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DEPARTMENT OF COMPUTER SCIENCE

Multiple Description Image and Video Coding for P2P Transmissions

Marisa Raposo Quaresma

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Dissertation written under the supervision of
Dra. Maria Manuela Areias Costa Pereira de Sousa, Assistant Professor of Computer Science at the University of Beira Interior and co-supervision of Dr. Mário Marques Freire, Full Professor of Computer Science at the University of Beira Interior, Portugal

To my mother and sister.

Acknowledgments

First of all I would like to dedicate this thesis to my mother and sister.

To my mother, who has been giving me the opportunity to realize all my dreams. Thank you for being the extraordinary person that you are.

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Abstract

Peer-to-Peer (P2P) media streaming is, nowadays, a very attractive topic due to the bandwidth available to serve demanding content scales. A key challenge, however, is making content distribution robust to peer transience. Multiple description coding (MDC) has, indeed, proven to be very effective with problems concerning the packets' losses, since it generates several descriptions and may reconstruct the original information with any number of descriptions that may reach the decoder. Therefore multiple descriptions may be effective for robust peer-to-peer media streaming. In this dissertation, it will not only be showed that, but also that varying the redundancy level of description on the fly may lead to a better performance than the one obtained without varying this parameter. Besides that, it is shown, as well, that varying the Bitrate on the fly outperforms the redundancy on it. Furthermore, the redundancy and the Bitrate were varied simultaneously. Thus, it is shown that this variation is more efficient when the packet loss is high.

The experiments reported above were done using an experimental test bed developed for this purpose at the NMCG lab of the University of Beira Interior. It was also used the REGPROT, a video encoder developed by our research team, to splitted the video into multiple descriptions, which were, later, distributed among the peers in the test bed. After the request of the client, the referred encoder decoded the descriptions as they were being received.

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List of Symbols

D_0 Minimal Central Distortion

D_M Maximal Distortion

D_1 Lateral Distortion

D_2 Lateral Distortion

R_1 Side Rate

R_2 Side Rate

R_T Total Rate

t Time

List of Abbreviations and Acronyms

CIF	Common Intermediate Format
CPU	Central Processing Unit
DHT	Distributed Hash Tables
DWT	Discrete Wavelet Transform
F-MDC	Flexible Multiple Description Coding
Fps	Frames per second
GOP	Group of Pictures
HD	Hard Disk
IPTO	Information Processing Techniques Office
LAN	Network of Local Area
MDBA	Multiple Description Bit Allocation
MDC	Multiple Description Coding
MDSQ	Multiple Description Scalar Quantization
OS	Operating System
P2P	Peer-to-Peer
PSNR	Peak Signal-to-Noise Ratio
QCIF	Quarter Common Intermediate Format
RAM	Random Access Memory
RSS	Really Simple Syndication

Chapter 1

Introduction

1.1 Dissertation Focus and Scope

This Dissertation is devoted to the problem of multiple description image and video coding for transmission over peer-to-peer networks. Therefore, it addresses two research areas that traditionally have evolved separately: the area of multiple descriptions coding and the area of peer-to-peer networks.

Along the years, P2P networks have experienced a huge increase of popularity. In the way that it allows the trade of files in an easy way. The P2P networks consist in an overlay network in which each node, also called peer, is connected to a set of nodes/peers. The communication between each node is done by a P2P application that will bring to the client node the knowledge of the bandwidth of the sender nodes as well as of the nodes that contain the files that the client node is looking for. If there is lots of traffic in the network the communication between the nodes can be hard, therefore the same file is distributed for more than one node, so if the node that is sending the file is down, the other can continue sending the file, without breaking the download in the client node.

The classic technique to combat transmission errors is Forward Error Correction (FEC). FEC involves the addition of redundant data to the compressed signal, which allows the decoder to correct errors up to a certain level. This redundancy increases the total number of bits required and thus reduces compression. Moreover, FEC code must be designed with a worst case channel scenario in mind. For channels that have a highly variable quality, this worst case may imply the need for a very powerful code, and hence highly or even prohibitive amount of redundancy, which will severally reduce the compression performance. In the case of burst errors the error correction capability is often exceeded or the block is error-free in which case additional redundancy is wasted. To overcome this limitation, FEC is often enhanced by a technique known as interleaving. For burst errors, this effectively reduces concentration of errors in single code words, more precisely, a burst of consecutive symbol errors causes a maximum of symbol errors in each code word. Thought interleaving can be implemented with low complexity it suffers from increased delay, depending on the number of interleaved blocks. Therefore interleaving is a frequently used technique for bursty channels if additional delay is acceptable. These problems can be solved if used unequal loss protection. Closed-loop error control techniques like Automatic Repeat Request (ARQ) have been shown to be more effective than FEC and successfully to wireless video transmission. Retransmission of corrupted data frames, however, introduces additional delay. Moreover, it is possible to combine FEC-ARQ to be successfully to wireless video transmission. The frameworks for P2P communication should join optimization of source coding and channel coding, should present great robustness and adaptability to adverse transmission condition and should make efficient use of limited network resources. In this way, the joint source and channel coding method, known as multiple description coding (MDC), has proven to be an effective way to provide error resilience with a relatively small reduction in compression ratio. This coding scheme assumes that there are several parallel channels between the source and destination, and that each channel may be temporarily down or suffering from long burst errors. Furthermore, the error events of different channels are independent so that the probability that all channels simultaneously experience losses is small [2].

