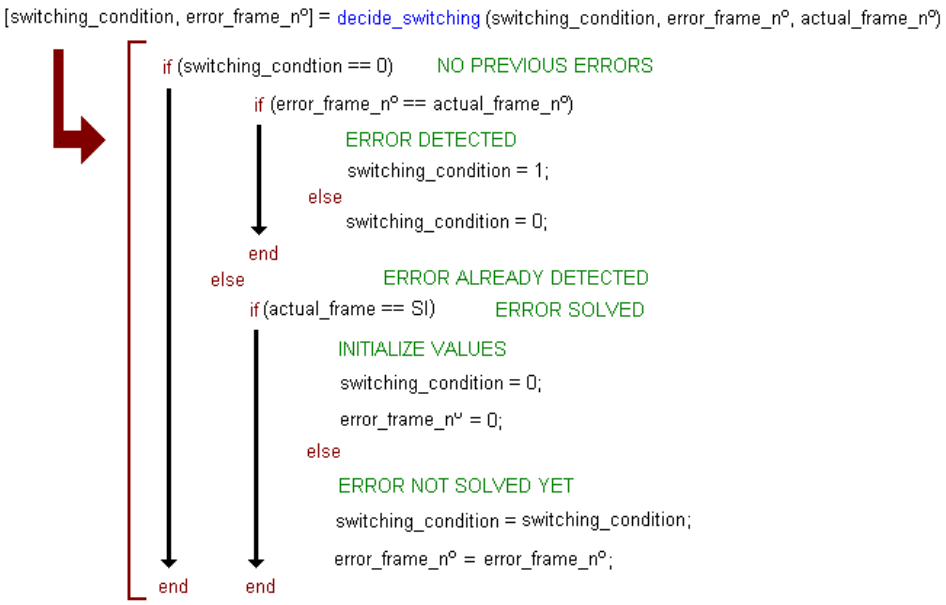


Figure 5.20: Simplified algorithm of *decide_switching* routine.

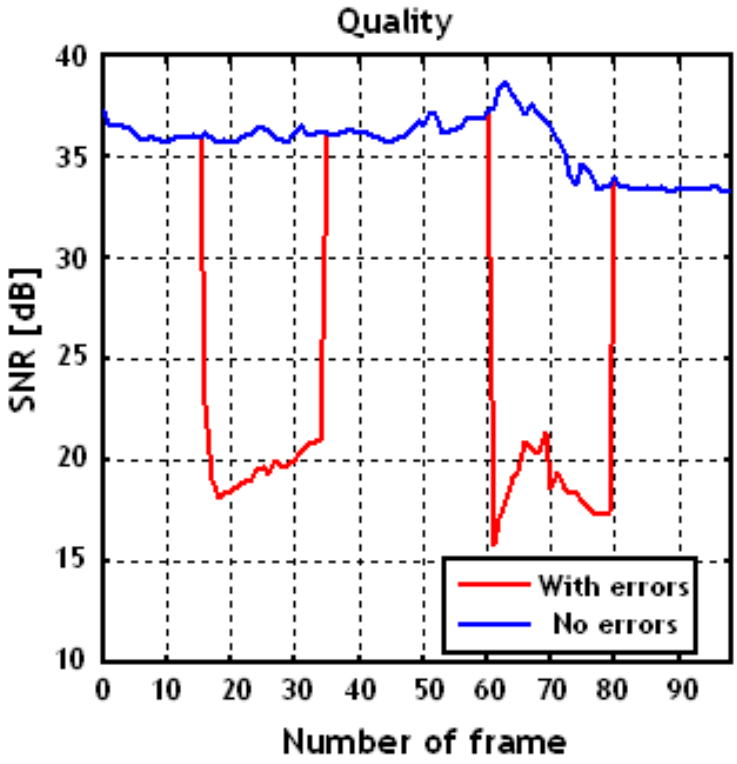


Figure 5.21: Graph showing the effect of a big delay.

Chapter 6

Simulation results

6.1 Simulation overview

The main objective of this project is to obtain a conclusion about the utility of the new frame types introduced by the new video coding standard H.264/AVC, called SP and SI frames. These new frame types have been designed with the intention of recovering the good quality while decoding after an error in transmission. This was already possible in previous standards with the existing frame types. By sending an I frame that uses no prior frames as a reference, it is possible to stop the error that propagated due to the usage of temporal dependencies. However, the size of a frame that uses spatial dependencies is much bigger than those with temporal dependencies. In previous methods, frames with spatial dependencies were sent periodically without taking into consideration if there had been an error or not. That can be considered as a waste of resources.

In the method that has been previously explained, the frames with spatial prediction, and therefore they are able to stop error propagation, are only sent in case of a notification of an error in transmission. The main advantage of these two frames is their capability to adapt themselves to the channel situation. In this chapter a deep comparison with the method of using I and P frames has been done in order to obtain

(a) Mother and Daughter (b) Foreman

Figure 6.1: Images of the two videos used.

a proper conclusion about the utility of this method.

Two videos have been used in this analysis, both of them very utilized for testing purposes. They are known as Foreman and Mother and Daughter (Figure 6.1). Foreman is a sequence where it appears a very expressive man making faces and at the end of the video there is a big camera movement to show a building. On the contrary, Mother and Daughter is a very static sequence where it appears a mother talking to her daughter but making very little movements and without a change in the background.

For every video sequence two different resolutions have been used, CIF (352x288) and QCIF (176x144). The bigger the resolution the more information is sent and the better is the quality. As it has been explained before, the method of error simulation counts the number of TBs in every packet and then the system checks this number of values in the error file. If it reads a 1, an error is simulated and the packet is discarred. The bigger the resolution, the more bits a sequence has to send and therefore more packets are created (it has been fixed a maximum size of 700 bytes per RTP packet). If there are more packets, there will be also more values to read in the error file and therefore, more probabilities of having an error in the transmitted sequence.

The switching process can be implemented in different parts of the UMTS network. In this project two possible locations have been judged as interesting, the Media Server and the Radio Link Controller. In the previous chapter the advantages and disadvantages of these two locations have been described. The way to simulate it is to use

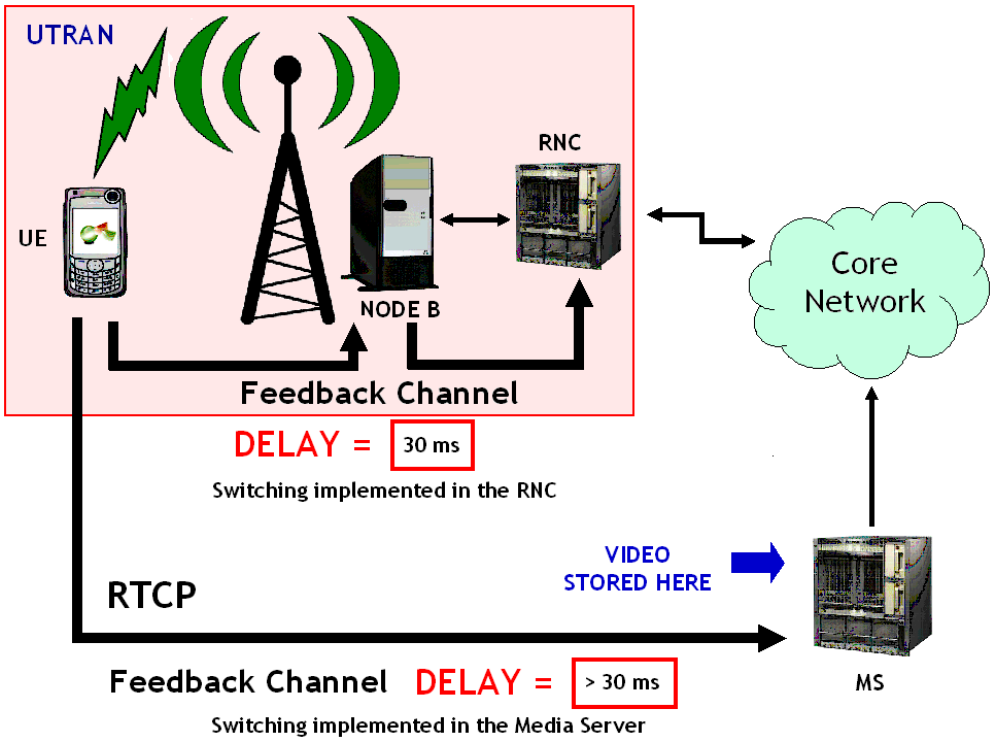


Figure 6.2: Different delays depending on the feedback channel

different delays for the feedback channel as it is shown in Figure 6.2.

Using some information about different studies from the network, it has been considered that in case of implementing the process in the RNC, the delay of the feedback channel is about 30 ms. The next frame just after the error has occurred, the system already knows about it because of the way of simulating the delay (frames per second used in the decoding is equal to 10). Bigger delays are simulated to consider also the case of implementing the switching process in the Media Server. In case of simulating a bitstream with only I and P frames, the delay does not affect the transission because no switching is implemented.

The whole simulation process is depicted in Figure 6.3. Firstly the bitstreams are encoded. The version of the encoder used is the one previously modified in order not to have problems with the reconstructed images while using SP and SI frames. All the modification process has been explained deeply in chapter 4. The encoder is configured to packetize RTP packets with a fixed size of 700 bytes. All videos have 300 frames, but only 100 are codified because of skipping 2 frames every 3 (normal in video applications). This increases the information to send because there are more difference between two consecutive codified images.

The velocity of encoding is set to 10 frames per second and that means that the video lasts 10 seconds. It can be considered as normal that every 10 seconds in any video there is a change of sequence. In case of the bitstreams with SP or SI and P frames, there is an I frame at the begining of the sequence because there is no previous picture to which temporal prediction can refer. The multipicture buffer is the place where previous pictures are stored to be used as a reference for this type of prediction. In our simulation its size is set to one, so only the last picture is used as a reference for P and SP frames.

The extended version of H.264/AVC is used because it is the only profile where the usage of SP and SI frames is allowed. The value of the QP, quantization parameter of the P and I frames, is the main configuration parameter to control the bandwidth needed. It has been selected to work for the two videos in CIF resolution for a bit-

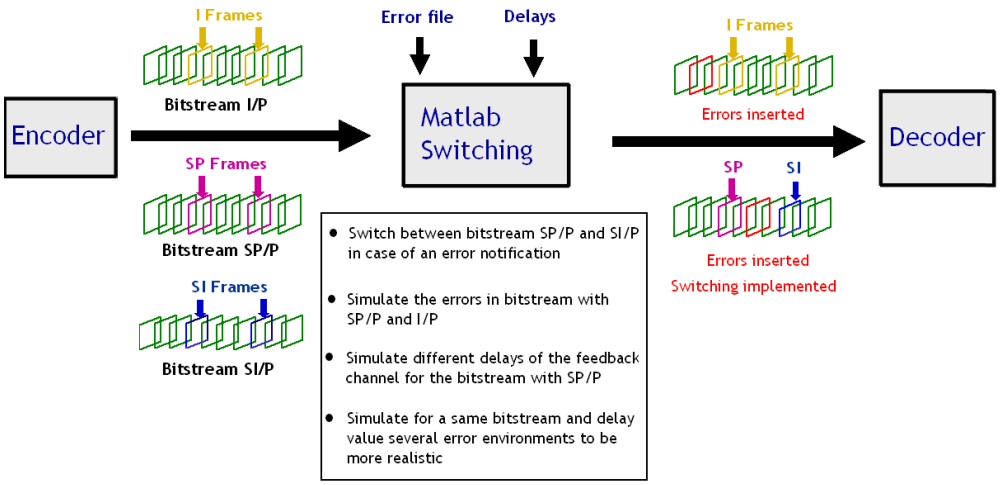


Figure 6.3: Simulation process

rate of about 220 kbits per second and for the two vidos in QCIF resolutions for a bitrate of about 80 kbits per second. Several values around these mentioned have been simulated to observe the behaviour in their surroundings. The PQP and SPQP values, quantization parameters of SP and SI frames, have been chosen following the recomendations explained in chapter 3.

Different GOP (for bitstreams with I and P frames) and SP periods (for bitstreams with SP or SI and P frames) are encoded to observe its influence on the performance of the system.

The encoded sequences are processed by the program that implements the switching process, performed in Matlab. For every bitstream different delays of the feedback channel are simulated. With a bitstream and a delay value, different parts of the error file are read to have different error situations simulated and make the whole system more realistic.

All the simulated bitstreams are decoded and the obtained results are processed by some programs, averaging the resulting values from the different errors situations simulated. Several graphs have been depicted and the most relevant are analysed in next sections.

The whole process last plenty of time not only because of the enormous number of simulations performed but also because of the encoding and decoding time needed, that is quite long. Even though it is true that the complexity of the encoding and decoding process is important, the system nowadays is rather inefficient from this point, specially for real time applications. The code of both, the encoder and decoder should be optimized, although it is also important to point out that this is normal when a standard is stil developing.

6.2 Encoding step

The first step of the simulation is the encoding of every video for different QP parameters in order to obtain results for different bitrates that will help to see more clearly their behaviour. In order to compare the two different methods, not only the two bitstreams with SP and SI frames inserted are encoded but also a bitstream with only I and P frames. For the three bitstreams similar values of bitrates have been seeked.

The first graph Figure 6.4 depicts the different efficiency for the two resolutions used, CIF and QCIF in case of using a bitstream with SP frames inserted. In this example shown, the video utilized is Mother and Daugther. It is possible to observe that QCIF resolution offers a better behaviour, since for the same number of bits sent, the quality is more than three dBs higher.

Another important aspect to remark is shown in Figure 6.5. A bitstream with I and P frames codified, from Mother and Daugther in QCIF resolution, is depicted. The intention of this graph is to analyse the effect of using different GOPs. The bigger the GOP, the best behaviour it has, and the difference between using a GOP of 5 and a GOP of 50 frames is 2.5 dBs. The same behaviour has been observed in other videos. The reason is found in the bigger size of I frames (spatial prediction). The smaller the GOP, the more I frames there are and therefore the less efficient the transmission is in case of having no errors. Of corse the more prepared to stop error propagation is

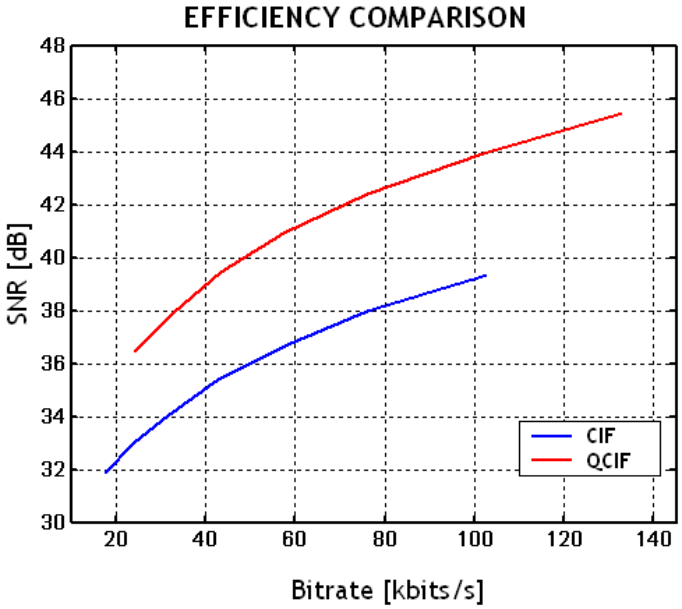


Figure 6.4: Comparison of QCIF and CIF resolution efficiency

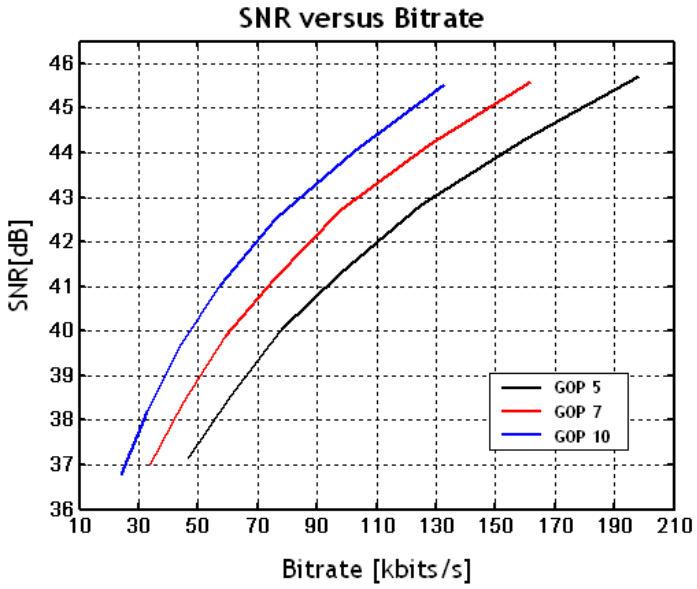


Figure 6.5: Different GOPs for Mother and Daughter QCIF

this bitstream.

The case of using different SP periods for a video sequence is not shown since the difference in the efficiency is very little and it has been cosidered as irrelevant. The only point to remark is that if we compare the SP bitstream with a bitstream of the same video with I and P frames (the GOP and the SP period is the same), the efficiency is much better for the SP bitstream. That is because of using temporal prediction always. For the SI bitstream the difference for different SP periods is extreamly important. In Mother and Daughter CIF it is of 6 dBs. For the same desired quality, more bitrate

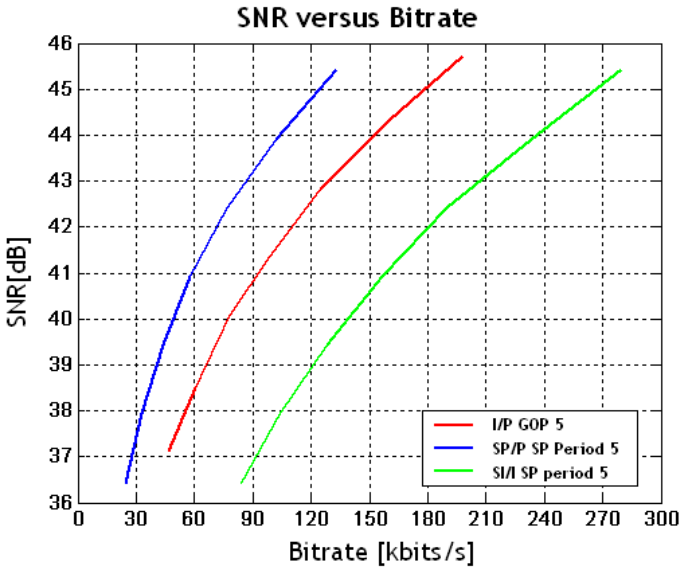


Figure 6.6: Comparison of quality for bitstreams with SP/P, SI/P and I/P frames

is needed for a bitstream with I frames inserted than for an SP bitstream, but still more bitrate than for an SI bitstream (Figure 6.6). This effect is due to the spatial prediction used in SI frames and because of the way of encoding these frames, suffering two quantification processes (by PQP and by SPQP).

To conclude, a comparison between Mother and Daugther and Foreman videos for same encoding features shows that the first one is more efficient, 7 or even 8 dBs. That is a great difference and the most probable reason has to do with the content of the video. Since Mother and Daughter is more static there is more temporal correlation among its images and its prediction is easier. As a consequence the residual sent is smaller.

6.3 The effect of the feedback channel delay

After being encoded, the switching process is implemented to the bitstreams. The effect of the errors in the channel is simulated as well as the delay of the feedback channel. In this subsection the effect of having different delays in this feedback channel is explained. As it has been explained before, if the switching process is implemented in the Radio Network Controller, the time needed to inform the switching program about the errors detected in the mobile station is smaller because this device is closer to this point of the UMTS network.

Normally, the feedback channel delay in those cases is about 30 ms. The system knows about the error only one frame after it has occurred because of the way of simmulating the delay. Therefore the switching will be implemented almost as soon as the error ocurrs. We can say that this system is well adapted to the channel situation.

On the other hand, in case the switching process is implemented in the Media Server, the time needed for the feedback channel is bigger as it is rather far away from the mobile station where the error is detected. Several values have been used,

simulating different UMTS scenarios.

The first two graphs depicted (Figure 6.7 and Figure 6.8) analyze the effect of different delays in the frame domain. These graphs have been obtained from Foreman video n CIF resolution and using an SP period of 5. In the first one, a small delay of 100 ms has been used. This corresponds with the switching process located in the RNC. There are two errors, the first in frame number 12 and the second in frame number 54. Since the system detects the error very soon, an SI frame is sent only a little frames later, instead of the corresponding SP frame. In the first error case the SI frame is sent in frame number 15 and in the second case in frame number 60 (both of them multiples of 5, because of the SP periodicity used).

The second graph show the case of having a big delay of 1500 ms. This situation corresponds to the switching process located in the Media Server. In this case the errors occurs exactly in the same frame positions (because we have used the same

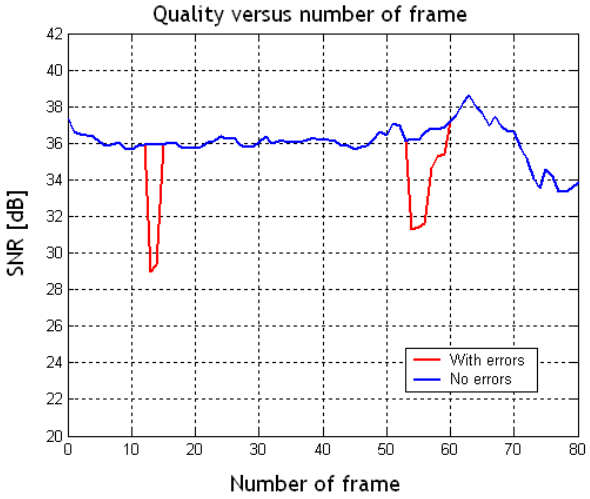


Figure 6.7: Effect of different delays (Delay 100 ms).

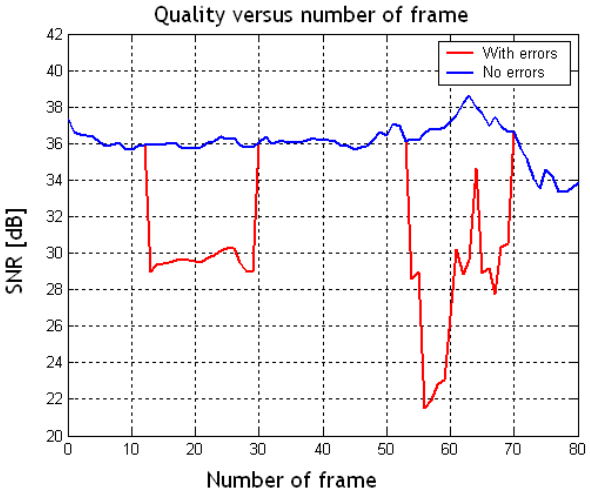


Figure 6.8: Effect of differents delays (Delay 1500 ms).

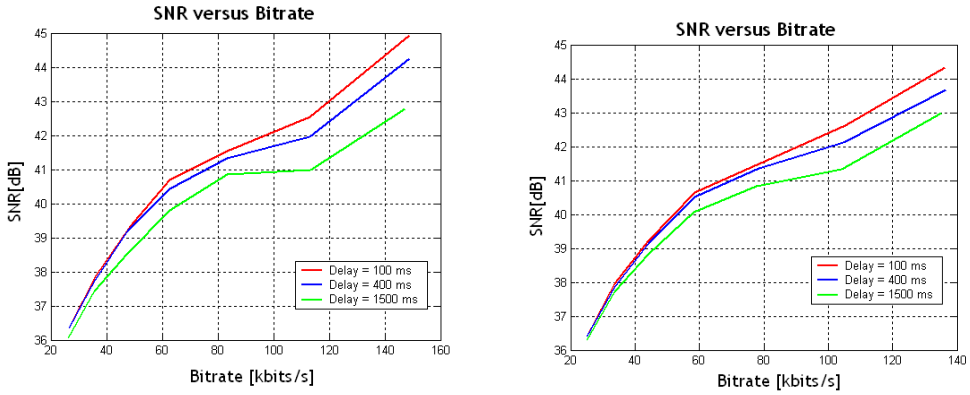
encoded sequence for both delays), in frame number 12 and in frame number 54. In this case the systems needs 15 frames to know about the error and for that reason the switching is not implemented untill frame number 30 and 70 (both of them also multiples of 5 because the same SP periodicity has been used). That supposes more than 15 frames with the error propagating while decoding and therefore, a loss in quality of approximately 7 dBs during those frames. That means a big degradation of the video is resulted with such a big delay in the feedback channel.

The effect of this degradation is shown in Figure 6.9. The video used is Foreman in CIF resolution and with and SP periodicity of 7. These curves depict the SNR in dBs depending on the bitrate. To obtain different bitrates in the encoded sequence, different QP parameters are used. The bigger the QP, the smaller the bitrate needed to transmit the information and the worst is the quality of the reconstructed video. It is possible to observe how a bigger delay decreases the performance of the system. For a bitrate of 220 Kbits per second the quality for a delay of 1500 ms is 1.5 dBs lower than the quality having a feedback delay of 100 ms.

Figure 6.9: Degradation of the quality due to the effect of the delay.

When utilizing different SP periodicity, the effect of having different delay values in the feedback channel changes. In 6.10(a) and 6.10(b) an SP periodicity of 5 and 50 is used respectively. It is possible to see how the influence of the different delays is very little for low bitrates. That is because of the way errors are simulated in the switching program, and for those values there are almost no errors. Consecuently very little switchings are required and their effect is insignificant. On the contrary, for high bitrates the number of errors in transmission increases and more switchings are needed. In those cases the effect of having different delays has a bigger influence on the system performance and for that reason the curves of the different delays are more separated.

For an SP period of 50, using a big delay supposes a degradation in quality but not as big as for an SP period of 5. That is because the switching is only possible in frame number 50 and even for small delays the system must wait untill this position



(a) SP period = 5

(b) SP period = 50

Figure 6.10: Effect of delay on different SP periodicity.

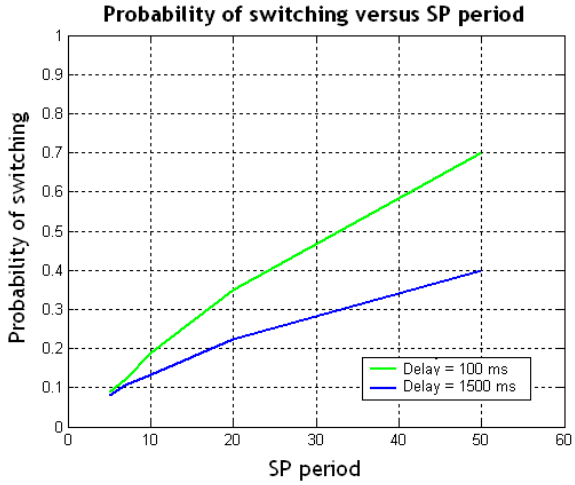


Figure 6.11: Probability of switching depending on the SP periodicity.

to implement it. However for an SP period of 5, if the delay is small, the switching can be implemented very little frames after the error has occured. In the graph this is described in the difference in the quality of the curves for every different delay. For an SP period of 5 and high bitrates, the difference in quality between a delay of 100 ms and a delay of 1500 ms is about 2dBs. For an SP period of 50 this difference is reduced to 1 dB

In Figure 6.11 the probability of switching depending on the SP periodicity for different delays, is depicted. The bigger is the SP periodicity the higher is the probability of switching because there is more time between two switching frames and it is more probable to have an error in between. However, for big delays the probability of switching decreases specially for big SP periods. The bigger is the SP period the more these two curves diverge. For an SP period of 5 the two curves coincide; for an SP period of 50, if the delay is 100 ms, the probability of switching is about 70 per cent and if the delay is 1500 ms this probability is only of 40 per cent.

The reason of these values is explained in Figure 6.12. In case of using an SP

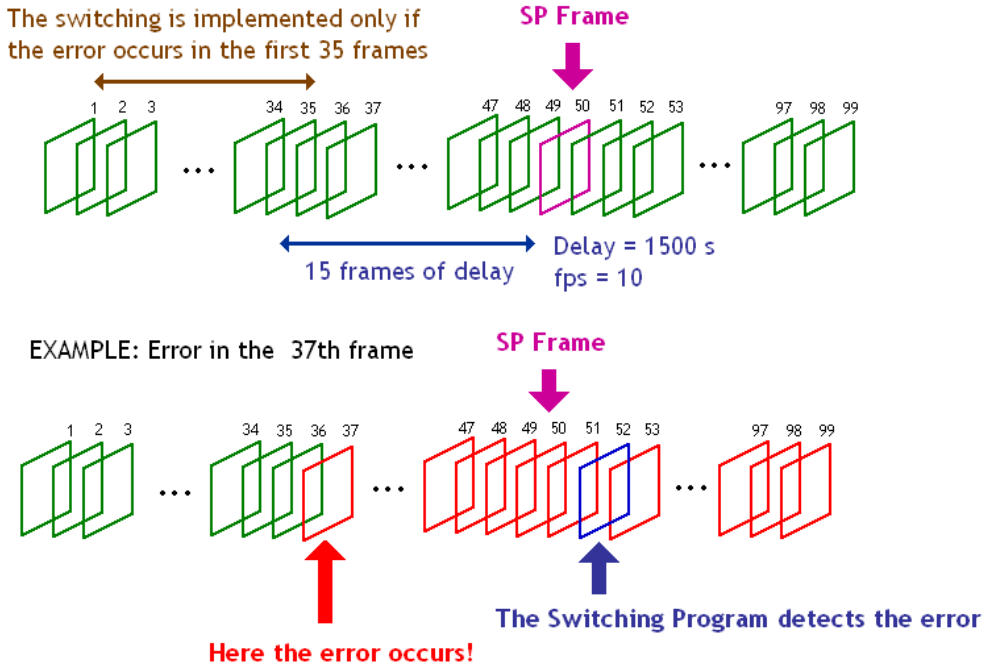


Figure 6.12: Switching process for big delays and big SP periods.

periodicity of 50, the switching is only possible in frame number 50 since the sequence has only 100 frames. If the delay of the feedback channel is equal to 1500 ms, that means that the system needs 15 frames after the error has occured to know about it. 50 - 15 = 35, this is the number of the last frame where the error must occur to implement the switching. Therefore the good quality after an error is recovered only if the error occurs in the first 35 frames of the sequence. An example is also shown below, where the error occurs in frame number 37. That means that it is in frame number 52 when the system knows about the error. In that case the swiching can not be implemented because it is only possible in frame number 50, 2 frames before we know about the error. For that reason there is bad quality in the system from frame number 37 until the end of the sequence, in frame number 99.

6.4 Performance Analyisis

This is the last section of this chapter and analyses the performance of the implementation of the switching process. The effect of the encoding and the delay of the feedback channel has already been explained. In next paragraphs a comparison between the behaviour of the decoded I/P bitstreams and SP/P bitstreams is described to help to find a conclusion about if SP/P bitstreams are efficient and interesting to be used in future applications.

The first graph (Figure 6.13) shows the effect of using different SP periodicities on the quality of the reconstructed image. The video used for this graph is Foreman in CIF resolution, but the results are very similar for the other videos and resolution.

It is possible to observe that the smaller the periodicity the best is the performance. However this difference is not extreamly big. The three curves almost coincide for small bitrates, and for a big bitrate the difference is bigger; this is because of the way of simulating the errors in this simulation. For big delays the probability of having an error in a frame increases and in those cases it is more interesting to be prepared to it, having an SP frame close to implement the switching.

SP frames have a little worse quality than P frames but they can be switched to SI frames if it is needed. In this example we are using a small delay, smaller than 100 ms, and that means that only one frame after the error has occurred, the system already knows about it. As it has been explained before, this is the case with the best performance. The bigger the delay, the worse is the adaption of the system to the errors in transmission. That means that the small loss of quality, because of sending

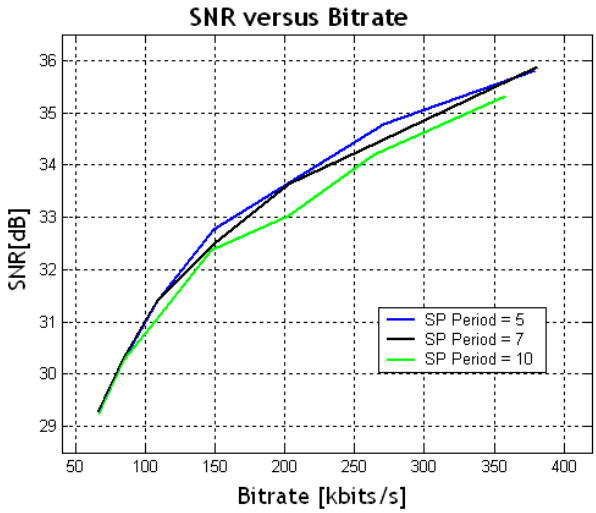


Figure 6.13: Performance of different SP periodicities in decoding.

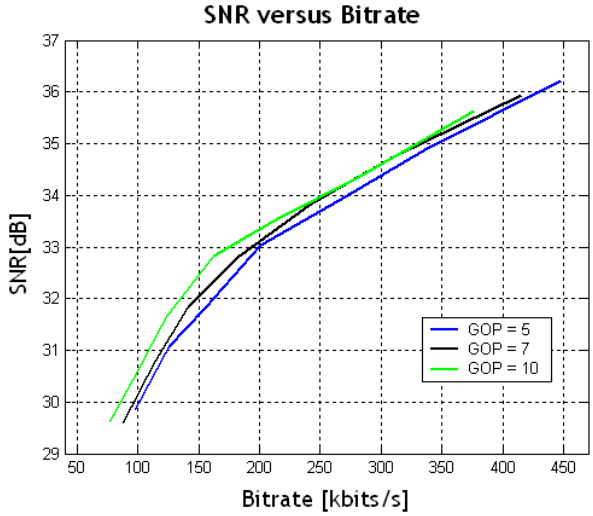


Figure 6.14: Performance of different GOPs in decoding.

SP frames with a small periodicity, is worth due to the ability these frames have to adapt themselves to the error environment. The smaller the SP periodictiy, the sooner it is possible to implement the switching and send an SI instead of an SP frame to stop the propagation of the error.

The second graph (Figure 6.14) show the performance of using different GOPs for the same video and resolution as in the previous picture. The smaller the GOP, the more I frames must be sent and therefore, the more bits are transmitted. However, in case of error, the problem is solved earlier because the frames with no dependences to prior frames are closer one from its following. In case of small bitrates, this supposes small probabilty of having an error in a concrete frame. Then it is really more interesting a large GOP, because I frames are not needed to solve problems and they are too big. However for big bitrates the errors increase and then the difference of performance for the different GOPs are not so clear since large GOPs suppose more time to wait to solve them.

The next two pictures show the degradation of the encoded sequence due to the errors in transmission. Both of them correspond to Foreman video encoded in QCIF resolution. The first one (Figure 6.15) depicts the degradation of a sequence encoded with SP frames inserted. The black curve is the encoded sequence or an ideal decoded sequence in case of a perfect transmission. It is possible to observe how the red curve is closer to the black one for small bitrate values. That is because in those cases less errors affect the transmission.

The second picture (Figure 6.16) show the same degradation but for a sequence with I frames inserted periodically. The effect of the errors is very simillar and the degradation increase with the bitrate because there are more errors per frame affecting the transmission.

The next three graphs compare the results of the original encoded sequence, the decoded sequence with SP frames inserted and the decoded sequence with I frames for three different videos. The encoded sequence would be the reconstructed image in the decoder in case of a perfect transmission without errors.

The GOP and the SP period chosen are the same, because they have a similar reaction capability in case of an error for a small delay of the feedback channel. In case of an error and a small delay, the system detects it a frame after it has occured and the switching process is implemented in the next SP position. A small delay has been chosen because in other case the reaction of the stream with SP frames insterted is much slower and therefore its ability of stopping error propagation is not so comparable. An SP period of 5 has been utilized because as it has been shown in a previous picture, this small periodicity has the best performance.

The first graph (Figure 6.17) corresponds to the video Foreman in QCIF resolution. In the three pictures it is possible to observe how for slow bitrates, the decoded sequence with SP frames is closer to the encoded sequence or with no errors in transmissions. As we have already explained this is because the error rate for these bitrate values is quite low and therefore there is almost no degradation of the quality and almost no SI frames are sent (what would increase the size of the transmission).

The second graph (Figure 6.18) represents an encoded and a decoded sequence of the video Mother and Daughter in QCIF resolution. We can appreciate how the bigger

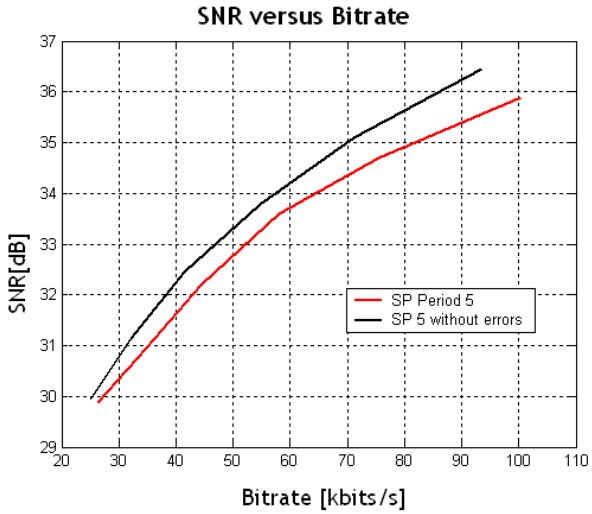


Figure 6.15: Performance of the encoded and the decoded sequence (SP/P).

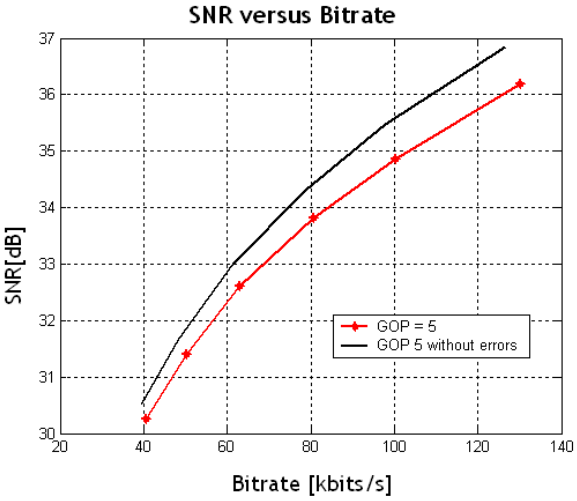


Figure 6.16: Performance of the encoded and the decoded sequence (I/P).

the bitrate the more separate are the curves of the encoded sequence and the decoded one. This is also because of the effect of more errors affecting the transmission, what decreases the quality and increase the bitrate because of the need of sending SI frames.

The third graph (Figure 6.19) corresponds to the Foreman video with CIF resolution. Here it is possible to see how the distance between the curve of the decoded bitstream with SP frames and the decoded bitstream with I frames becomes smaller while increasing the bandwidth. This fact have the same explanation used before, the bigger the bitrate, the more probability of having an error on a frame. The switching is more probable and therefore more SI frames are sent. These SI frames are even bigger than I frames, and as a consequence more bits are needed for the tranmsission. This added to the fact that the quality is degradated because of having more errors makes the switching process not so interesting. In this example the difference between

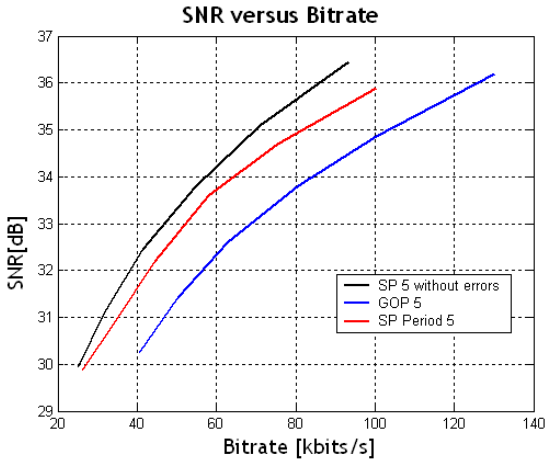


Figure 6.17: Effect of transmission errors on efficiency. Foreman (QCIF)

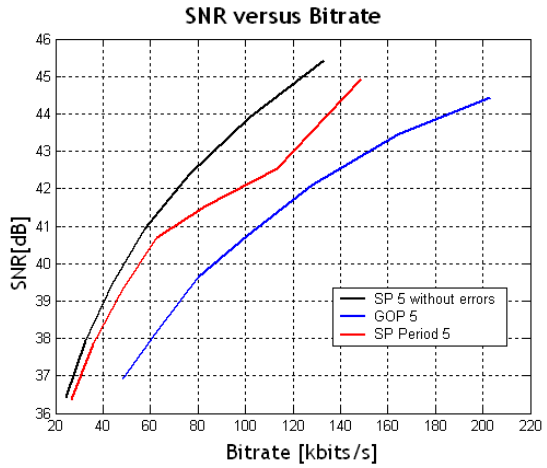


Figure 6.18: Effect of transmission errors on efficiency. Mother and Daughter (QCIF)

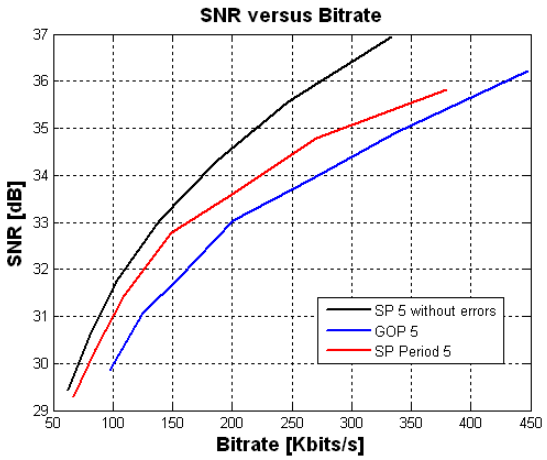


Figure 6.19: Effect of transmission errors on efficiency. Foreman (CIF

the two graphs changes from 1 dB for small bitrates to 0.5 dBs for big bitrates.