On that account this dissertation has two objectives. The first consists in defining how many descriptions shall be sent, which depends on the packets loss during the transmission, and the second consists in defining the redundancy of each description, which will depend on the errors rate of the arrived packets. Therefore, the MDC coder used in this dissertation will adapt the redundancy in the descriptions taking into account the error rates arrived in the packets and the number of descriptions taking into account the packet loss. So each time that there is an increase of packets loss, the quantity of description used will increase, and vice-versa. In chapter four, it is presented how the redundancy varies according to the errors rate. The quantity of descriptions used was based in 2^n, where $n = 1, 2, 3$. In this way, the encoder will always be satisfied even with the channels alterations.

1.2 Problem Statement and Objectives

Nowadays, many are the techniques that allow video files to have a better quality but sending these video files through the network with the same quality is still hard to achieve. Even with the increase of the bandwidth and high speed that the networks now achieve, when there is a lot of traffic many packets can be lost in the network, which becomes a problem when a video file of a high importance is sent and some of its information is lost, or when the source that is sending the information goes down the receiver has to downloaded it again.

If the video file is also only split into multiple packets, but these packets have no Redundancy at all and if one packet is lost, important information will be lost, which will decrease the quality of the received video file. So, in order to solve these problems, the objective of this thesis is to show how using multiple description coding together with P2P networks will improve the quality of the received video file, since the multiple packets are distributed for more than one source. Therefore, if one of the sources goes down the entire video file will not be lost.

It is also objective of this dissertation to show the importance of splitting the video into multiple packets with redundancy. To achieve this, the encoding technique used is multiple descriptions coding, which allows the video to be split into multiple descriptions with redundant information. If one of the descriptions is lost along the network the other will still be received, which will allow the receiver not to lose the entire information of the lost description, since the received description has information that concerns to the lost description. In this way, the video quality shall not be dramatically decreased.

1.3 Main Contributions

This section describes, in the opinion of the Author, main contributions resulting from this research programme for the advance of the state of art on media streaming over peer-to-peer networks.

The first contribution of this dissertation is the proposal of a robust scheme for media streaming over peer-to-peer networks using multiple description coding. The video encoder used to split the video into multiple descriptions is named REGPROT, which has been developed by others members of the team in an coordinated effort.

The second contribution of this dissertation is a study of the performance and robustness of the proposed scheme for media streaming over peer-to-peer networks using multiple description coding with two, four or eight descriptions.

1.4 Organization of the Dissertation

This dissertation is organized into five chapters. The first chapter begins with the focus and scope of this thesis, followed by the problem statement to which some solutions will be presented in chapter four, then, finally the organization of the thesis shall be presented followed by the main contributions that this dissertation will have in future work.

The second chapter is dedicated to P2P networks. Firstly, a brief history of P2P networks will be presented, followed by a description of their features. Then some of the disadvantages and advantages of this type of networks will be stated. This chapter also contains the classification of P2P systems. One of the main objectives of this thesis is P2P streaming, so in this chapter is presented a description of P2P streaming, as well as the network topology and advantages and disadvantages of P2P streaming. In the end, five of the most used P2P applications will be described.

Chapter three is devoted to multiple description coding. In here the multiple description problem is presented, as well as some of the main approaches of MDC. Video streaming is also stated in here, in order to explain how the video streaming works and which the uses of this technique are.

The main contributions of this dissertation are described in chapter four. In that chapter, a state of the art on video streaming over P2P is introduced. This description will also allow presenting an overview of everything that has been done in video streaming over P2P networks so far. It is described the proposed P2P experimental test bed to solve the problems stated in the presented chapter. It is also described the multiple description encoder used in the splitting of video files, and then, relevant results, which show that joining P2P networks with multiple descriptions is a good idea, are presented.

Finally, in chapter five the main conclusions of this work and the perspectives for a future work will be presented.

Chapter 2

Peer-to-Peer Networks

2.1 Introduction

Peer-to-Peer network concept was at first proposed by Vannevar Bush in July 1945. He defined this concept as follows:

“Consider a future device for individual use, which is a sort of mechanized private file and library. It needs a name, and to coin one at random, “memex” will do.

A memex is a device in which an individual stores all his books, records and communications, and which is mechanized so that it may be consulted with exceeding speed and flexibility. It is an enlarged intimate supplement to his memory. It consists of a desk and while it can presumably be operated from a distance, it is primarily the piece of furniture at which he works. On the top are slanting translucent screens, on which material can be projected for convenient reading. There is a keyboard, sets of buttons and levers. Otherwise it looks like an ordinary desk” [3].

Then, in the late 1960’s, Lick Licklider and Lawrence Roberts used Vannevar Bush concept and conceived and planned the first peer-to-peer network, the ARPANET. Arpanet was developed by the Information Processing Techniques Office (IPTO) and went into labor on August 30, 1969 [4]. The aim goal was to share computing resources around the United States of America, which would integrate different kinds of existing networks as well as future technologies with a common network architecture that would allow every host to be an equal player [5].

Nowadays, P2P stands for a system or an application that uses distributed resources to execute a function, in which each node can act as a server or a client [6].

2.2 Peer-to-Peer Features

Many are the features that identify a peer-to-peer system. However, a system does not have to have all of them. That shall be discussed later to be considered a P2P System. In this way, the features of a peer-to-peer system are: self-organization, role symmetry, resource sharing, scalability, peer autonomy and resilience. In the following a brief description of these features is presented.

2.2.1 Self-organization

Self-organization means that peers cooperate in the formation and maintenance of the overlay[1] [7]. In a P2P system that is self organized, the different system components work together without any central management instance assigning roles and tasks. The structures of these systems are difficult to determine because system-wide and governing policies do not apply [8].

2.2.2 Role Symmetry

In contrast to client server computing, where the roles of the end points are asymmetric, peers are functionally equal (symmetric roles). Each peer may store objects, support queries and perform routing of messages [9].

[1] Overlay: is a logical layer for message delivery between peers[6].