The bigger difference between the efficiency of the bitstream with SP frames and the bitstream with I frames is in Mother and Daugther video, where the difference is almost 2 dBs. This have to do with the video content. This video is more static, the image have little changes between two consecutive frames, and therefore the temporal dependences are higher. Therefore temporal prediction is better and the residual obtained is smaller. As a consequence a good quality is obtained for P frames with a smaller charge of bits and it is worther to send the less I frames, as possible depending on the error statistics.

For other SP periodicties the curves of the bitstream with SP frames and the one with I frames are closer one to the other. With such a big distance between two SP frames, the system losses its capability of solving the problem in a short period of time. Moreover, the bigger the GOP, the less I frames to send, which increased enormously the bitrate of the stream. Eventhough the system with I frames losses its capability too, an small rate of errors does this larger GOP more efficient. In case of error, the system needs more time to stop its propagation but this error is enough unprobable that in media the stream works better. However, for different SP periodicities this effect is not so clear since an SP frame has a similar size than a P frame and the degradation of the quality is not so critical.

The last three graphs depicts the probability of switching for the three previous videos. For big SP periodicities the probability of switching is greater. This is because there is more time between two consecutive switching frames and therefore, for a same error rate the probability of having an error between two SP frames increases. It is also possible to observe the effect of the bitrate. Because of the way of simulating the effect of the errors in the switching routines, the bigger the bitrate the more values from the error file are read and therefore the more probabilities of finding a 1 (that corresponds to an error to simulate). The effect of increasing the bitrate is greater for big SP periods and for them the slope of the curve is steeper.

The first graph (Figure 6.20) shows the probability of switching for Foreman video in CIF resolution for SP periodicities of 5, 7 ad 10. For slow bitrates only about a 5 percent of the switching frames are SIs. For big bitrates this value depends extreamly on the SP periodicity. For SP periodicity of 5 this value is 20 percent, 30 when the peridicity is 7 and 40 for a periodicity of 10.

The second graph (Figure 6.21) represents Foreman video in QCIF resolution for SP periodicities of 5 and 7. The values of the probability of switching are rather smaller than for CIF resolution and that is because in CIF resolution frames are bigger. For small bitrates, switching probabilities are around a 1 percent for both curves. However these values diverge more for big bitrates. At those points, the probability value is 5 percent and 7 percent for SP periods of 5 and 7 respectively. Even so, the probability is still pretty low.

The third graph (Figure 6.22) is the probability of switching of Mother and Daughter video in QCIF resolution for SP periodicities of 5, 7, 10, 20 and 50. For small bitrates and SP periodicities of 5, 7 and 10, the probability is about 2 percent. For an SP period of 20 this value increases to 5 percent and for an SP period of 50 the probability is higher than a 20 percent. Even for small bitrates, if the SP period is

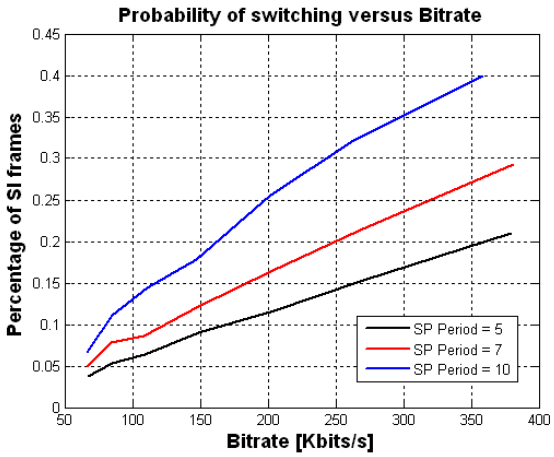


Figure 6.20: Probability of switching. Foreman (CIF).

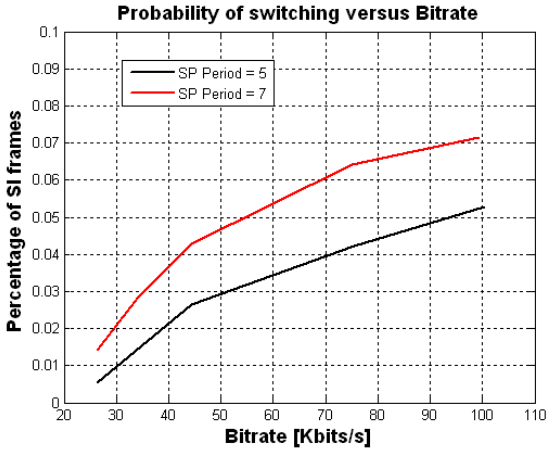


Figure 6.21: Probability of switching. Foreman (QCIF).

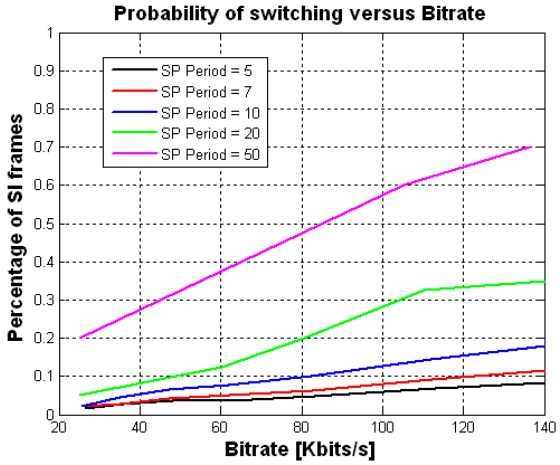


Figure 6.22: Probability of switching. Mother and Daughter (QCIF).

big there is plenty of time between two SP frames and the probability of switching is pretty high. For big bitrates these values increse for all periodicities. In case of an SP period of 50 this value reaches the 70 percent, what means that almost all SP frames are switched to an SI frames. This is because it is extreamly probable to have an error in the time that lastes the transmission of 50 frames.

Chapter 7

Final Comments

This final chapter pretends to be more than a resume of the whole master thesis. In the first section the results obtained in the previous chapter are commented in order to obtain some conclusions about its utility. The resum of the project process is explained later, reminding breefly the different steps carried out during the whole process. The last section talks about different possible works that can be done as a continuation of this master thesis. As well as these future challenges, some personal observations are listed to explain the most important impressions of the author about the whole process and the problems found during its development.

7.1 Conclusions of the simulation results

In the last chapter it has been explained several simulations changing various parameters, depicting some graphs that show the effect that these parameters have on the performance of the switching process. Most of the parameters can be chosen before encoding but the delay of the feedback channel. This value depends on the location of the switching process within the UMTS network but also on its concrete characteristics.

The delay of the feedback channel has a great influence on the performance of the system when sending encoded bitstreams with SP frames inserted. For bitstreams with I frames inserted this is not so because the switching is not implemented. The degradation of the performance decrease with the increase of the delay. The system knows later about the problem in transmission and it has to wait to implement the switching until there is a notification of error. Therefore the smaller the delay, the sooner are SI frames switched and the better are the results.

The best option is to choose a location where the delay would be smaller than 100 ms because that means only one frame after the error has occured the system knows about it. Then it is able to send an SI in the next switching position, stopping error propagation. These small delays are possible when the switching process is located in the Radio Network Controller and are not very probable when it is located in the Media Server. Situating the switching process in the Radio Network Controller implies to send not only the bitstream codified with SP and P frames but also the one with SI and I frames until this point in the UMTS network. This requires to double the bandwidth avalaible.

Regarding the SP periodicity to choose, graphs showed that small SP periods have a better performance than big ones. SP frames have a worse quality than P frames because of the way they are encoded (they are quantized twice). However, this loss of quality is not very marked and the benefits of using an small SP periodicity are really important even for small probability of switching. The smaller the periodicity the sooner the switching would be implemented in case of a problem in transmission and therefore the sooner the error propagation would be stopped. Moreover, the size of SP frames is comparable to the size of P frames, and SI frames, bigger than I frames, are only sent in case of error.

If there is a big delay in the feedback channel, even for small SP periods the system lasts some time to implement the switching. However, it would be less time than in case of a big SP period. When using a bitstream with I frames inserted periodically instead of SP frames, the choice of the GOP depends on the probability of error. Small GOPs are interesting in a situation with a lot of errors in transmission. However that implies a large charge in bitrate since I frames are encoded with much more bits than a P or an SP frame. In any case, if large GOPs are used because the probability of error is small, when it finally occurs the error propagates throughout plenty of pictures until there is an I frame, and the effect of this distortion is not desirable for the viewers.

(a) Frame number 31

(b) Frame number 32

(c) Frame number 39

(d) Frame number 40

Figure 7.1: Error propagation in time.

Figure 7.1 tries to help the reader to see the effect of error propagation. The video used for this figure is Foreman in CIF resolution, encoded inserting I frames with a GOP of 10. The images shown are diferent frames of the reconstructed video in the decoder before error has occured, during its propagation and after the error has been solved. The first one is frame number 31, before the error (that occurs during the transmission of frame number 32, shown in the second image). The third one is the last frame until the error is solved by the corresponding I frame (frame number 39). In the last image, graph 40 is shown. The good quality has been recovered and there are no artifacts in the image. The error has propagated from frame number 32 to frame 40. During 8 frames the image that the viewer saw was distorsioned. In case of having used a SP periodicity of 5 and with a delay lower than 200 ms, an SI frame would have been sent in frame number 35, solving the problem 5 frames earlier.

In order to have a clear idea of whether this method is usuful or not, in the previous chapter some graphs were depicted comparing the efficiency results of the decoded sequence from a bitstream with SP periodicity of 5 and a bitstream with I frames and a GOP of 5. The reason of comparing two sequences with the same periodicity is because both of them have the same capability of stopping error propagation for small delays. The results show an improvement of even two dBs in quality for the same bitrate in Mother and Daughter video between the proposed method using SP frames and the already existing method using I frames. For Foreman video this improvement is a little lower.

As we have mentioned before, the efficiency of the bitstream with I frames improves if a bigger GOP is used. This is because of the low probability of error and it can be very close to the efficiency of using SP frames with a periodicity of 5. This can lead to think that this method is not so interesting. However, the main point is the ability it has to stop an error whenever it occurs after a short number of frames. This method do not have to wait plenty of frames and the bitrate do not increase dramatically, what would not be efficent in case of a long period of time with no errors. For that reason the main advantage of this method is its capacity to adapt itself to the error situation of the channel and therefore it is efficient for both, in case of having plenty of errors and in case of perfect transmission.

This capability is specially important for applications when the video is preencoded before being sent; if for example, the channel is tested and there are not a lot of errors it is possible to choose a longer GOP. If later the situation changes and there are plenty of burst of errors, the quality of the video will be really inadequate for the viewer since it would not be possible to change the GOP later.

On the contrary, in case of choosing a short GOP because it has been considered that the errors in transmission are rather probable, if later there are no errors, there will have been sent plenty of I frames without any necessity. In an SP bitstream there would have not been almost any switching, saving a lot of bandwidth.

In a situation with plenty of errors, specially for big SP periods where there is a long time between two switching frames, there is a great probability of sending an SI. In case of sending almost always SI frames, this method is not so interesting because there is no save of bitrate. However this method is always prepared for a change in the error situation.

7.2 Summary of the project execution

In this section the Master Thesis is summerized to give a complete view of the whole process. Figure 7.2 enumerates the different steps of the project that are breefly described next.

In the beginning my knowledges about video coding were rather poor and therefore, the first step consisted in reading a plenty of papers about this topic. The first ones talked about the history of video coding, the different existing standards and how they worked, remarking the differences with the new standard H.264/AVC. The papers consulted later were about a new strategy introduced by H.264/AVC, SP and SI frames. Since this project works with this two frame types, the information read about the different encoding process of this frames regarding P and I frames was deeply studied. In order to introduce myself in context, also some papers about UMTS networks were consulted. They helped to understand the packetization process, information that was used in the program that implements the switching process to simulate errors in transmission and to activate the switching.

After the documentation step, the effect of the different quantization parameters of SP and SI frames was studied. The different way of encoding these frames introduced two new quantization parameters, the so-called PQP and SPQP. The different values of these parameters affected the size and the distortion not only of the SP and the SI frames but also of the following P frames because of using temporal prediction. This obligues them to refer to previous pictures to calculate its predicted image.

SP and SI frames are based on the fact that both of them offer a perfect reconstruction of a frame, that means that the reconstructed image is exactly the same in an SP and an SI frame. This is possible if every frame is packetized in only one RTP packet. However, the situation to evaluate is with RTP packets of 700 bytes and then the system crashed in these new frame types. To solve the problem the encoder code was modified to allow packetization in 700 bytes only for I and P frames and not for SP and SI, allowing a perfect switching between them.

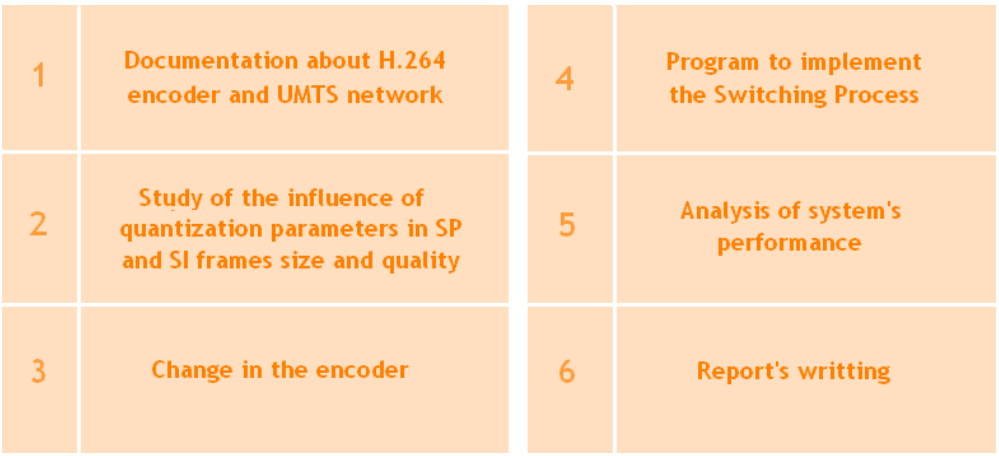


Figure 7.2: Steps of the Master Thesis's development.

Then, a set of routines were programmed to implement the switching process. The system has as inputs a sequence encoded twice, one with SP frames inserted and the other with SI frames; a file with the error information that will be used to simulate errors in transmission and the data introduced by the user about the delay of the feedback channel (that informs the system about the errors occured). The error file corresponds to the information meassured from a wireless link in a real UMTS network. In a real system the errors in transmission and the delay would no be simmulated. As soon as the system knows about the error the switching would be implemented but only in a switching frame.

The fith step consisted in simulating different situations to study the behaviour of this method in comparison to other using the already existing frames. The objective was to obtain a conclusion about which of these two methods is more efficient to stop error propagation while decoding. The different situations simulated consisted in varying the SP periodicity, the GOP, the delay of the feedback channel and the bandwidth avalaible in the channel. The different bandwidth resulted in the encoded bitstream was controlled by the quantization parameter.

After obtaining a conclusion that is explained in the previous section, the last step was to write this report to explain the whole Master Thesis execution, describing deeply all the parts in which it was divided.

7.3 Problems found and future challenges

This last section of the Master Thesis report explains the main problems found during the project execution as well as possible challenges for those who want to follow investigating in this field.

The first problem I found was the vast documentation needed to start this project because of my lack of knowledges in this field. Because the topic is so specific, the papers that explained the concepts about H.264/AVC were normally rather complex, specially for someone who started from very little information about the topic. Moreover, since the project also requires to have some knowledges about RTP and TB packets, it was also needed to documentate about UMTS networks.

The main problem of this work was the fact that the H.264/AVC was not prepared to packetize SP and SI frames into RTP packets of a fixed size. This problem was detected by the time of analysing the results obtained in the simulations performed to observe the behaviour of the switching process. By the time of sending an SI frame to avoid error propagation after an error in transmission the good quality was recovered. However, the frames that followed it had again bad quality because the reconstructed image of an SP and an SI frame were not identical. Since they were used as a reference for next P frames, their predicted image was not the same and this fact introduced distortion in the reconstructed image.

After noticing the error, plenty of time had already been used for simulating and analysing the results. The encoder was modified but it was needed to simulate again not only to obtain the results about the behaviour of the switching process but also the analysis about the influence of the different quantization parameters of SP and SI frames on its size and quality was repeared.

The last important aspect to remark about the project execution is the long time needed for simulating. In order to have realistic results, several situations of the transmission channel were simmulated for every encoded sequence and later the values obtained were averaged. Moreover, the fact of analyse different periodicities, delays and bitrates increased in an important way the number of simmulations performed. Consecuently the files resulting from this simulations were very numerous and the computers had usually problems to work with folders with so many files.

This last problem is one of the challenges I find more important to overcome. The encoder code is not optimized because this standard is the newest one and is not completely developed. The different improves have been added without worring about the programming aspect and now its execution is so slow that is not interesting for a real time application like video transmission for mobiles.

The encoder code have to be modified also to allow SP and SI frames to be packetized into RTP packets with a fixed size. The solution used in this project was to packetize I and P frames in RTP packets of 700 bytes but SP and SI frames formed an only RTP packet. The challenge is to prepare the encoder to be able to implement this packetization, obtaining identical reconstructed images for both.

Another possible continuation can be implement this system in a real UMTS network, modifying the routines that implements the switching process since the errors and the delay are not simmulated in a real transmission. The system can be tested and the results anylised to obtain a final conclusion about the viability of this method in comparison with other previously existing.

To finish I want to thank the TU university for allowing me developing a Master Thesis in such an interesting field of research. This have been a very important oportunity for me to learn how to perform a serious investigation and to improve my oral presentations. I am grateful for the support of my two supervisors, Luca Superiori and Wolfgang Karner who have helped me always I needed, in the documentation step and also in the execution of the thesis even when they had plenty of important tasks to do. In my last sentence I want to record the incredible experience that my stay in Vienna has been for me, and all the great moments lived in this beatiful city.

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APPENDIX A: Optimization of PQP and SPQP values

The effect in rate and distortion of the different quantization parameters is studied in this appendix. General conclusions are that PQP influence size of SP and quality of SP and SI frames whereas SPQP afects size of SI and quality of SP and SI frames. Figure 7.3 [17] depicts how the increase of SPQP parameter decrease the size of SI frames and increase the distortion of SP frames.

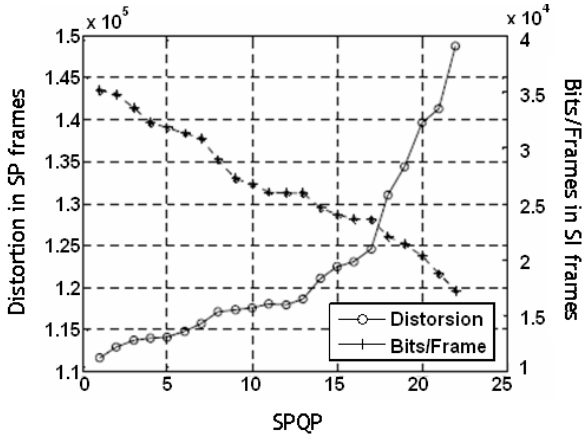

Figure 7.3: Distortion of SP frames and size of SI frames depending on SPQP

These conclusions explains the behaviour, however optimized values were seek for this project. The choice of the values of these parameters have followed the recommendation described in [18] and resumed next.

In the considered scenario, SP frame positions are spaced regularly in the transmitted video stream and SI frames can be sent in these positions in case of packet losses to stop error propagation. The proportion of SI frames transmitted instead of SP frames depends on the packet error rate and on the SP period. The probability of transmitting an SI frame at an SP frame position is denoted as x. In this optimization, the quality is considered as a constraint and the bit-rate is minimized what is equivalent to minimizing the expected size of a frame sent at an SP position:

$$\text{Min. } \mathcal{R} = xR_{SI} + (1-x)R_{SP_1} \tag{7.1}$$

$$\text{s.t. } D_{SP_1} = D_{SI} = D \tag{7.2}$$

where R_{SI}, R_{SP_1}, D_{SI} and D_{SP_1} denotes the rate and distortion of SI and primary SP frames respectively. The model used to estimate this values assumes that the image signal s and the error signal e are stationary and jointly Gaussian zero-mean signals and that the intra-frame encoders perform at their theoretical rate-distortion optimum.

The expresion for the rate of primary SP frames used is:

$$R_{SP_1} = \frac{1}{8\pi^2} \iint_{\Lambda} max(0, log_2(\frac{\Phi_{ee}(\Lambda)}{\theta_1}))d\Lambda \tag{7.3}$$

where Φ_{ee} is the spatial power spectral density (PSD) of the prediction error e before being quantized and $\Lambda = (\omega_x,\omega_y)$ is a vector representing spatial frequency.

The expresion for the rate of SI frames used is:

$$R_{SI} = \frac{1}{8\pi^2} \iint_{\Lambda} max(0, log_2(\frac{\Phi_{ss}(\Lambda)}{\theta_2}))d\Lambda \tag{7.4}$$

where Φ_{ss} is the PSD of the image signal s. In H.264 θ_1 and θ_2 are the two quantization parameters SPQP and PQP. Since the intra prediction is assumed in the optimum, the signal l_{rec} is considered transmitted in SI frames. This signal is the compressed version of the reconstruction of the primary SP frame. At high rates, the PSD of this signal is considered to be Gaussian and equal to Φ_{ss}.

To obtain the expression of the distortion of primary SP frames it has been assumed again that at high rates the PSD of the reconstructed image is close to that of the original signal s. Another assumption taken into consideration is that the distortion introduced by the second quantization (SPQP) can be added directly to the distortion of the first quantization (PQP). The mean square error distortion for the primary SP is then considered as:

$$D_{SP_1} = D_1 + D_2 \tag{7.5}$$

$$D_1 = \frac{1}{4\pi^2} \iint_{\Lambda} min(\theta_1, \Phi_{ee}(\Lambda))d\Lambda \tag{7.6}$$

$$D_2 = \frac{1}{4\pi^2} \iint_{\Lambda} min(\theta_2, \Phi_{ss}(\Lambda))d\Lambda \tag{7.7}$$

Since the reconstruction of an image encoded as an SI frame and as an SP frame are identical, their distortions are also equal:

$$D_{SI} = D_{SP_1} \tag{7.8}$$

With this expresions for rate and distortion at high rates ($\Phi_{ee}(\Lambda) \gg \theta_1$ and $\Phi_{ss}(\Lambda) \gg \theta_2$) and the equalty constraint, that sets the quality of SP and SI frames equal to the mean quality of the whole encoded stream, (3.1) and (3.2) are reduced to:

$$\text{Min. } (x-1)log(\theta_1) - xlog(\theta_2) \tag{7.9}$$

$$\text{s.t. } \theta_1 + \theta_2 = D \tag{7.10}$$

from where a relation between the optimized parameters is obtained:

$$\theta_1^* = \frac{x}{1-x}\theta_2^* \tag{7.11}$$

For any Gaussian signal with a continuous PSD, at high frequencies the slope of the rate-distiortion function is expressed:

$$\frac{dR}{dD} = \frac{-1}{2log(2)\Theta} \tag{7.12}$$

where Θ is in H.264 the quantization parameter that generates the rate-distortion curve. From this equation it is possible to obtain the slope of the distortion-rate functions of the error signal emphe and the reconstructed signal in an SP before being quantized by SPQP $\tilde{s}$:

$$\frac{dR_e}{dD_e} = -0.85 * 2^{\frac{PQP-12}{3}} \tag{7.13}$$

$$\frac{dR_{\tilde{s}}}{dD_{\tilde{s}}} = -0.85 * 2^{\frac{SPQP-12}{3}} \tag{7.14}$$

The optimal values of $SPQP^*$ and PQP^* are resulted by combining (3.11),(3.13) and (3.14):

$$SPQP^* = PQP^* + 3log_2(\frac{x}{1-x}) \tag{7.15}$$

Since an assumption is to have the same qualtiy in SP frames as in other frames of the video stream, $SPQP^*$ and PQP^* can not exceed the quantization parameter QP used to encode the P frames. The last restriction is that the three quantization parameters must be integers. With all these constraints and equations the only possible optimum values are shown in Table 7.1 , as well as the optimal values derived empirically.

x model	≤ 0.2	≥ 0.2 and ≤ 0.5	≥ 0.5
x empirical	≤ 0.1	≥ 0.1 and ≤ 0.2	≥ 0.2
PQP	QP - 1	QP - 2	QP - 3
$SPQP$	QP - 10	QP - 5	QP

Table 7.1: Optimal values for PQP and SPQP

APPENDIX B: Quantization parameter analysis

Results from the analysis of the influence of the different quantization parameters (QP, PQP and SPQP) from FOREMAN video with QCIF resolution.