2.2.3 Resource Sharing

Each peer can act as a server and, therefore share resources, information and services. The resource contribution shall be fair, in this way, it can be established that peer resource contribution never exceed a certain bound and that the resource contribution shall be mutually beneficial [9].

2.2.4 Scalability

Nowadays, several P2P applications work with millions of peers, therefore an important dimension of scalability is the ability to operate the P2P overlay, as the size grows by a hundred times or more [10]. When a system is looking for an acceptable scalability it must have into account four features [11]:

1. Scalable – The system must have the capability of being modifiable. If the system can be modified without being replaced, then it has this quality.

2. Downtime – Scalability requires the interruption of the operation in some times. When a system needs an extra storage place, it might be necessary to switch it off. The time that the interruption lasts determines the scalability of the system.

3. Seamless Scalability – This feature enables the system to be restored and repaired without being needed a complete interruption.

4. Non-Seamless Scalability – Unlike the previous feature, in this one the system shall be stopped to be updated. For

instance, when it is needed the upgrade of the system memory the system must be ceased.

2.2.5 Autonomy

In Peer-to-Peer Systems, each peer determines: 1) its abilities based on its own resources; 2) when it joins the overlay; 3) which requests it makes to the overlay; 4) and when it leaves the overlay.

However, peer autonomy leads to unpredictability in services, i.e., a peer that searches for an information and does not found its information, may not know if the information was in the overlay or not.

2.2.6 Resilience

In a peer-to-peer system, a peer can join or leave the system. To deal with this problem, P2P systems implement a stabilization routine which repairs continuously the overlay as peers come and go, updating control information and routing tables to ensure that that the overlay remains connected [12].

2.3 Drawbacks of P2P Networks

Even though a peer-to-peer network is very cheap and simple to install and use, it has several drawbacks, especially when we are dealing with a large network, such as:

1) User's performance – it can affect the user's performance due to the resource of sharing [9].

2) Not very secure – they are not very secure because we cannot guarantee that the user will administer his machine appropriately, i.e., that the user will not share any illegal material, or even, that a virus is not in his shares [10].

3) Hard to back up – it is difficult to back up all the data scattered over many workstations [10].

4) Decentralization of resources – it may be difficult for the user to locate particular resources [9].

Despite all these disadvantages if the users can cooperate well, peer-to-peer networking is a good way to share resources [9].

2.4 Advantages of P2P Networks

The advantages of peer-to-peer network are:

1) Use of less expensive computer hardware - In P2P networks, resources are distributed over many computers, therefore there is no need for an high end server computer [10].

2) Easy to administer – P2P is easy to set up, because each machine performs and administers its own resources [10].

3) All the needed software is included in the operating system [9].

4) The computer's peers do not do not depend on a central server for their resources [9].

5) P2P computers have more scalability and tolerance to faults, because they do not depend on a central server [12].

2.5 Classification of P2P Systems

In what refers to classification, some aspects shall be taken into account. Those are: functional classification, degree of decentralization and structures of the information system [12].

2.5.1 Functional Classification

In this point of view, a peer-to-peer system can be classified in three categories, as Mário Freire and others stated in Universal Multiservice Network. Those categories are [12]:

1) Management and contents-sharing applications - The nodes in these systems contain digital media files that will be shared on the network. When a node wants some files, it sends a query through the network and waits for a reply. Then the node that has the file that the other node wants, will share its file with the node that has requested it.

2) Distributed processing - In this kind of systems, each computer can act both as client and server, i.e., a node can contain the media files and share it with other nodes while it receives files as well.

3) Collaboration and communication - In what concerns these systems, the nodes communicate among them sharing information about which contents they have, and then they collaborate among them in order to share their files in a more effective way.

2.5.2 Degree of Decentralization

Based on the degree of decentralization a P2P system can also be classified in tree categories [12]:

1) Pure decentralized systems – In these systems, the nodes can communicate directly without needing an intermediate central point. Examples of these systems are the GNUTELLA and the Freenet applications.

2) Partially decentralized systems – In this kind of systems there is a super node that acts like a server. The other nodes send a request to the super node, which will respond with the content of their requests, or, in case, that it does not have the content, it will send the request to other nodes. As examples of these systems we have: KazaA, Morpheus, iMech, among others.

3) Hybrid decentralized systems – Regarding these systems, there is a central server in touch with a central directory of shared resources and users, to which a peer connects. The advantages of these systems are the efficient location of resources and the network global view. Napster and Bittorrent are examples of these systems.

2.5.3 Structure Degree of the Information System

A peer-to-peer system can be classified according to the degree of its information system in three categories:

1) Structured systems – Due to the relation between the content and its node, the localization of the contents depends on the overlay [13]. These systems goal is to turn P2P networks dynamic, which is possible due to a localization algorithm that

is, at the same time, a routing algorithm, in an environment that is completely distributed [12].

2) Unstructured – In this category, the nodes can communicate among them in a free way, i.e., the files are managed and stored by their own and the topology network is arbitrary. These kinds of systems have the advantages of being flexible and dynamic [12].

3) Loosely Structured – These systems are characterized by the facts that the overlay is built independent from the application, and that the search depends on the overlay structure and how the data is stored [12].

2.6 P2P Streaming

With the continuous growth of P2P networks, many have been the studies. Based on them researchers wondered how the transmission of media should perform in this kind of networks, thus, P2P streaming[2] systems started to appear. P2P streaming is a method for multicasting or broadcasting media over the Internet, using P2P networks [14].

2.6.1 Network Topology Classification

To implement this kind of systems three main questions shall be answered [15]:

1) Which overlay network topology shall be constructed?

2) What peers shall be selected to send the media data?