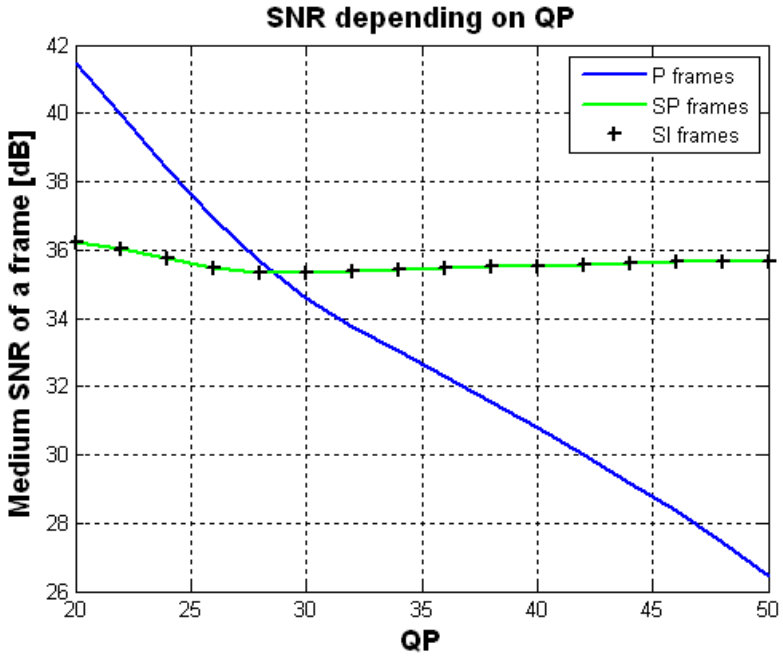

Figure 7.4: SNR(QP)with PQP = 26 and SPQP = 26

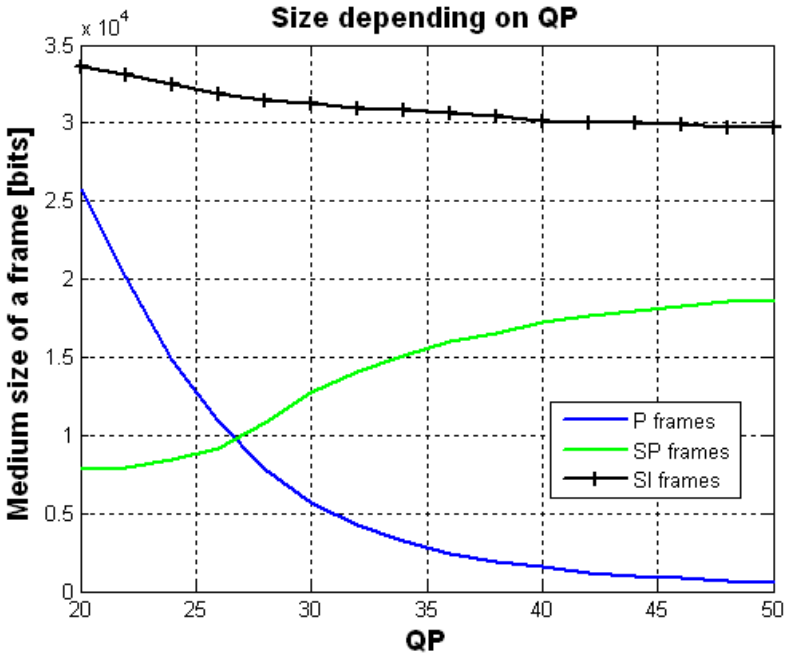

Figure 7.5: Size(QP)with PQP = 26 and SPQP = 26

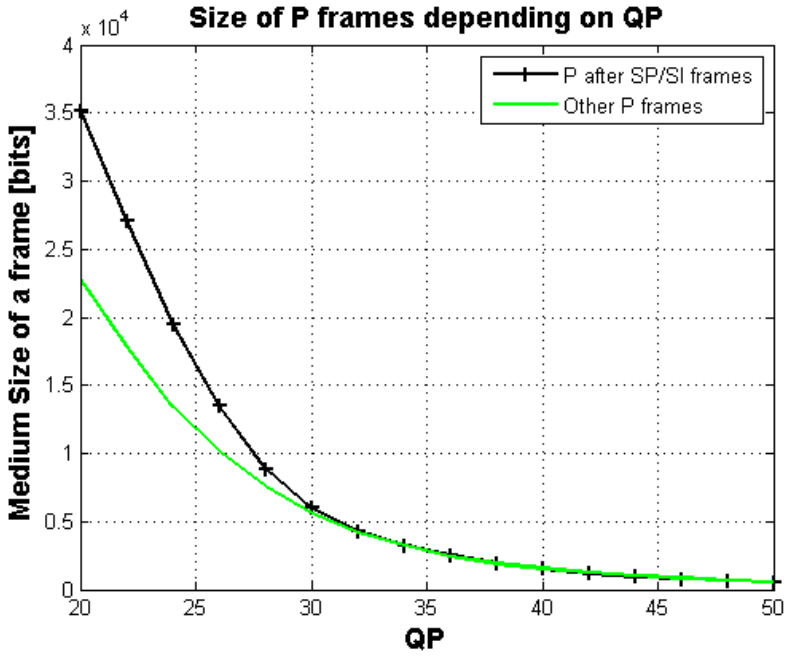

Figure 7.6: SNR(QP)with PQP = 26 and SPQP = 26

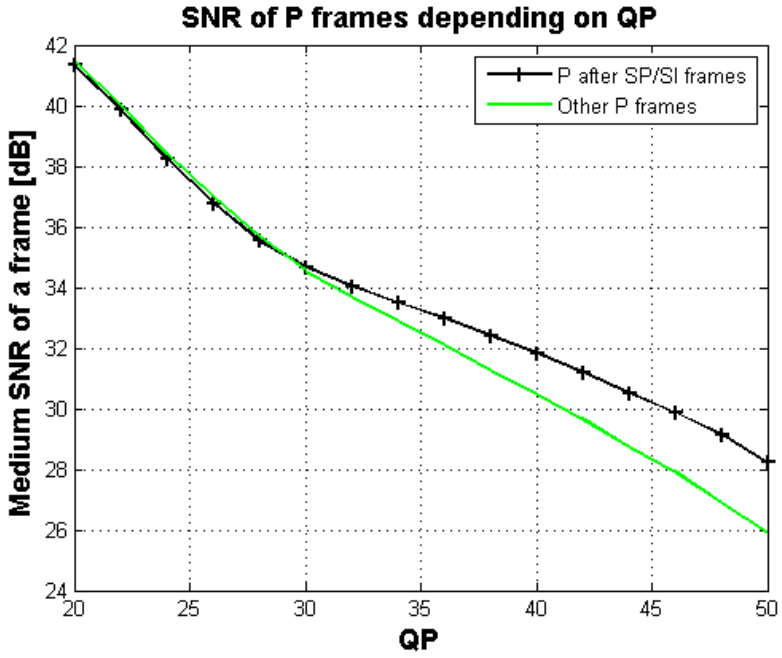

Figure 7.7: Size(QP) with PQP = 26 and SPQP = 26

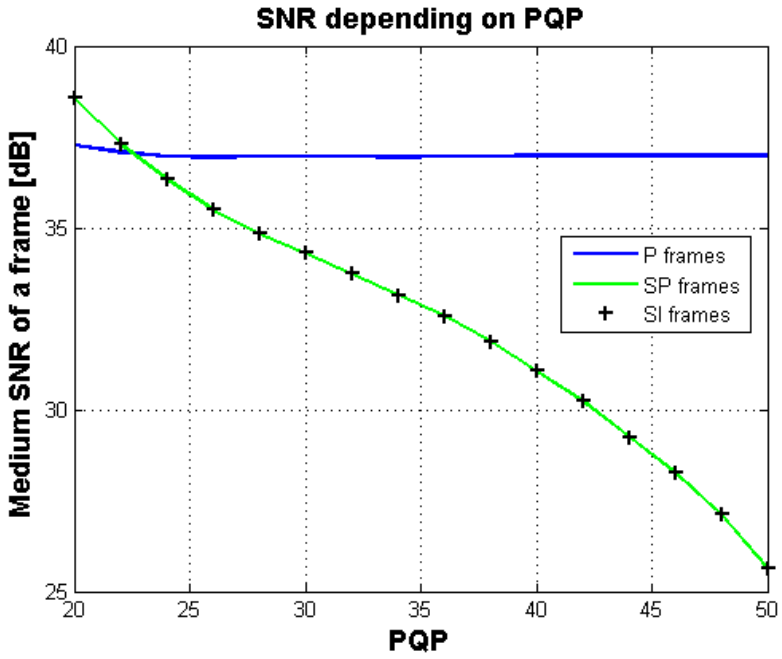

Figure 7.8: SNR(PQP) with QP = 26 and SPQP = 26

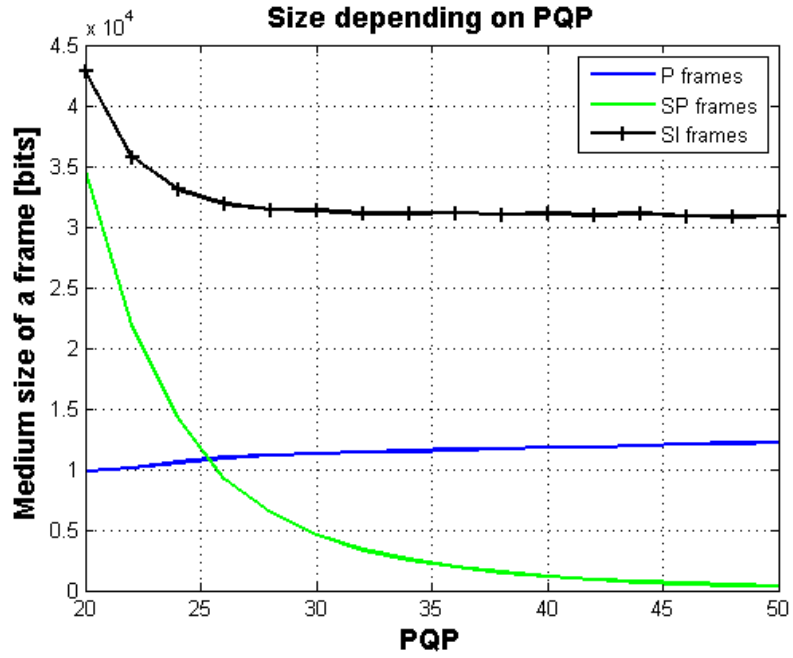

Figure 7.9: Size(PQP) with QP = 26 and SPQP = 26

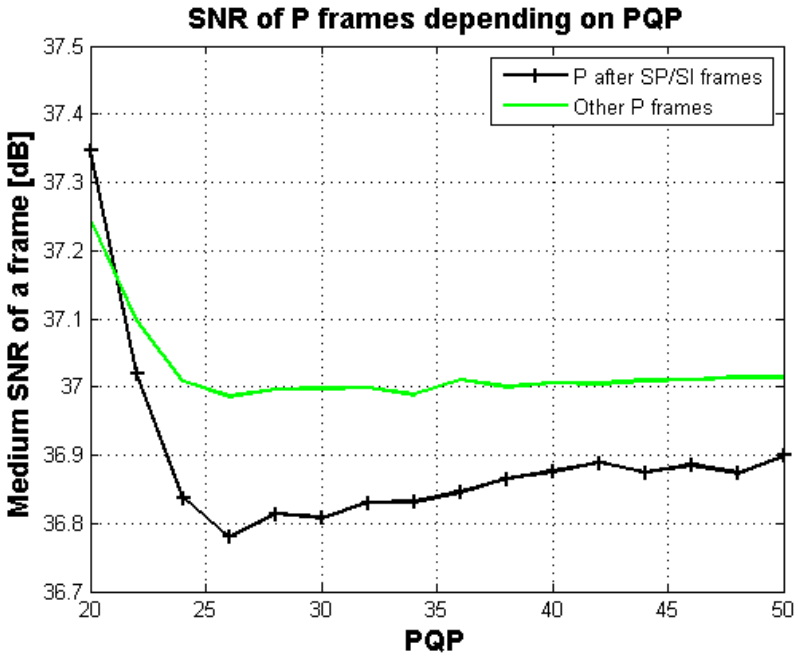

Figure 7.10: SNR(PQP) with QP = 26 and SPQP = 26

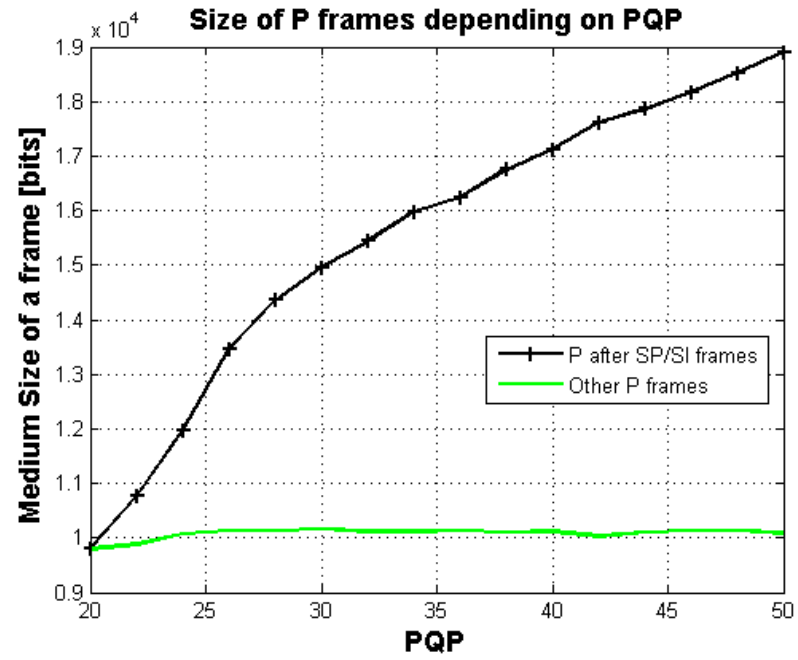

Figure 7.11: Size(PQP) with QP = 26 and SPQP = 26

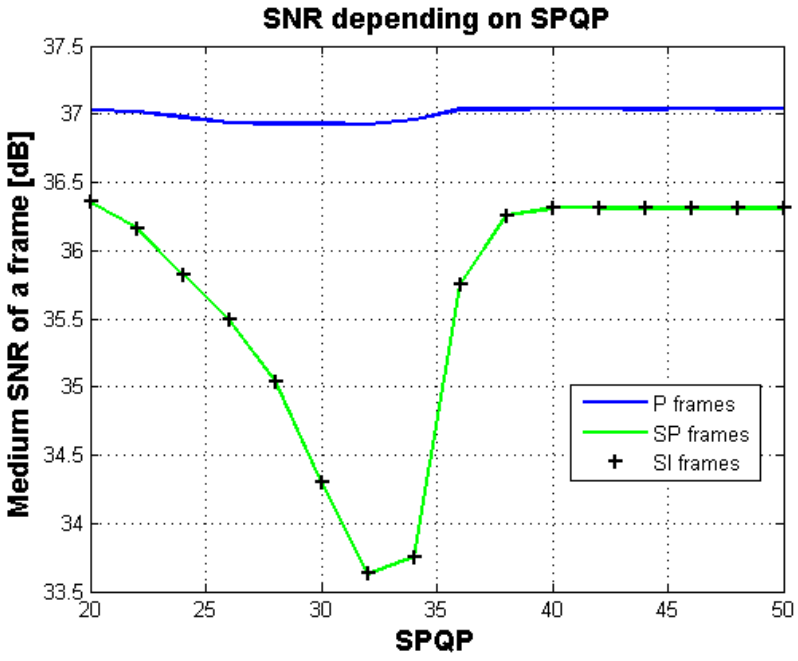

Figure 7.12: SNR(SPQP) with QP = 26 and PQP = 26

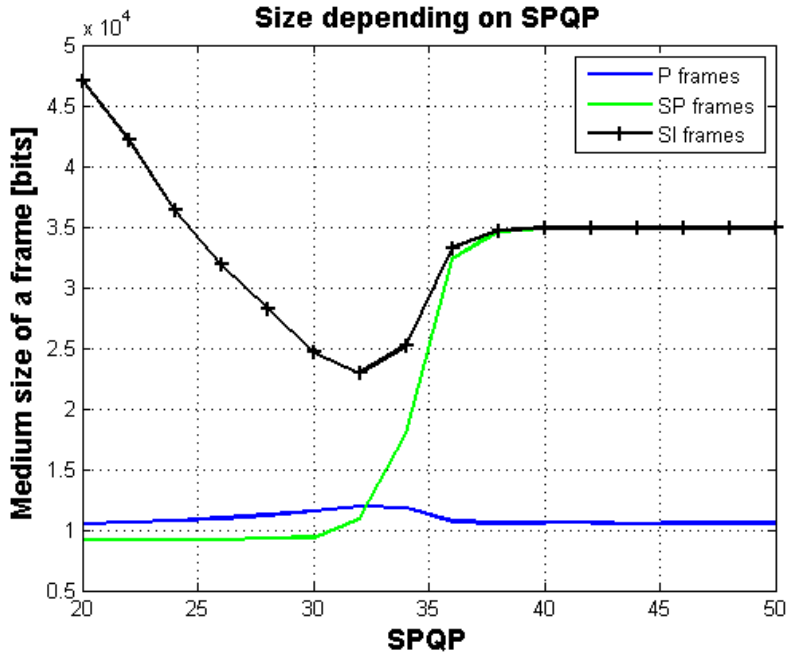

Figure 7.13: Size(SPQP) with QP = 26 and PQP = 26

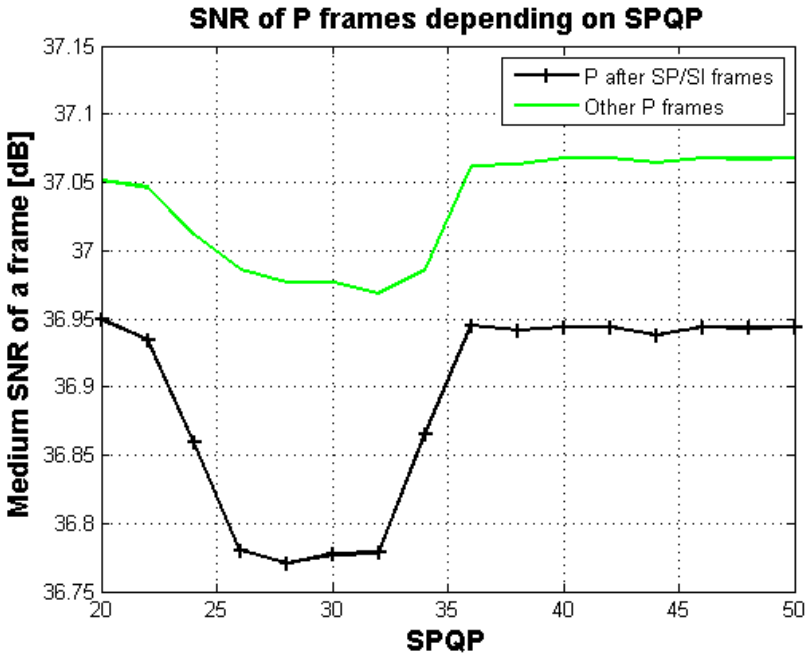

Figure 7.14: SNR(SPQP) with QP = 26 and PQP = 26

Figure 7.15: Size(SPQP) with QP = 26 and PQP = 26

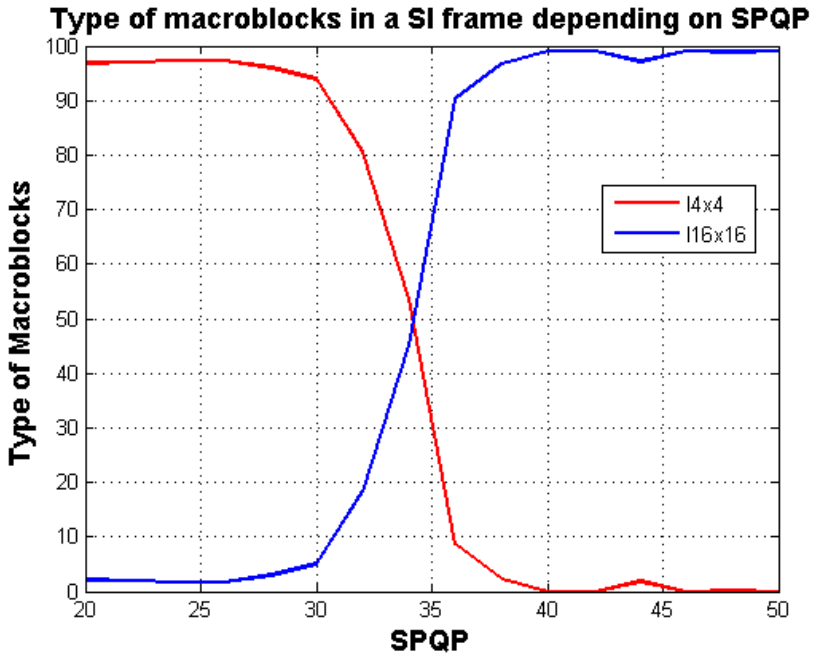

Figure 7.16: Type of macroblocks in an SP frame depending on SPQP

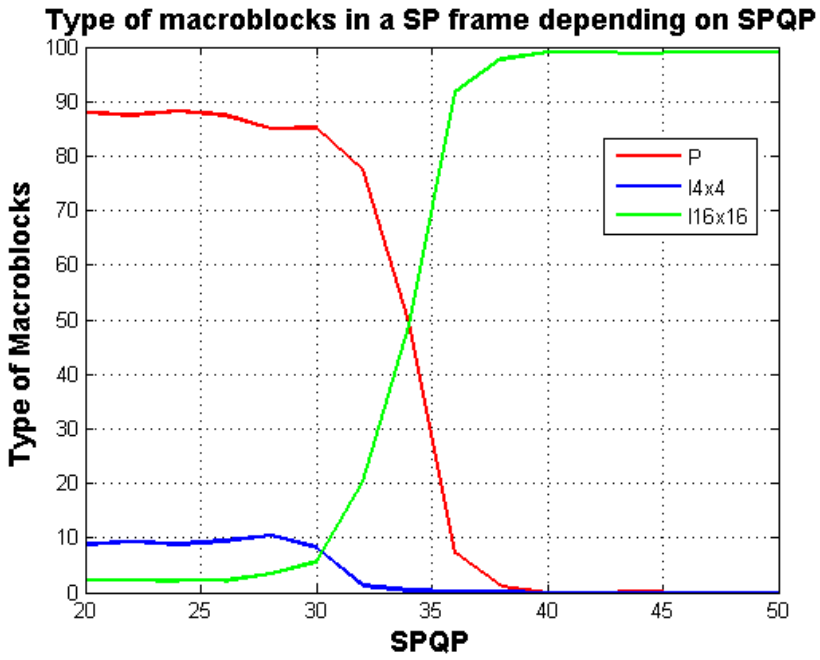

Figure 7.17: Type of macroblocks in an SI frame depending on SPQP

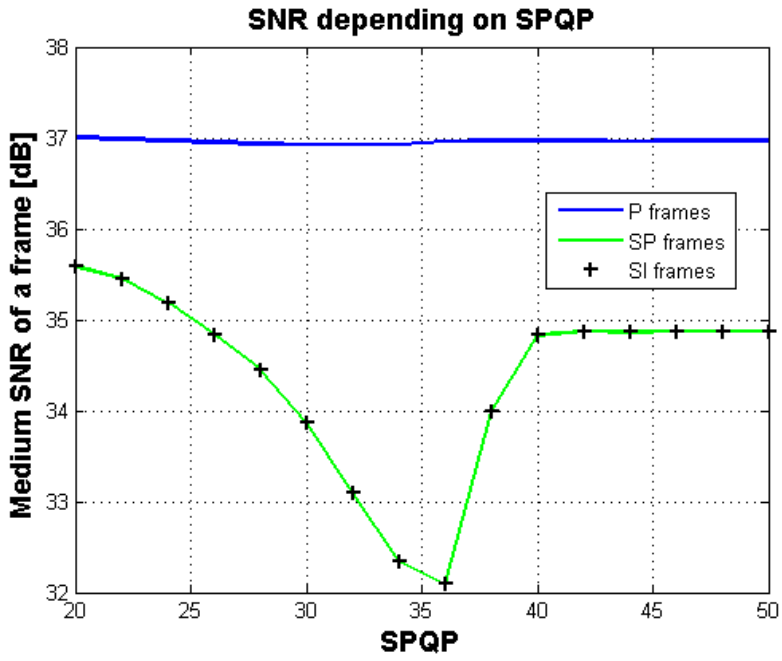

Figure 7.18: SNR(SPQP) with QP = 26 and PQP = 28

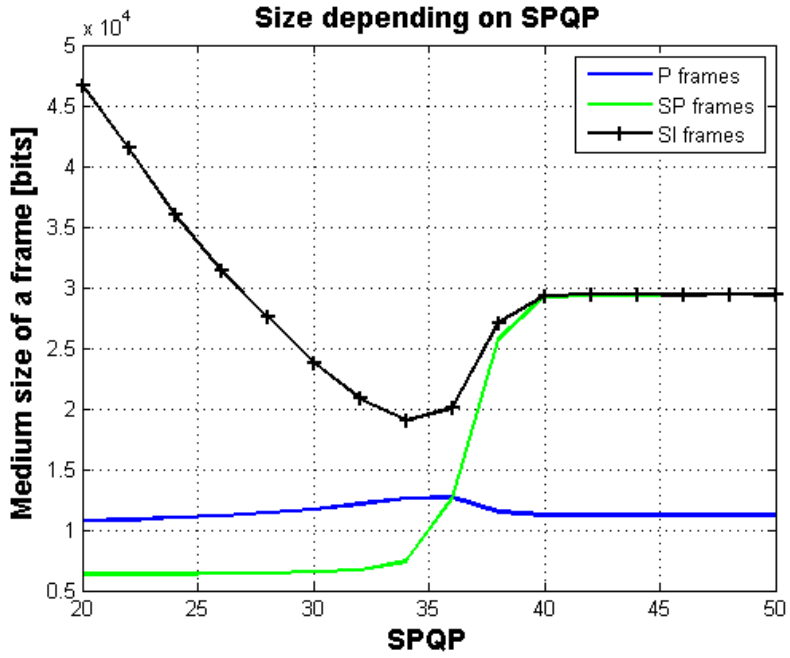

Figure 7.19: Size(SPQP) with QP = 26 and PQP = 28

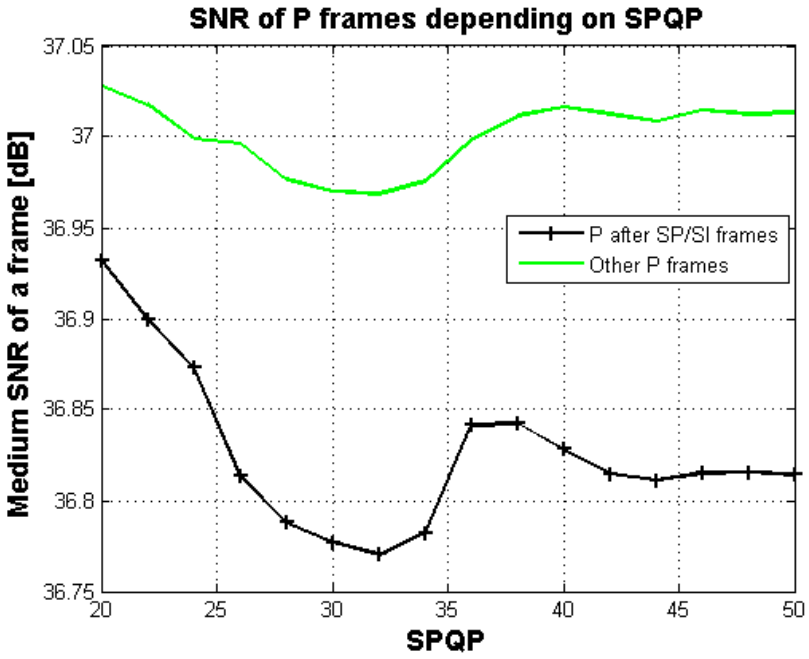

Figure 7.20: SNR(SPQP) with QP = 26 and PQP = 28

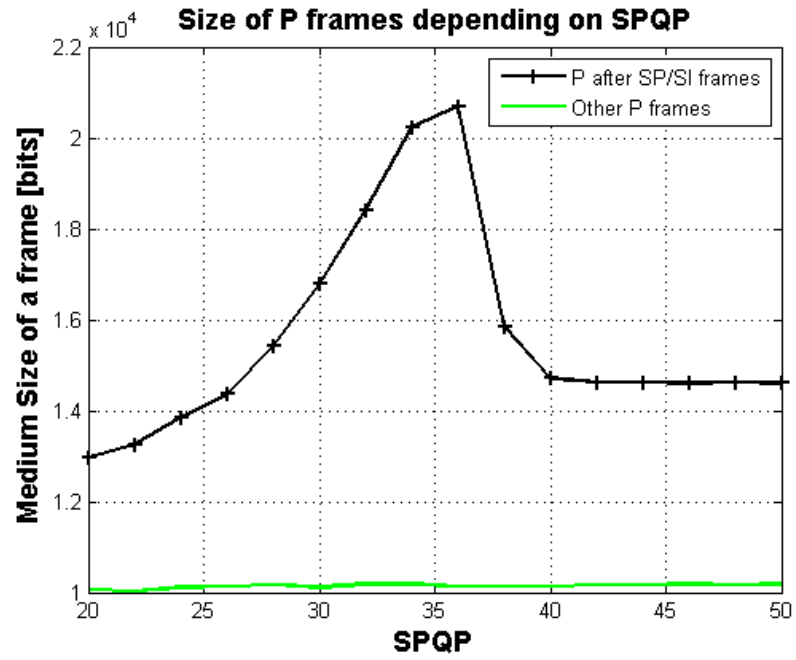

Figure 7.21: Size(SPQP) with QP = 26 and PQP = 28

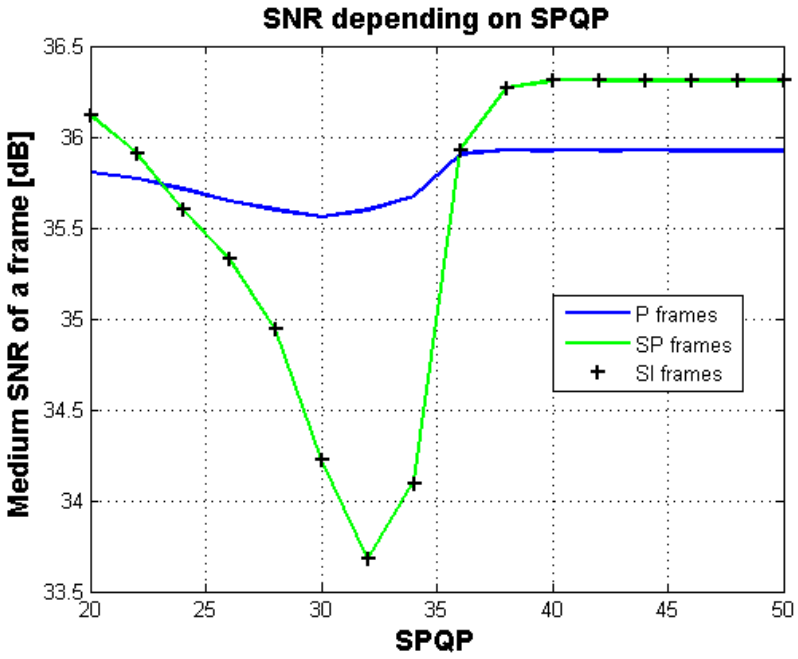

Figure 7.22: SNR(SPQP) with QP = 28 and PQP = 26

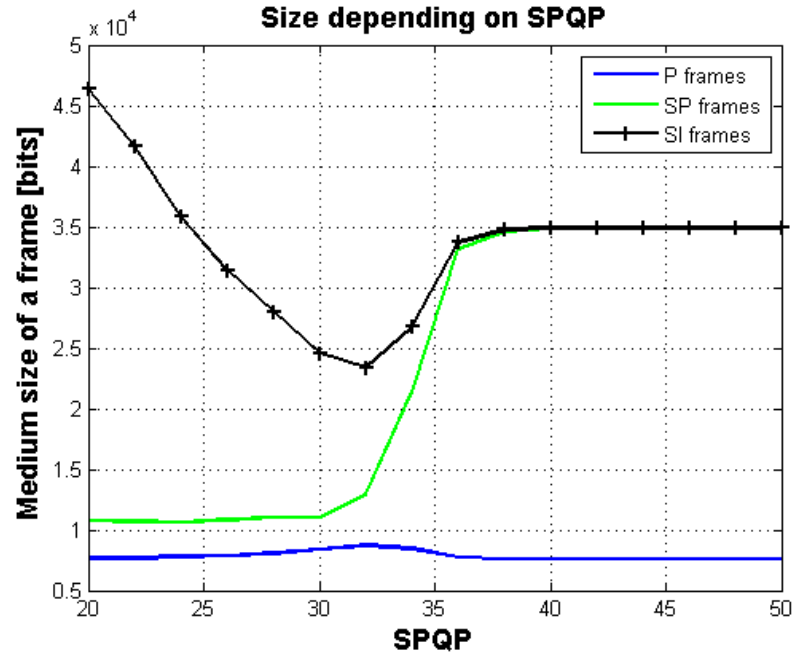

Figure 7.23: Size(SPQP) with QP = 28 and PQP = 26

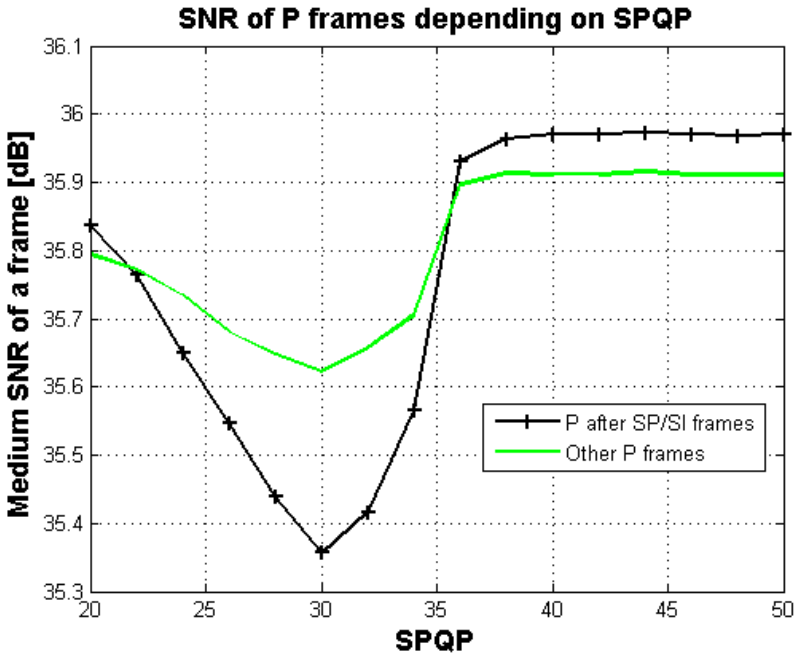

Figure 7.24: SNR(SPQP) with QP = 28 and PQP = 26

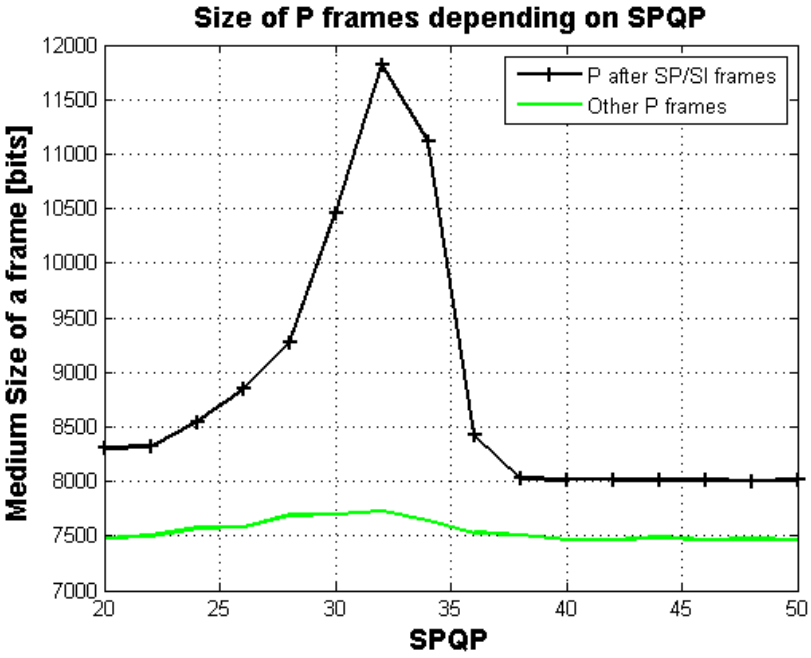

Figure 7.25: Size(SPQP)with QP = 28 and PQP = 26

APPENDIX C: Switching Process analysis

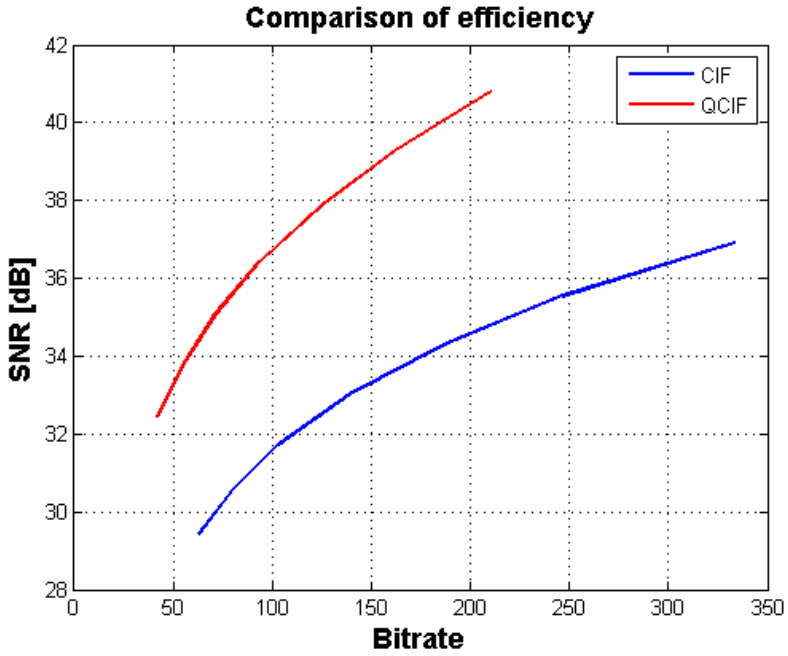

Figure 7.26: Comparison of CIF and QCIF resolution in Foreman video.

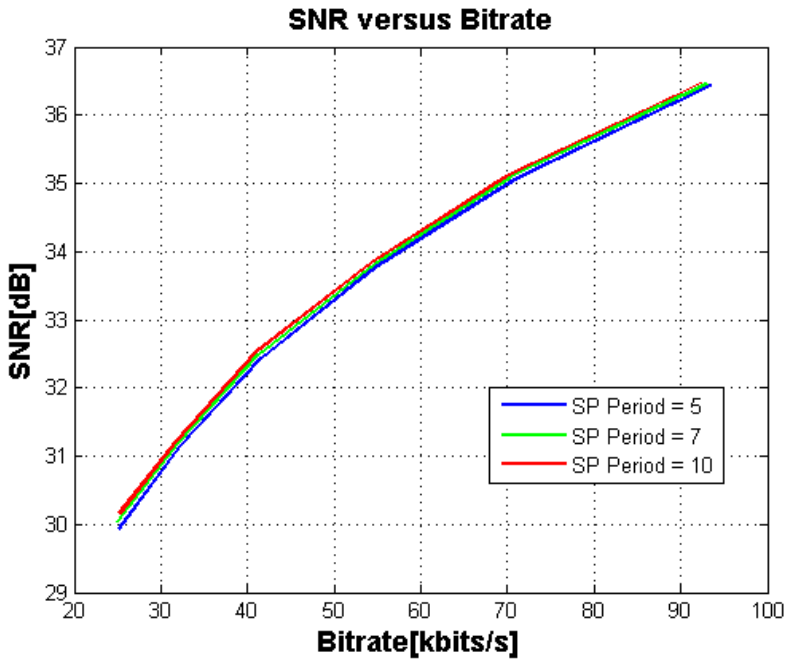

Figure 7.27: Foreman QCIF - Encoding - Performance of different SP periods.

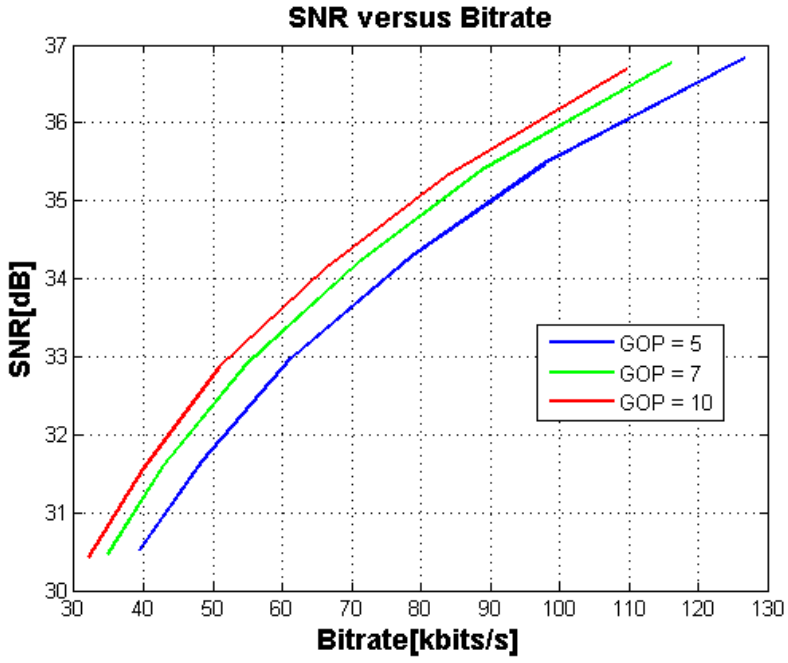

Figure 7.28: Foreman QCIF - Encoding - Performance of different GOPs.

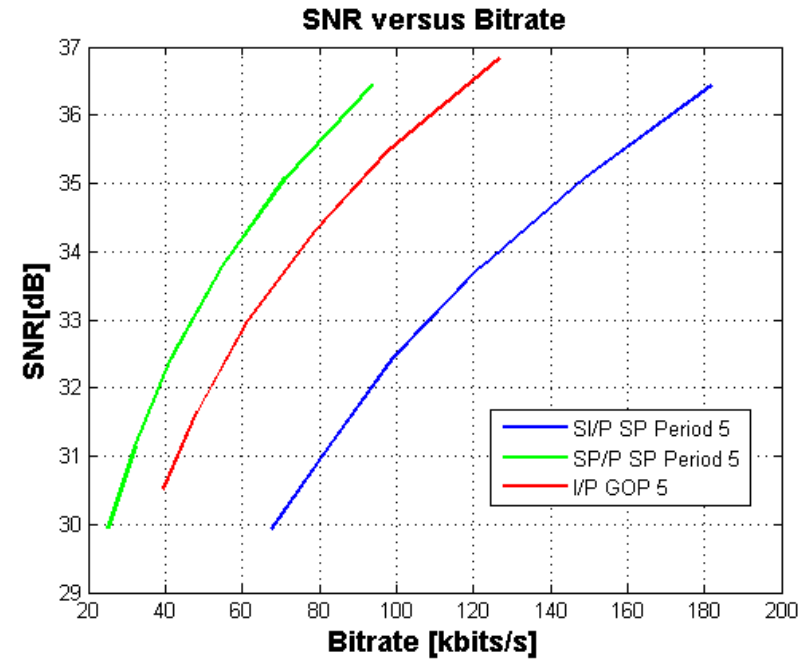

Figure 7.29: Foreman QCIF - Encoding - Comparison of performance.

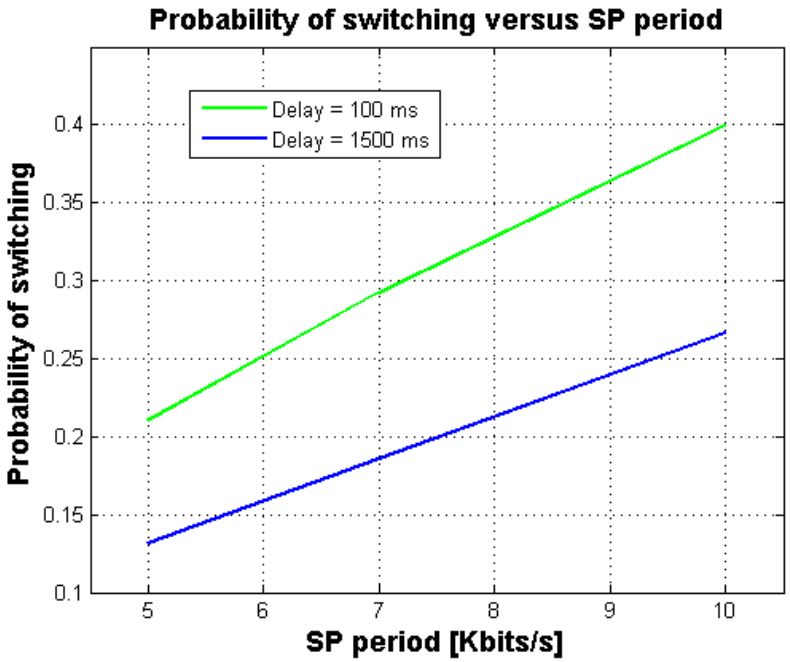

Figure 7.30: Foreman CIF - Decoding - SP Period equal to 5.

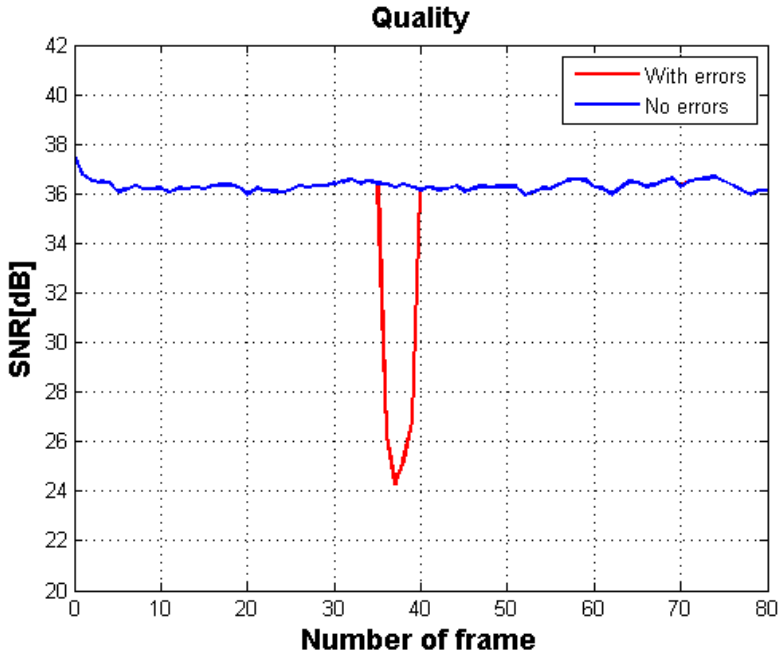

Figure 7.31: Foreman QCIF - Decoding - Feedback channel delay 100 ms.

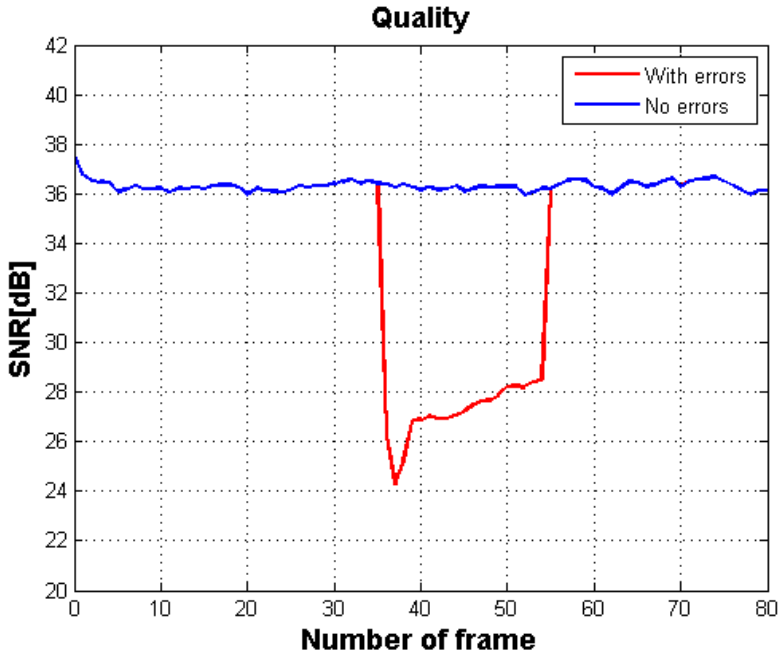

Figure 7.32: Foreman QCIF - Decoding - Feedback channel delay 1500 ms.

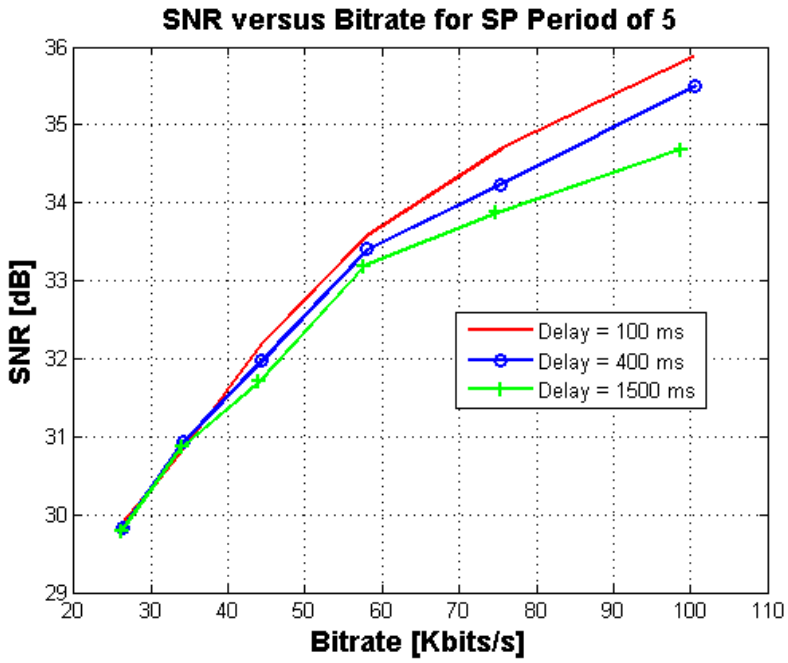

Figure 7.33: Foreman QCIF - Decoding - SP Period equal to 5, different delays.

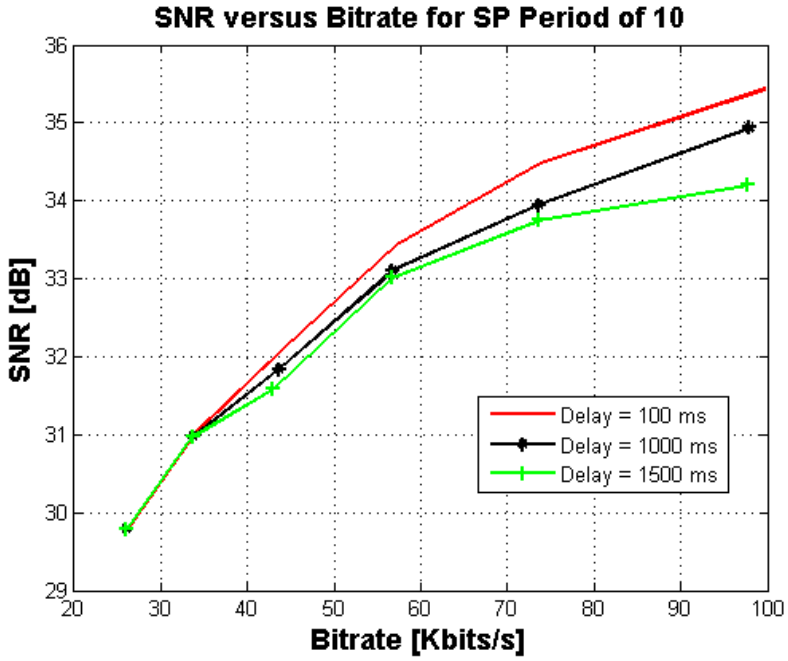

Figure 7.34: Foreman QCIF - Decoding - SP Period equal to 10, different delays.

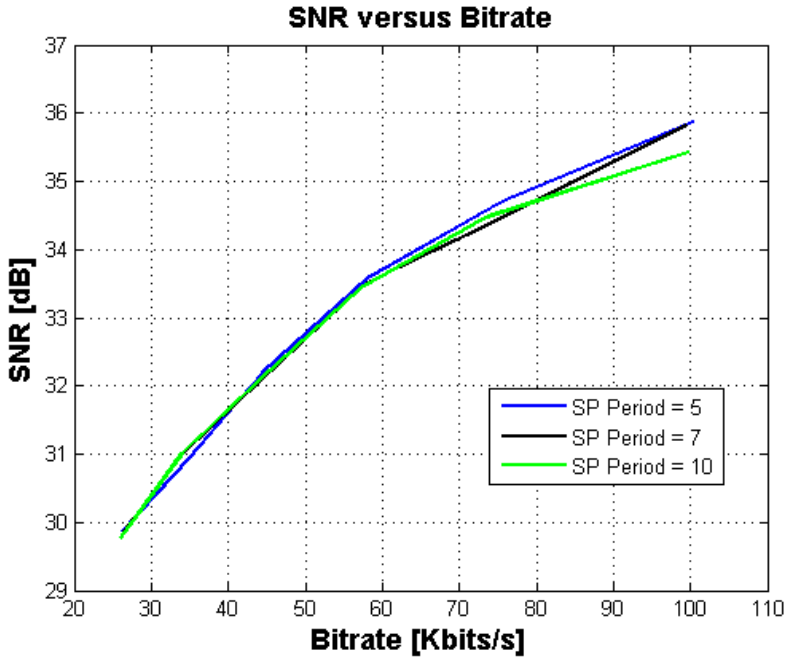

Figure 7.35: Foreman CIF - Decoding - Delay 100 ms, Performance of different SP Periods.