[2] "*Streaming refers to a delivery method whereby a content data stream is delivered from a server to a client or clients in a continuous fashion and consumed in real time by client applications*" [5].

3) How to overcome the unpredictable behaviors of peers joining and leaving the network?

P2P streaming systems may be classified into three categories according to the network topology, as follows:

1) Tree-based topology – In these, the peers are organized into a multicast tree to deliver data. When a peer receives data, it will send copies of the received data to all of its children. An example of this kind of topology is the Peercast System. In this topology there are three selection strategies: random, round-robin and smart selections. In what concerns smart selection there are two ways of doing it [15]: a) according to physical placement; b) according to the bandwidth of the peers. This topology has a drawback, which consists in the fact that a single peer may not contribute with a full streaming bandwidth to the receiving peer causing performance problems. Since the media data may not be completed at the end of the transmission.

2) Forest-based topology – This is based in a forest of multicast trees, which are constructed, in order to distribute and forward the data in a decentralized, scalable and self-organized way, into the bandwidth of a peer [15]. An example is the split stream. Even though, this topology seems to be an effective one, the fact of a peer receives data only from a single peer, will result in the same problem of the tree-based topology.

3) Mesh Topology – This is a multi-sender scheme, in which a peer can select and receive data from other set of peers at the same time. Each peer contributes with a portion of its bandwidth. Besides that, the set of peers can change due to their unpredictability online/offline status. An example of this kind of topology is the DoNet [16]. The major challenge is how

to select the sender peers and how to schedule the data among them.

4) Multicast – Multicast allows the communication among nodes. The communication is done by messages that are sent to a set of nodes that can be anywhere. It is very important in P2P distributed systems because in these systems the peers need to be informed of a specific event [17].

2.6.2 Implementations of P2P Streaming Systems

In what concerns peer-to-peer streaming implementation, in the following are the two most popular [16]:

1) SHOUTCAST – it is a free audio homesteading solution, which allows anyone on the Internet to broadcast audio from their PC to listeners across the Internet or any other IP-based network.

2) ICECAST – it is an open source of peer-to-peer streaming. It is community based and supports an open source streaming called Ogg Vorbis.

2.6.3 Advantages and Disadvantages of P2P Streaming

P2P streaming systems have the advantage of the ability of the peers to send media, in real-time, to a large audience. Though, this advantage is the main reason why so many researches have been done, these systems have the drawback of being unreliable, due to unpredictable behavior of peers disconnect from the system any time [1]. Another advantage is the reduction of bandwidth cost and the improvement of the end user experience, since it delivers an excellent quality of sound and picture. The increasing of the total ability of the network ability is other advantage [18].

2.7 Peer-to-Peer Applications

With the widespread of the Internet, many have been the P2P file sharing applications developed along the years that allow the users to share the information along the network. Therefore, only the five most popular P2P applications in the year 2009 will be considered in this section [19].

2.7.1 μTorrent

μTorrent [20] is considered the most popular client of P2P [19]. It is known for its characteristic of being user friendly, allowing the user to use it, without even realize that it is running, because it does not consume any valuable system resource. It only occupies 220 KB of size in hard disk, it is also easy to identify, as many icons have been created, along with the installation of this client on the computer. It has into account the system language as it is the systems language that it chooses. Nevertheless, if the user desires other language, he can choose it.

The μTorrent has some important characteristics such as:

1) Bandwidth Prioritization- This will allocate the bandwidth, according to the importance of the information that is being received [21].

2) Scheduling- Which allows to prioritize the tasks, according to their importance [20].

3) Really Simple Syndication (RSS) auto-downloading – Many RSS feeds have some files attached, so with the right

configuration, when a new RSS appears, the μTorrent will automatically download the file, without the user's intervention [20].

2.7.2 BitComet

Although bitcomet [22] is not as user-friendly as μTorrent is, it also is very good. In the way, that it has an embedded browser, which makes easier the search for the torrents. It allows the user to preview the files while downloading them, which is very useful when the file that is being downloaded is a video or music. It allows multi mirror download, which will automatically search for the file in many servers, and the data will be downloaded at the same time, in order to increase the download speed. Furthermore it has a multi-section download, which will split the file into sections and download the data at the same time.

2.7.3 LimeWire

LimeWire [23] is one of the best file sharing applications developed until today. It can be gathered in two versions: LimeWire and LimeWirePro. The difference between them resides in a most effective way of searching, and the connection with more sources in the second one. Besides this, LimeWire enables the user to search the files by title, artist, album, track number, genre, year, length and Bitrate, which makes the search more effective than if you can only choose the name and need to wait that all the files are scanned by the name that is given. Nevertheless, it has the drawback of having the firewall to firewall ability. Thus, if the user is behind a router or a network of local area (LAN), he has some difficulties to get connected.

2.7.4 Azureus

Azureus [24], which is now called Vuze [25], is very popular because of its ability on the search of high quality content on the Vuze hard disk (HD) network, of being compatibly with all operating systems, of being easy to configure, and of allowing the multi download files for different sources. Though these are some important characteristics, the Vuze has a strong drawback. On that account, if the user does not have enough random access memory (RAM) space, the system performance may be degraded, making the access to other resources of the system very low, or even making the system to crash.

2.7.5 BitLord

BitLord [26] is very popular for its user-friendly characteristic, and because it does not need setup.

BitLord has some strong advantages. It is stable and fast, easy to understand and use, and consumes very low central processing unit (CPU) usage. It has an intelligent connection, which will allow it to connect to many different peers. It has also smart rate control that will allow it to have a maximum download rate. In this way, if a child tries to access some adult content, he will not be able to access, because all this content is restricted to the use of a credit card – though this may be seen as a disadvantage, if the user has to pay for it – so with this client the parents do not need to worry about their kid's safety.