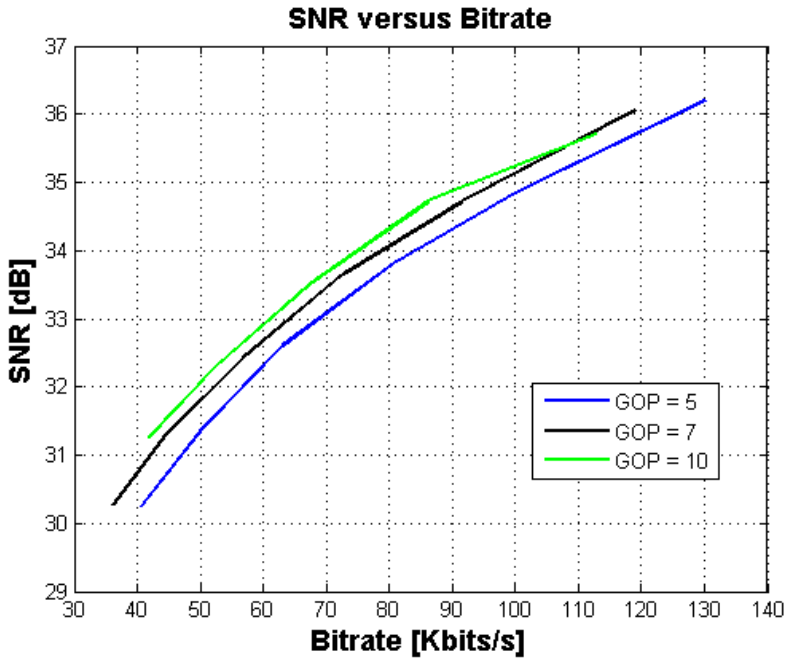

Figure 7.36: Foreman CIF - Decoding - Performance of different GOPs.

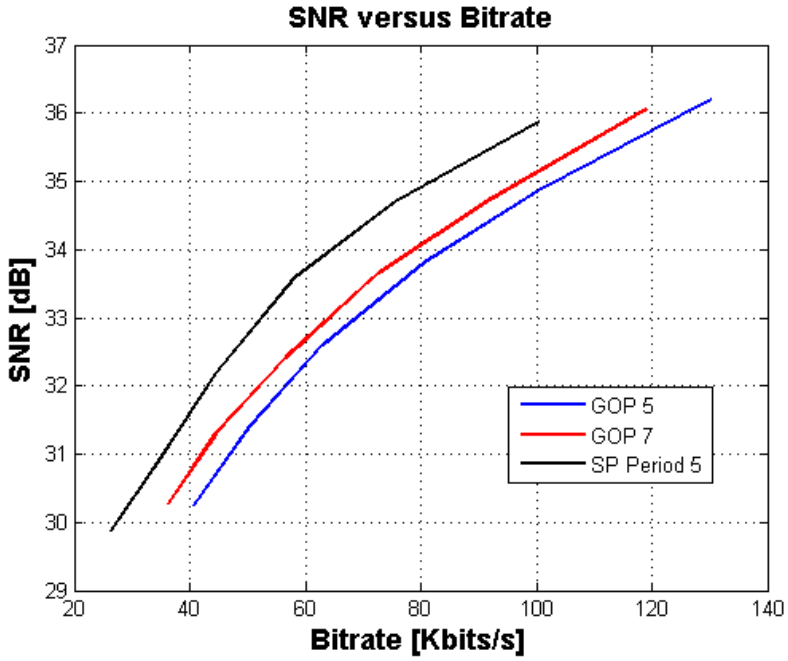

Figure 7.37: Foreman CIF - Decoding - Delay 100 ms, Performance SP/P and I/P.

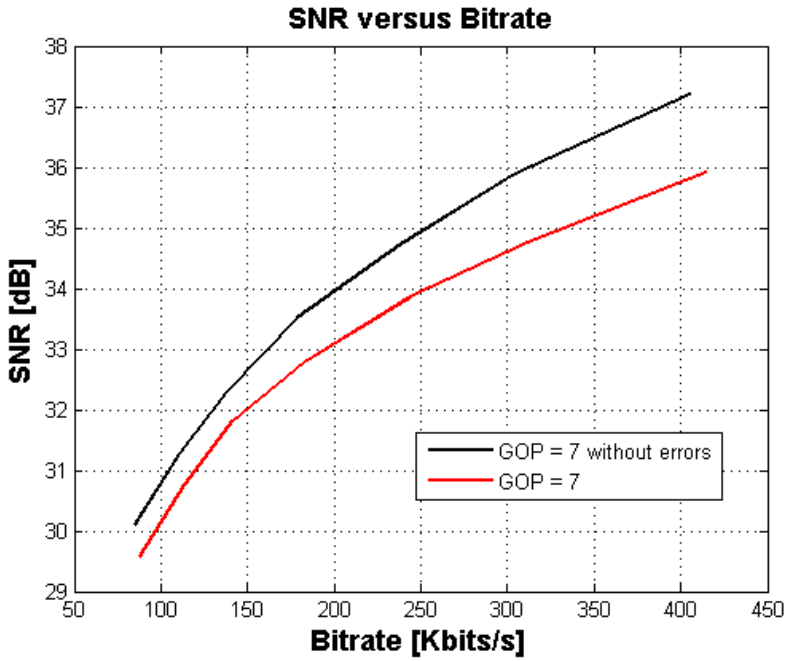

Figure 7.38: Foreman CIF - GOP 7, quality degradation in transmission.

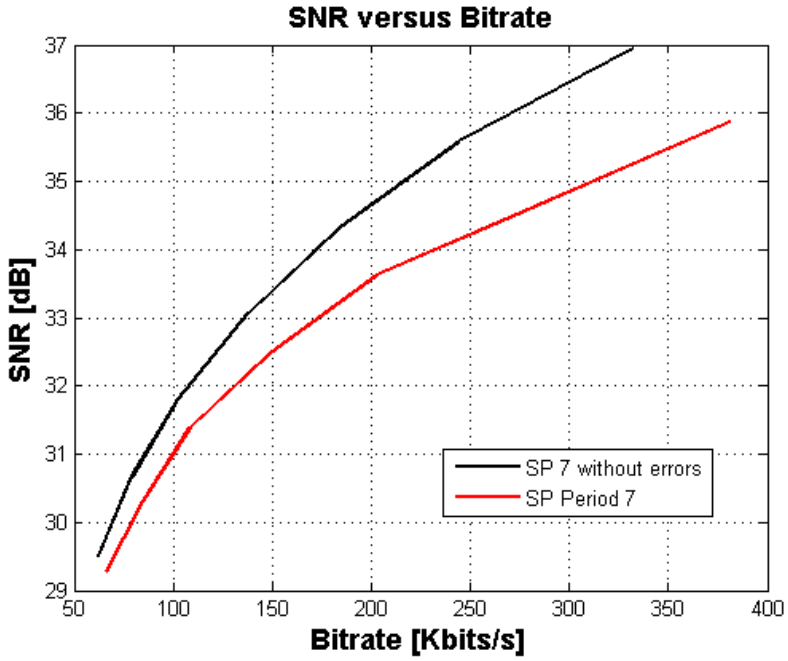

Figure 7.39: Foreman CIF - SP Period 7, delay 100 ms, quality degradation in transmission.

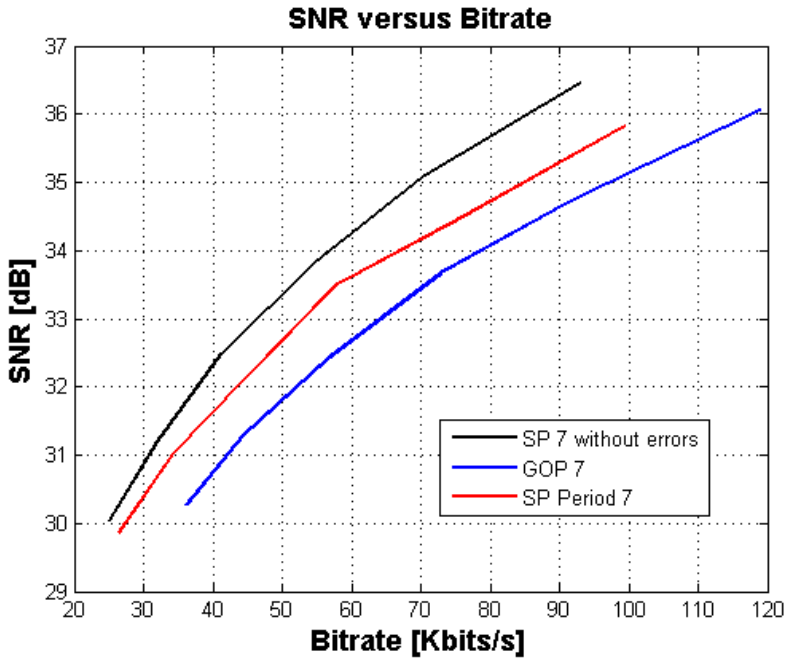

Figure 7.40: Foreman QCIF - Period 7, Efficiency comparison.

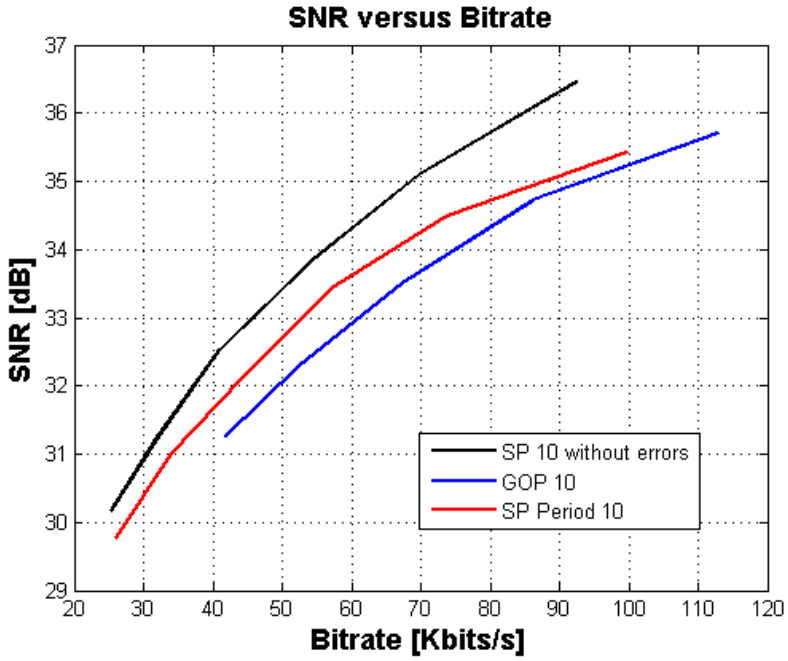

Figure 7.41: Foreman QCIF - Period 10, Efficiency comparison.

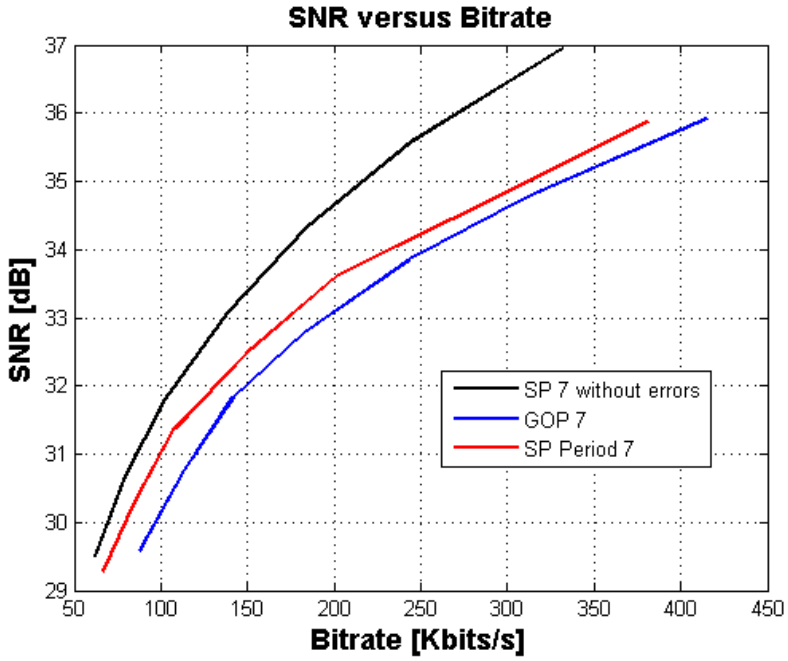

Figure 7.42: Mother and Daughter CIF - Period 7, Efficiency comparison.

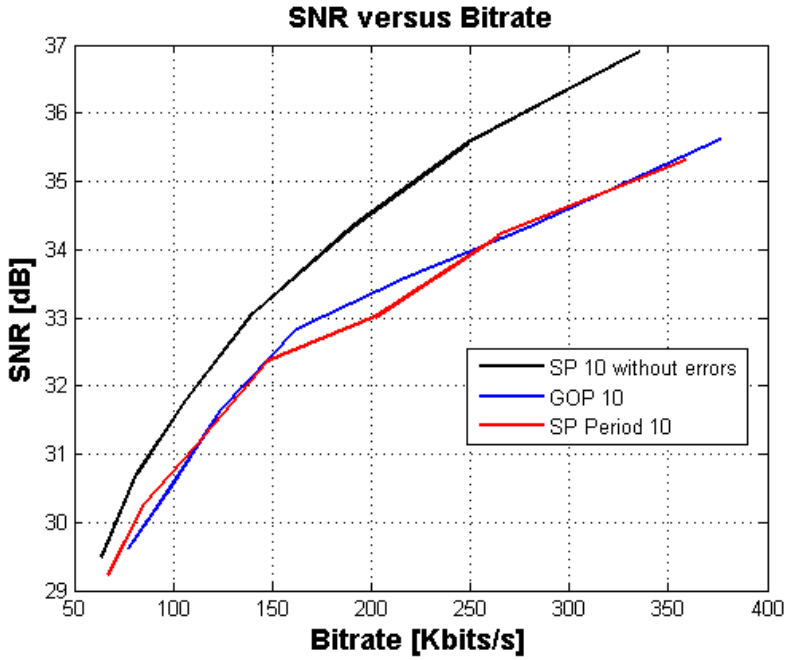

Figure 7.43: Mother and Daughter CIF - Period 10, Efficiency comparison.

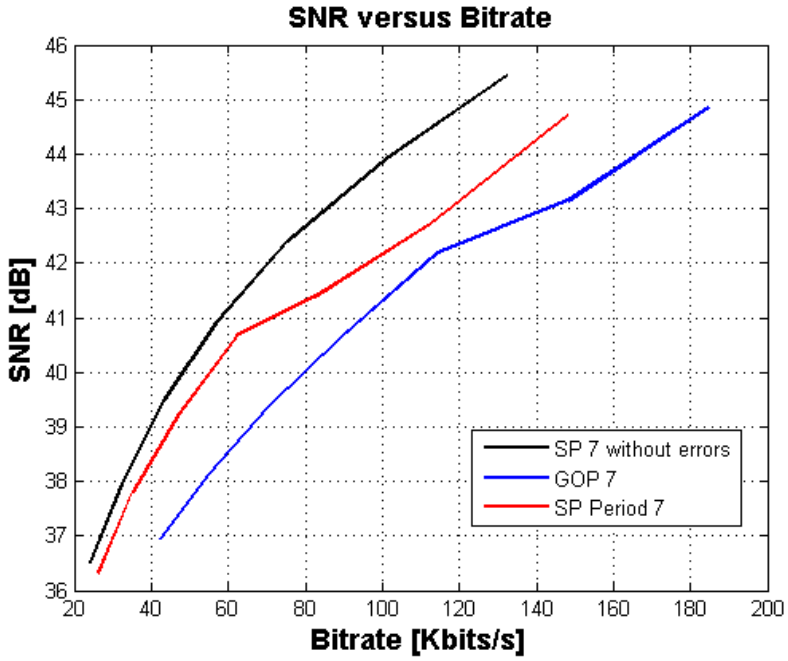

Figure 7.44: Foreman CIF - SP Period 7, delay 100 ms, quality degradation in transmission.

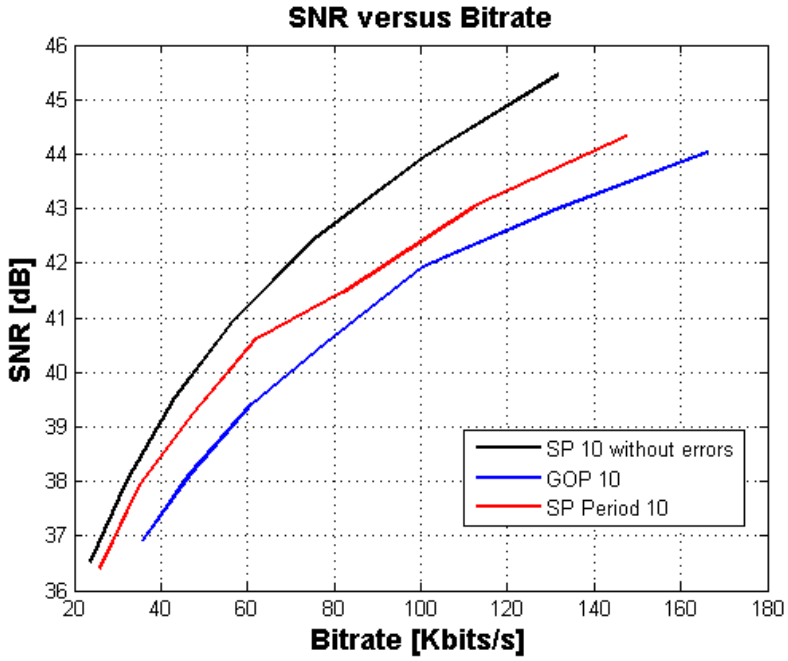

Figure 7.45: Foreman CIF - SP Period 7, delay 100 ms, quality degradation in transmission.

APPENDIX D: Encoder's configuration file

The different features of the encoded sequence are decided in the configuration file. The types of frames in the bitstream is chosen by selecting the periodicity of I and SP. If it is a bitstream with only P and I frames, the SP periodicity is set to 0 and the GOP is set in the field IntraPeriod. In case of a bitstream with P and SP/SI frames, the value of the previous field must be 0 and the SP periodicity is set to the desired value. If the bitstream is with SI frames instead of SP frames, SI_FRAMES parameter must be 1. Another selectionable characteristics are the video to encode, its resolution, the profile of the encoder, the value of the different quantization parameters involved, which packetization to use, etcetera. In this project the packetization used is RTP packets with fixed size of 700 bytes and the parameters to fixed are: OutFileMode, SliceMode and SliceArgument.