2.8 Conclusions

In this chapter a brief history of peer-to-peer network was presented, as well as the classification of a P2P network. It was also stated that P2P streaming is a good way of sending media files, in real time, into a large audience due to the fact that even if a peer goes down the other peers containing the same file will continue to transmit without breaking the transmission, which will be shown in chapter four. In the end the most popular [19] peer-to-peer applications were introduced. LimeWire was the one chosen in this dissertation to implement the sharing of the media files.

Chapter 3

Multiple Description Coding

3.1 Introduction

Multiple description coding is an invention by Bell Laboratories in connection with Communication Speed over the Telephone Network [26]. Based on the idea of channel splitting, Gersho, who learnt this problem from Goodman, shared it with Witsenhausen, Wolf, Wyner, Ziv and Ozarow. In September 1979 IEEE Information Theory Workshop proposed the MD problem: “Suppose we wish to send a description of a stochastic process to a destination through a communication network. Assume that there is a change that the description will be lost. Therefore we send two descriptions and hope that one of them will get through, then we wish the combined descriptive information to be as large as possible” [28].

Figure 1 shows the stated problem. As we can see, at the receiver, the MDC decoder combines the information of the two descriptions and reconstructs the original sign. It can combine the data of the two of the two descriptions, when both are received, or when only one is received, the encoder will only decode the data from the received decoder, which might not have a good quality.

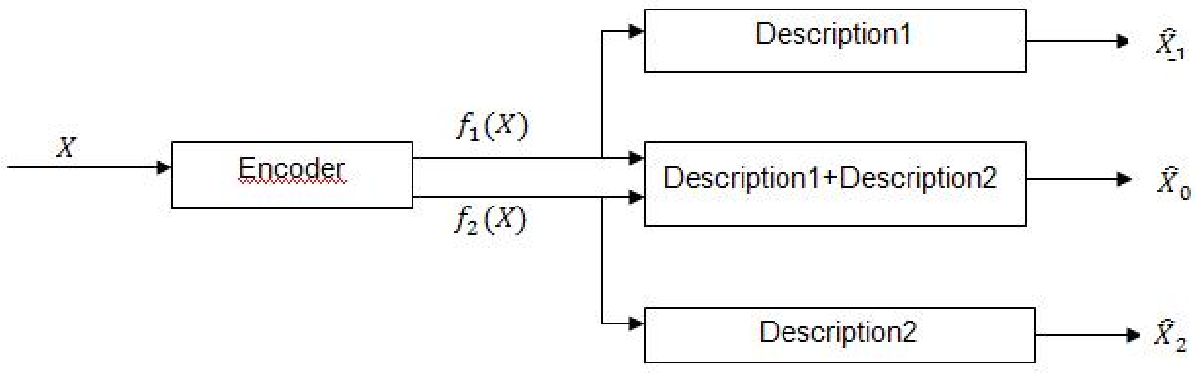

Figure 1. The channel splitting problem [28].

3.2 MDC Approaches

3.2.1 Multiple Description Scalar Quantization

Multiple description scalar quantization (MDSQ) was first proposed by Vaishampayan [28], who suggested its theory. The MDSQ encoder consists in a standard scalar quantizer and an index assignment. In the standard scalar quantizer, the two quantizers will operate in parallel at the quantization stage[3]. Then the assignment index will split the signal in two descriptions [29].

3.2.2 Multiple Description Transform Coding

In this approach, the Redundancy is applied. At first, the input signal is decorrelated, using a transform. The resulting coefficients are put into pairs, taking into account their variance. Later, a coefficient is sent to description 1 and the other to description 2. Since there is a

[3] Quantization stage is the stage in which loss of insignificant information occurs.

Redundancy in both descriptions if one is lost, some of the information of the lost description can be estimated from the received one. If both descriptions are received then the exact values can be determined in order to take the inverse transformation [29].

3.2.3 Multiple Description Bit Allocation

The MDC presented in this dissertation is based on the multiple description bit allocation (MDBA), first presented by Manuela Pereira in [28]. MDBA is based in discrete wavelet transform (DWT), which is an advantage since it has a 3D scan-based DWT, which allows the development of a strip-based MDC. In the MDBA approach, the video is given to the encoder which produces two bitstreams with the same rate. Then, these bitstreams are sent into three decoders over two noisy channels. If both descriptions are received, the central decoder will receive information from both descriptions. If only one description is received, the side decoder will receive the information from one of them. It is also introduced in this scheme the explicit redundancy, which permits each sample to be coded with different redundancy every time that it is transmitted across the network.

In [28], the author faced two problems:

1. If in a description, one of the subbands is rightly encoded then in the next description it will be coarse encoded. Which is a drawback, since the division of subbands into redundant subbands may affect the performance of MDC.

2. The other problem was which quantity of redundancy should be used in the different descriptions. In order to solve this problem, it was considered the model and the state of the channel.

So, the generation of the descriptions (two) were made by taking into account three conditions [28]:

1) The central decoder has to reconstructed the original sequence from two descriptions with minimal central distortion D_0.

2) A balanced MDC encoder must generate two descriptions each with a side rate R_1= R_2= R_T/2, where R_T is the total rate and R_1, R_2 are the side rates .

3) When the channel is noiseless, the side decoders must reconstruct the original sequence from a single description with a side distortion $D_1 \leq D_M$ and $D_2 \leq D_M$, where D_M is the maximal distortion.

3.3 Video Streaming

With the continuous growth of the internet, many have been the tried techniques in order to send files over the network. One of the major problems that the researchers have been facing with is how to send a video signal/file across the network. In this way, when the first P2P applications appeared, the first attempt [1] was to send the entire video across the network and wait for the user on the other side to receive it. Nonetheless, they faced with hours of delay, waiting for the video file to be entirely downloaded, so that the other user could watch it. In order to deal with this problem, a new technique called Video Streaming was developed.

Video streaming consists in compressing video files into packets, and sending them up to the receiver. Then, the receiver can watch the video as it is being received [30]. When there is too much traffic across the network, many delays of the video can happen. To solve this problem, the buffering technology was developed. "*Buffering is the process where a large number of packets are collected before the video will begin*" [31].

In this way, the video streaming will solve the problem stated above in two ways [30]:

1) The video is compressed into a smaller size and then sent across the network.

2) Even though, a specific player is needed, the receiver can watch the video, while receiving it.

Nowadays, many are the uses of the video streaming. The movies, songs and TV shows that appear in our homes in the TV are sent by video streaming. The websites that have many movies and songs stored use also video streaming to display them.

Recently, even the universities are joining the video streaming, in order to expand their learning distance programs, many are the people that are joining these programs so that they can change their careers or improve their knowledge without having to go to the University or even have to receive a video DVD by mail. With video streaming, they can learn in real time.

In order to improve the medicine students' training, the hospitals are joining video streaming, so that they can learn while an operation is running, or a patient is being cured [31].

3.4 Conclusions

The problem statement and the most used multiple description coding were presented in this chapter. Then, it was introduced the multiple description coding MDBA [28], which is the most important one, since it is in this one that the REGPROT encoder - introduced in chapter four - is based. Finally, the video streaming is described, being possible to conclude that multiple description coding seems a good solution for video streaming, because, as stated in chapter one, MDC will decrease the impact errors that appear in the transmission, and the reception of the video is always ensured.

Chapter 4

Multiple Descriptions Coding Over P2P Networks

4.1 Introduction

With the continuous growth of the popularity of P2P applications, much attention has been paid to P2P streaming by the research community.

As stated in chapter two, P2P streaming is a method for multicasting or broadcasting media, for example audio or video, over the Internet using P2P network. The aim for this approach is to allow bandwidth-consuming streaming media to be delivered to a large number of consumers without unnecessary network congestion [14]. In this way, a lot of research work has been made over the last years to discover how multiple descriptions coding behave in P2P network, e. g. [32], [33], [34], [35].

Many have been the discussions about which are the best parameters to obtain a better video quality. Some say that the only factor that matters is the upload bandwidth, i.e., a peer that contributes with a high upload bandwidth receives more descriptions and, consequently, a better video quality. Therefore, the video quality is only determined by the number of received descriptions and not by which description is received [32]. Others say that the availability and bandwidth are important characteristics to have into account.

In [33], it is introduced a wavelet-based video MDC that fits the criteria for P2P networks. Here, the descriptions are put into senders peers, according to their availability and bandwidth.

In this way, the most important information should be sent to the most reliable peers. In [34], the researchers say that the important characteristics are the scalability of the P2P network and the error resilience. So, they introduce an MDC with spatial-temporal hybrid interpolation for video streaming, where two streams of low resolution are added to improve the scalability and the error resilience. A new method of MDC for P2P streaming called Flexible Multiple Description Scalable Coding (F-MDC) is studied in [35], being shown that the changing of the Redundancy level of each description on the fly leads to a better performance of a P2P streaming system than the approach where this parameter is fixed.

In this dissertation a model of MDC called REGPROT is proposed, as stated previously in chapter one. This new model proves that changing the Redundancy on the fly is indeed effective, and that varying the Bitrate on the fly outperforms the other. In addition, an extra contribution from this model is that it is viable to achieve a video with better quality when the Redundancy and the Bitrate are simultaneously changed on the fly. In order to prove this a P2P experimental test bed was developed, which shall be presented forwards followed by the description of the proposed MDC.

4.2 Description of the P2P Experimental Test Bed

In this section, it is proposed a P2P experimental test bed for real-time video streaming. To design this experimental test bed it was taken into account the low cost of the resources, the scalability and the reliability of the P2P network.

As it is shown in Figure 2, there are six peers (from Peer1 to Peer6), one client, a main switch, (Switch1) through which all information will pass, and a secondary switch (Switch2). Here all the peers will act as servers.

It was also taken into account that in real-time one does not know which operating system (OS) a peer will have, so Peer1, Peer3 and Peer5 have the OS Windows XP and Peer2, Peer4 and Peer6 have the OS Ubuntu Server 8.10. The client has both so it will decide which one it will use.