The identifiers used for the different quantization parameters used in the configuration file are:

- **QP** = QPISlice and QPPSlice because the same values is used in this project for the quatization parameter in I and P frames coincide.
- **PQP** = QPSPSlice
- **SPQP** = QPSP2Slice

It is shown below a configuration file with SP frames:

```
# <ParameterName >= <ParameterValue ># Comment
#
# See configfile.h for a list of supported ParameterNames
#
# For bug reporting and known issues see:
# https://ipbt.hhi.de

###############################################
# Files
###############################################


InputFile = "mother_daughter_qcif.txt"# Input sequence
```

InputHeaderLength = 0 # If the inputfile has a header, state it is length in byte here

StartFrame = 0 # Start frame for encoding. (0-N)

FramesToBeEncoded = 100 # Number of frames to be coded

FrameRate = 30.0 # Frame Rate per second (0.1-100.0)

SourceWidth = 176 # Frame width

SourceHeight = 144 # Frame height

TraceFile = “trace_enc.txt”

ReconFile = “test_recMOTHER_DAUGHTER_sp_5_17_16_7.yuv”

OutputFile = “testMOTHER_DAUGHTER_sp_5_17_16_7.264”

##
Encoder Control
##

ProfileIDC = 88 # Profile IDC (66=baseline, 77=main, 88=extended; FREXT Profiles: 100=High, 110=High 10, 122=High 4:2:2, 244=High 4:4:4, 44=CAVLC 4:4:4 Intra)

LevelIDC = 40 # Level IDC (e.g. 20 = level 2.0)

IntraPeriod = 0 # Period of I-pictures (0=only first)

EnableOpenGOP = 0 # Support for open GOPs (0: disabled, 1: enabled)

IDRIntraEnable = 0 # Force IDR Intra (0=disable 1=enable)

QPISlice = 17 # Quant. param for I Slices (0-51)

QPPSlice = 17 # Quant. param for P Slices (0-51)

FrameSkip = 2 # Number of frames to be skipped in input (e.g 2 will code every third frame)

ChromaQPOffset = 0 # Chroma QP offset (-51..51)

UseHadamard = 1 # Hadamard transform (0=not used, 1=used for all subpel positions, 2=use only for qpel)

DisableSubpelME = 0 # Disable Subpixel Motion Estimation (0=off/default, 1=on)

SearchRange = 16 # Max search range

NumberReferenceFrames = 1 # Number of previous frames used for inter motion search (0-16)

PList0References = 0 # P slice List 0 reference override (0 disable, N ¡= NumberReferenceFrames)

Log2MaxFNumMinus4 = 0 # Sets log2_max_frame_num_minus4 (-1 : based on FramesToBeEncoded/Auto, ¿=0 :

Log2MaxFNumMinus4)

Log2MaxPOCLsbMinus4 = -1 # Sets log2_max_pic_order_cnt_lsb_minus4 (-1 : Auto, ¿=0 : Log2MaxPOCLsbMinus4)

GenerateMultiplePPS = 0 # Transmit multiple parameter sets. Currently parameters basically enable all WP modes (0: diabled, 1: enabled)

ResendPPS = 0 # Resend PPS (with pic_parameter_set_id 0) for every coded Frame/Field pair (0: disabled, 1: enabled)

MbLineIntraUpdate = 0 # Error robustness(extra intra macro block updates)(0=off, N: One GOB every N frames are intra coded)

RandomIntraMBRefresh = 0 # Forced intra MBs per picture

InterSearch16x16 = 1 # Inter block search 16x16 (0=disable, 1=enable)

InterSearch16x8 = 1 # Inter block search 16x8 (0=disable, 1=enable)

InterSearch8x16 = 1 # Inter block search 8x16 (0=disable, 1=enable)

InterSearch8x8 = 1 # Inter block search 8x8 (0=disable, 1=enable)

InterSearch8x4 = 1 # Inter block search 8x4 (0=disable, 1=enable)

InterSearch4x8 = 1 # Inter block search 4x8 (0=disable, 1=enable)

InterSearch4x4 = 1 # Inter block search 4x4 (0=disable, 1=enable)

IntraDisableInterOnly = 0 # Apply Disabling Intra conditions only to Inter Slices (0:disable/default,1: enable)

Intra4x4ParDisable = 0 # Disable Vertical & Horizontal 4x4

Intra4x4DiagDisable = 0 # Disable Diagonal 45degree 4x4

Intra4x4DirDisable = 0 # Disable Other Diagonal 4x4

Intra16x16ParDisable = 0 # Disable Vertical & Horizontal 16x16

Intra16x16PlaneDisable = 0 # Disable Planar 16x16

ChromaIntraDisable = 0 # Disable Intra Chroma modes other than DC

EnableIPCM = 1 # Enable IPCM macroblock mode

DisposableP = 0 # Enable Disposable P slices in the primary layer (0: disable/default, 1: enable)

DispPQPOffset = 0 # Quantizer offset for disposable P slices (0: default)

##
B Slices
##

NumberBFrames = 0 # Number of B coded frames inserted (0=not used)

QPBSlice = 30 # Quant. param for B slices (0-51)

BRefPicQPOffset = 0 # Quantization offset for reference B coded pictures (-51..51)

DirectModeType = 1 # Direct Mode Type (0:Temporal 1:Spatial)

DirectInferenceFlag = 1 # Direct Inference Flag (0: Disable 1: Enable)

BList0References = 0 # B slice List 0 reference override (0 disable, N ¡= NumberReferenceFrames)

BList1References = 1 # B slice List 1 reference override (0 disable, N ¡= NumberReferenceFrames)

1 List1 reference is usually recommended for normal GOP Structures.

A larger value is usually more appropriate if a more flexible

structure is used (i.e. using HierarchicalCoding)

BReferencePictures = 0 # Referenced B coded pictures (0=off, 1=B references for secondary layer, 2=B references for primary layer)

HierarchicalCoding = 0 # B hierarchical coding (0= off, 1= 2 layers, 2= 2 full hierarchy, 3 = explicit)

HierarchyLevelQPEnable = 1 # Adjust QP based on hierarchy level (in increments of 1). Overrides BRefPicQPOffset behavior.(0=off, 1=on)

ExplicitHierarchyFormat = “b1r0b3r0b2e2b0e2b4r2”# Explicit Enhancement GOP. Format is FrameDisplay_orderReferenceQP.

Valid values for reference type is r:reference, e:non reference.

ReferenceReorder = 1 # Reorder References according to Poc distance for HierarchicalCoding (0=off, 1=enable)

PocMemoryManagement = 1 # Memory management based on Poc Distances for HierarchicalCoding (0=off, 1=on)

BiPredMotionEstimation = 0 # Enable Bipredictive based Motion Estimation (0:disabled, 1:enabled)

BiPredMERefinements = 3 # Bipredictive ME extra refinements (0: single, N: N extra refinements (1 default)

BiPredMESearchRange = 16 # Bipredictive ME Search range (8 default). Note that range is halved for every extra refinement.

BiPredMESubPel = 1 # Bipredictive ME Subpixel Consideration (0: disabled, 1: single level, 2: dual level)

##
SP Frames
##

SPPicturePeriodicity = 5 # SP-Picture Periodicity (0=not used)

QPSPSlice = 16 # Quant. param of SP-Slices for Prediction Error (0-51)

QPSP2Slice = 7 # Quant. param of SP-Slices for Predicted Blocks (0-51)

SI_FRAMES = 0 # SI frame encoding flag (0=not used, 1=used)

SP_output = 0 # Controls whether coefficients will be output to encode switching SP frames (0=no, 1=yes)

SP_output_name = “low_quality.dat”# Filename for SP output coefficients

SP2_FRAMES = 0 # switching SP frame encoding flag (0=not used, 1=used)

SP2_input_name1 = “high_quality.dat”# Filename for the first swithed bitstream coefficients

SP2_input_name2 = “low_quality.dat”# Filename for the second switched bitstream coefficients

##
Output Control, NALs
##

SymbolMode = 0 # Symbol mode (Entropy coding method: 0=UVLC, 1=CABAC)

OutFileMode = 1 # Output file mode, 0:Annex B, 1:RTP

PartitionMode = 0 # Partition Mode, 0: no DP, 1: 3 Partitions per Slice

##
CABAC context initialization
##

ContextInitMethod = 1 # Context init (0: fixed, 1: adaptive)

FixedModelNumber = 0 # model number for fixed decision for inter slices (0, 1, or 2)

##
Interlace Handling
##

PicInterlace = 0 # Picture AFF (0: frame coding, 1: field coding, 2:adaptive frame/field coding)

MbInterlace = 0 # Macroblock AFF (0: frame coding, 1: field coding, 2:adaptive frame/field coding, 3: frame MB-only AFF)

IntraBottom = 0 # Force Intra Bottom at GOP Period

##
Weighted Prediction
##

WeightedPrediction = 0 # P picture Weighted Prediction (0=off, 1=explicit mode)

WeightedBiprediction = 0 # B picture Weighted Prediciton (0=off, 1=explicit mode, 2=implicit mode)

UseWeightedReferenceME = 0 # Use weighted reference for ME (0=off, 1=on)

##
Picture based Multi-pass encoding
##

RDPictureDecision = 0 # Perform RD optimal decision between different coded picture versions.

If GenerateMultiplePPS is enabled then this will test different WP methods.

Otherwise it will test QP +-1 (0: disabled, 1: enabled)

RDPictureIntra = 0 # Perform RD optimal decision also for intra coded pictures (0: disabled (default), 1: enabled).

RDPSliceWeightOnly = 1 # Only consider Weighted Prediction for P slices in Picture RD decision. (0: disabled, 1: enabled (default))

RDBSliceWeightOnly = 0 # Only consider Weighted Prediction for B slices in Picture RD decision. (0: disabled (default), 1: enabled)

##
Loop filter parameters
##

LoopFilterParametersFlag = 0 # Configure loop filter (0=parameter below ingored, 1=parameters sent)

LoopFilterDisable = 0 # Disable loop filter in slice header (0=Filter, 1=No Filter)

LoopFilterAlphaC0Offset = 0 # Alpha & C0 offset div. 2, -6, -5, ... 0, +1, .. +6

LoopFilterBetaOffset = 0 # Beta offset div. 2, -6, -5, ... 0, +1, .. +6

##
Error Resilience / Slices
##

SliceMode = 2 # Slice mode (0=off 1=fixed #mb in slice 2=fixed #bytes in slice 3=use callback)

SliceArgument = 700 # Slice argument (Arguments to modes 1 and 2 above)

num_slice_groups_minus1 = 0 # Number of Slice Groups Minus 1, 0 == no FMO, 1 == two slice groups, etc.

slice_group_map_type = 0 # 0: Interleave, 1: Dispersed, 2: Foreground with left-over,

3: Box-out, 4: Raster Scan 5: Wipe

6: Explicit, slice_group_id read from SliceGroupConfigFileName

slice_group_change_direction_flag = 0 # 0: box-out clockwise, raster scan or wipe right,

1: box-out counter clockwise, reverse raster scan or wipe left

slice_group_change_rate_minus1 = 85 #

SliceGroupConfigFileName = "sg0conf.cfg"# Used for slice_group_map_type 0, 2, 6

UseRedundantPicture = 0 # 0: not used, 1: enabled

NumRedundantHierarchy = 1 # 0-4

PrimaryGOPLength = 10 # GOP length for redundant allocation (1-16)

NumberReferenceFrames must be no less than PrimaryGOPLength when redundant slice enabled

NumRefPrimary = 1 # Actually used number of references for primary slices (1-16)

##
Search Range Restriction / RD Optimization
##

RestrictSearchRange = 2 # restriction for (0: blocks and ref, 1: ref, 2: no restrictions)

RDOptimization = 1 # rd-optimized mode decision

0: RD-off (Low complexity mode)

1: RD-on (High complexity mode)

2: RD-on (Fast high complexity mode - not work in FREX Profiles)

3: with losses

DisableThresholding = 0 # Disable Thresholding of Transform Coefficients (0:off, 1:on)

DisableBSkipRDO = 0 # Disable B Skip Mode consideration from RDO Mode decision (0:off, 1:on)

SkipIntraInInterSlices = 0 # Skips Intra mode checking in inter slices if certain mode decisions are satisfied (0: off, 1: on)

Explicit Lambda Usage

UseExplicitLambdaParams = 0 # Use explicit lambda scaling parameters (0:disabled, 1:enable lambda weight, 2: use explicit lambda value)

LambdaWeightIslice = 0.65 # scaling param for I slices. This will be used as a multiplier i.e.

lambda=LambdaWeightISlice * 2$\hat{}$((QP-12)/3)

LambdaWeightPslice = 0.68 # scaling param for P slices. This will be used as a multiplier i.e. lambda=LambdaWeightPSlice * 2$\hat{}$((QP-12)/3)

LambdaWeightBslice = 2.00 # scaling param for B slices. This will be used as a multiplier i.e. lambda=LambdaWeightBSlice * 2$\hat{}$((QP-12)/3)

LambdaWeightRefBslice = 1.50 # scaling param for Referenced B slices. This will be used as a multiplier i.e. lambda=LambdaWeightRefBSlice * 2$\hat{}$((QP-12)/3)

LambdaWeightSPslice = 1.50 # scaling param for SP slices. This will be used as a multiplier i.e. lambda=LambdaWeightSPSlice * 2$\hat{}$((QP-12)/3)

LambdaWeightSIslice = 0.65 # scaling param for SI slices. This will be used as a multiplier i.e. lambda=LambdaWeightSISlice * 2$\hat{}$((QP-12)/3)

LossRateA = 5 # expected packet loss rate of the channel for the first partition, only valid if RDOptimization = 3

LossRateB = 0 # expected packet loss rate of the channel for the second partition, only valid if RDOptimization = 3

LossRateC = 0 # expected packet loss rate of the channel for the third partition, only valid if RDOptimization = 3

NumberOfDecoders = 30 # Numbers of decoders used to simulate the channel, only valid if RDOptimization = 3

RestrictRefFrames = 0 # Doesnt allow reference to areas that have been intra updated in a later frame.

##
Additional Stuff
##

UseConstrainedIntraPred = 0 # If 1, Inter pixels are not used for Intra macroblock prediction.

LastFrameNumber = 0 # Last frame number that have to be coded (0: no effect)

ChangeQPI = 16 # QP (I-slices) for second part of sequence (0-51)

ChangeQPP = 16 # QP (P-slices) for second part of sequence (0-51)

ChangeQPB = 18 # QP (B-slices) for second part of sequence (0-51)

ChangeQPBSRefOffset = 2 # QP offset (stored B-slices) for second part of sequence (-51..51)

ChangeQPStart = 0 # Frame no. for second part of sequence (0: no second part)

NumberofLeakyBuckets = 8 # Number of Leaky Bucket values

LeakyBucketRateFile = “leakybucketrate.cfg”# File from which encoder derives rate values

LeakyBucketParamFile = “leakybucketparam.cfg”# File where encoder stores leakybucketparams

NumberFramesInEnhancementLayerSubSequence = 0 # number of frames in the Enhanced Scalability Layer(0: no Enhanced Layer)

SparePictureOption = 0 # (0: no spare picture info, 1: spare picture available)

SparePictureDetectionThr = 6 # Threshold for spare reference pictures detection

SparePicturePercentageThr = 92 # Threshold for the spare macroblock percentage

PicOrderCntType = 0 # (0: POC mode 0, 1: POC mode 1, 2: POC mode 2)

##
Rate control
##

RateControlEnable = 0 # 0 Disable, 1 Enable

Bitrate = 45020 # Bitrate(bps)

InitialQP = 0 # Initial Quantization Parameter for the first I frame

InitialQp depends on two values: Bits Per Picture,

and the GOP length

BasicUnit = 11 # Number of MBs in the basic unit

should be a fractor of the total number

of MBs in a frame

ChannelType = 0 # type of channel(1=time varying channel; 0=Constant channel)

##
Fast Mode Decision
##

EarlySkipEnable = 0 # Early skip detection (0: Disable 1: Enable)

SelectiveIntraEnable = 0 # Selective Intra mode decision (0: Disable 1: Enable)

##
FREXT stuff
##

YUVFormat = 1 # YUV format (0=4:0:0, 1=4:2:0, 2=4:2:2, 3=4:4:4)

RGBInput = 0 # 1=RGB input, 0=GBR or YUV input

BitDepthLuma = 8 # Bit Depth for Luminance (8...12 bits)

BitDepthChroma = 8 # Bit Depth for Chrominance (8...12 bits)

CbQPOffset = 0 # Chroma QP offset for Cb-part (-51..51)

CrQPOffset = 0 # Chroma QP offset for Cr-part (-51..51)

Transform8x8Mode = 0 # (0: only 4x4 transform, 1: allow using 8x8 transform additionally, 2: only 8x8 transform)

ReportFrameStats = 0 # (0:Disable Frame Statistics 1: Enable)

DisplayEncParams = 0 # (0:Disable Display of Encoder Params 1: Enable)

Verbose = 1 # level of display verboseness (0:short, 1:normal, 2:detailed)

```
##############################################
# Q-Matrix (FREXT)
##############################################

QmatrixFile = "q_matrix.cfg"
ScalingMatrixPresentFlag = 0 # Enable Q_Matrix (0 Not present, 1 Present in SPS, 2 Present in PPS, 3 Present in both SPS & PPS)
ScalingListPresentFlag0 = 3 # Intra4x4_Luma (0 Not present, 1 Present in SPS, 2 Present in PPS, 3 Present in both SPS & PPS)
ScalingListPresentFlag1 = 3 # Intra4x4_ChromaU (0 Not present, 1 Present in SPS, 2 Present in PPS, 3 Present in both SPS & PPS)
ScalingListPresentFlag2 = 3 # Intra4x4_chromaV (0 Not present, 1 Present in SPS, 2 Present in PPS, 3 Present in both SPS & PPS)
ScalingListPresentFlag3 = 3 # Inter4x4_Luma (0 Not present, 1 Present in SPS, 2 Present in PPS, 3 Present in both SPS & PPS)
ScalingListPresentFlag4 = 3 # Inter4x4_ChromaU (0 Not present, 1 Present in SPS, 2 Present in PPS, 3 Present in both SPS & PPS)
ScalingListPresentFlag5 = 3 # Inter4x4_ChromaV (0 Not present, 1 Present in SPS, 2 Present in PPS, 3 Present in both SPS & PPS)
ScalingListPresentFlag6 = 3 # Intra8x8_Luma (0 Not present, 1 Present in SPS, 2 Present in PPS, 3 Present in both SPS & PPS)
ScalingListPresentFlag7 = 3 # Inter8x8_Luma (0 Not present, 1 Present in SPS, 2 Present in PPS, 3 Present in both SPS & PPS)

##############################################
# Rounding Offset control
##############################################

OffsetMatrixPresentFlag = 0 # Enable Explicit Offset Quantization Matrices (0: disable 1: enable)
QOffsetMatrixFile = "q_offset.cfg"# Explicit Quantization Matrices file
AdaptiveRounding = 0 # Enable Adaptive Rounding based on JVT-N011 (0: disable, 1: enable)
AdaptRndPeriod = 1 # Period in terms of MBs for updating rounding offsets.
# 0 performs update at the picture level. Default is 16. 1 is as in JVT-N011.
AdaptRndChroma = 0 # Enables coefficient rounding adaptation for chroma
AdaptRndWFactorIRef = 4 # Adaptive Rounding Weight for I/SI slices in reference pictures /4096
AdaptRndWFactorPRef = 4 # Adaptive Rounding Weight for P/SP slices in reference pictures /4096
AdaptRndWFactorBRef = 4 # Adaptive Rounding Weight for B slices in reference pictures /4096
AdaptRndWFactorINRef = 4 # Adaptive Rounding Weight for I/SI slices in non reference pictures /4096
```

AdaptRndWFactorPNRef = 4 # Adaptive Rounding Weight for P/SP slices in non reference pictures /4096

AdaptRndWFactorBNRef = 4 # Adaptive Rounding Weight for B slices in non reference pictures /4096

```
###############################################
# Lossless Coding (FREXT)
###############################################
```

QPPrimeYZeroTransformBypassFlag = 0 # Enable lossless coding when qpprime_y is zero (0 Disabled, 1 Enabled)

```
###############################################
# Fast Motion Estimation Control Parameters
###############################################
```

UseFME = 0 # Use fast motion estimation (0=disable/default, 1=UMHexagonS
# 2=Simplified UMHexagonS, 3=EPZS patterns)
FMEDSR = 1 # Use Search Range Prediction. Only for UMHexagonS method
# (0:disable, 1:enabled/default)
FMEScale = 3 # Use Scale_factor for different image sizes. Only for UMHexagonS method
# (0:disable, 3:/default)
# Increasing value can speed up Motion Search.
EPZSPattern = 2 # Select EPZS primary refinement pattern.
# (0: small diamond, 1: square, 2: extended diamond/default,
# 3: large diamond, 4: SBP Large Diamond,
# 5: PMVFAST)
EPZSDualRefinement = 3 # Enables secondary refinement pattern.
# (0:disabled, 1: small diamond, 2: square,
# 3: extended diamond/default, 4: large diamond,
# 5: SBP Large Diamond, 6: PMVFAST)
EPZSFixedPredictors = 2 # Enables Window based predictors
# (0:disabled, 1: P only, 2: P and B/default)
EPZSTemporal = 1 # Enables temporal predictors
# (0: disabled, 1: enabled/default)
EPZSSpatialMem = 1 # Enables spatial memory predictors
# (0: disabled, 1: enabled/default)
EPZSMinThresScale = 0 # Scaler for EPZS minimum threshold (0 default).
# Increasing value can speed up encoding.
EPZSMedThresScale = 1 # Scaler for EPZS median threshold (1 default).
# Increasing value can speed up encoding.
EPZSMaxThresScale = 1 # Scaler for EPZS maximum threshold (1 default).
# Increasing value can speed up encoding.

APPENDIX E: Decoder's configuration file

Example of a configuration file of the decoder:

```
mezcla_5_17_100_50.txt          .....H.26L coded bitstream
decoder_video.yuv               .....Output file, YUV/RGB
mother_daughter_qcif.txt        .....Ref sequence (for SNR)
1       .....Write 4:2:0 chroma components for monochrome streams
1       .....NAL mode (0=Annex B, 1: RTP packets)
0       .....SNR computation offset
2       .....Poc Scale (1 or 2)
500000      .....Rate_Decoder
104000      .....B_decoder
73000       .....F_decoder
leakybucketparam.cfg            .....LeakyBucket Params
1       .....Err Concealment(0:Off,1:Frame Copy,2:Motion Copy)
4       .....Reference POC gap (2: IPP (Default), 4: IbP / IpP)
4       .....POC gap (2: IPP /IbP/IpP (Default), 4: IPP with frame skip = 1 etc.)

This is a file containing input parameters to the JVT H.264/AVC decoder.
The text line following each parameter is discarded by the decoder.
```