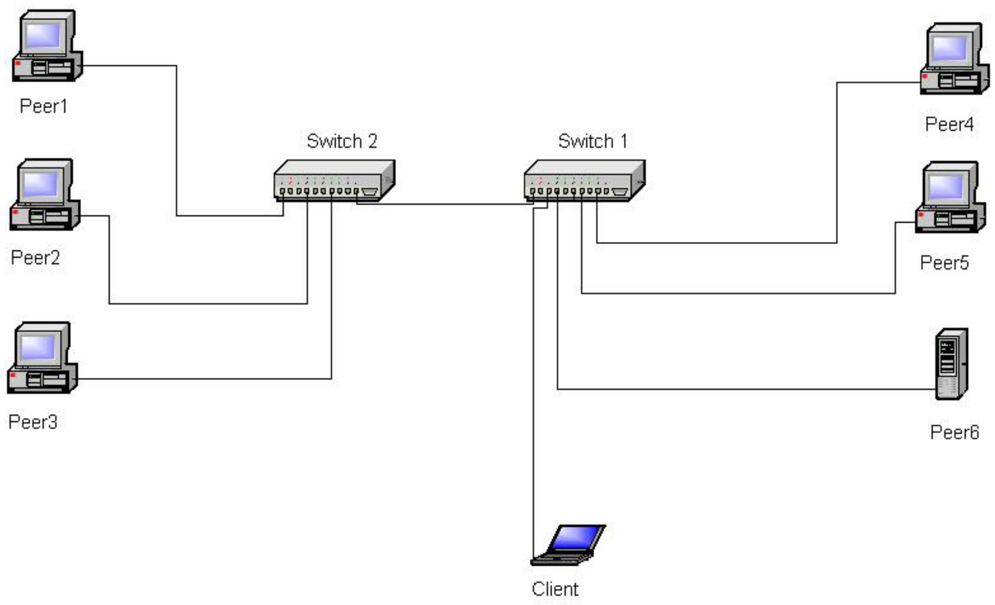

Figure 2. P2P Experimental Test bed.

Due to the peers frequent going down and the problems that might happen with uplink bandwidth, the video was encoded into multiple descriptions, and distributed into the peers (Peer1 up to Peer6).

In this way, the client will make its request to the main switch (Switch1) that will send its requests directly into Peer4, Peer5, Peer6 and Switch2. Switch2 will send the request into Peer1, Peer2 and Peer3. Then, after all the peers have responded, the client will choose, based on its uplink bandwidth, which peer will it download from. Finally, the chosen peers will start to transfer the data into the client.

This experimental test bed has the following advantages [36]:

1) If a peer goes down, only a single description will be lost and that will have limited impact on the video quality;

2) Splitting a video into multiple descriptions and distributing it into multiple server peers will reduce the load on each serving peer, which is an important factor for P2P application;

3) Not storing a full video into one server peer will prevent a client to have illegal access to this video.

4.3 Multiple Description Video Encoder

The video encoder used to split the video into multiple descriptions was developed by Ângelo Arrifano et al. in [39] and it is named REGPROT. At first, it performs a temporal motion compensated wavelet transform and then a multiple description bit allocation as in [36] and [37]. The total bit rate is efficiently distributed between two descriptions by the multiple description bit allocation, based on a Redundancy parameter. That parameter tunes the Redundancy between the descriptions based on present channel features. At the end, with the bit rates produced in the bit allocation module the coder produces JPEG 2000 compatible code-streams provided with error detection capabilities [39].

The modified JPEG 2000 error detection abilities allow the performance of central decoding, which uses all the available description information arriving at the decoder, even if it has errors originated from communication using unreliable channels.

The MDC scheme proposed in this dissertation uses the REGPROT encoder, which will be adapted to P2P transmissions. That adaption leads to two modifications:

1. Firstly to the modification of the descriptions, that may be two, four or eight. To make this choice it might be used the idea present by Antonio Ortega et. al. in [43]. In [43], the authors start with a

random number of descriptions N = 2, or N = 4, or N = 8. Whenever the condition of the channel changes, i. e., if the packet loss rate decreases the level of Redundancy shall decrease as well as the number of descriptions. In this way, if the rate $R_k = \frac{R}{N} + \frac{1}{2}\left(i - \frac{N-1}{2}\right) * \log_2 P$ [4] is negative or zero, it means that there are samples that shall not be sent. Consequently, the number of descriptions shall be changed to $N - k$, where $k = 0, \dots N$.

2. Secondly, the modification of Redundancy is made by taking into account [35]. In [35], the authors start with a medium Redundancy of 42% and then when the packet loss decreases the Redundancy falls to 33%. When the packet increases again, the Redundancy is changed to 51%

In order to test the proposed scheme in the experimental test bed, the video is going to be splitted by REGPROT into multiple descriptions, then the multiple descriptions, which may be two, four or eight, are going to be distributed among the six peers. The client will make its request and later while he/she is receiving the descriptions, the REGPROT decoder is going to decode the description on the client side at the same time that he or she is receiving it, i.e. in real time.

4.4 Performance Assessment

In this section, it is presented a performance assessment of the proposed experimental test bed with video streaming.

The tests were made for two, four and eight descriptions. The test of MDC with two descriptions is done to compare the performance of the F-MDC [35] with the REGPROT encoder, and to show that this MDC scheme is a good option for video streaming in peer-to-peer network. Later, the tests were done

[4] Where R is the bit rate, N is the number of descriptions, i is the number of GOP received and P is the packet loss probability.

for four descriptions. In here, three methods were compared. At first, it was compared the changing of the Redundancy on the fly with the changing of the Bitrate on the fly, then it was compared these two methods with changing simultaneously the Redundancy and the Bitrate on the fly. Finally, to confirm the results given by four descriptions, the same tests for MDC were done with eight descriptions.

4.4.1 MDC with Two Descriptions

Firstly, the encoder REGPROT is compared with the most effective MDC, as far as I know, called F-MDC [35] at 3 spatial and 4 temporal decompositions levels for some fixed rates, with N=2 descriptions, to test the performance between them. F-MDC consists in the generation of multiple description originated by a single scalable video bitstream.

In this way, as in [35], the Redundancy in each description is defined as the ratio of the amount of redundant bits that are not used and the amount of bits that are used when all descriptions are received: $\frac{Rnotused}{Rused}$. Results and analysis are presented in the following for Redundancy levels between 20% and 50%.

Figure 3, Figure 4 and Figure 5 show the PSNR obtained when coding quarter common intermediate format (QCIF) Foreman, with each description encoded at 100 kbps, 200 kbps and 300 kbps, respectively, and Figure 6 and Figure 7 show the PSNR obtained when coding QCIF Akiyo, at 50 kbps and 100 kbps respectively.

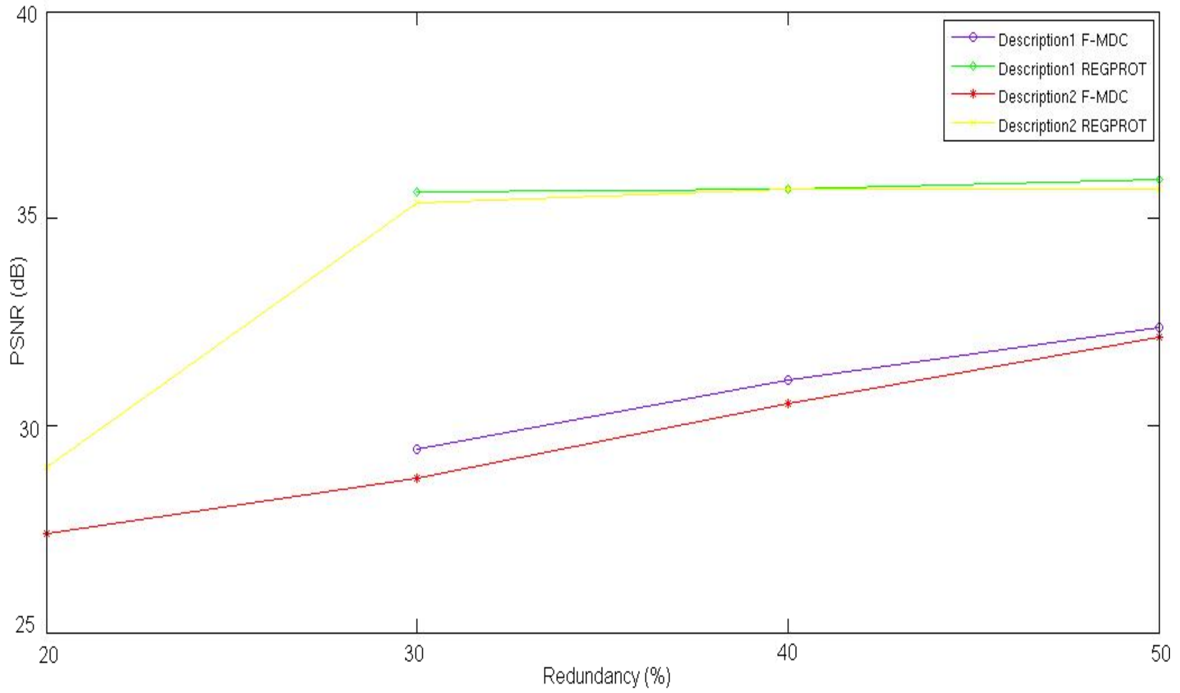

Figure 3. Foreman in format QCIF encoded at 100 Kbps.

As it may be seen in Figure 3, when each description is encoded at 100 kbps Bitrate, REGPROT outperforms F-MDC between 3.6 and 6 dB in description1, and in description2 between 1.6 and 6.6 dB.

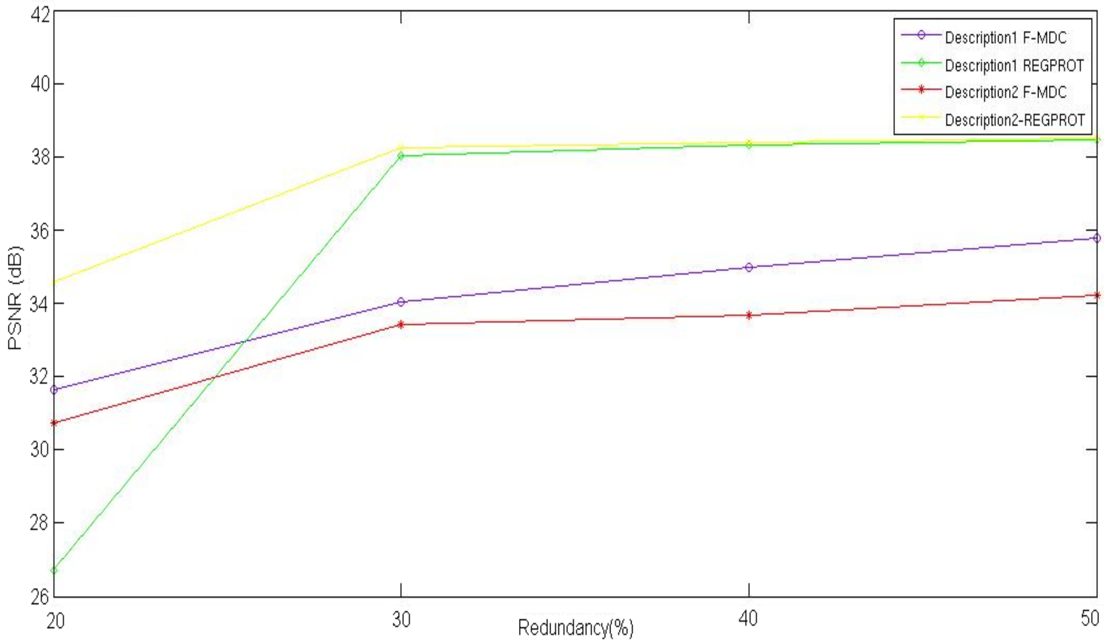

Figure 4. Foreman in format QCIF encoded at 200 Kbps.

Figure 4 shows the PSNR when each description is encoded at 200 kbps. In here, F-MDC outperforms REGPROT in description1 for the first Redundancy level (20%) in 4.9 dB, but for the others Redundancy levels (30%, 40%, 50%) the REGPROT outperforms F-MDC between 2.7 and 4 dB. While for the description2, the REGPROT outperforms F-MDC between 4.3 and 4.8 dB in all Redundancy levels.

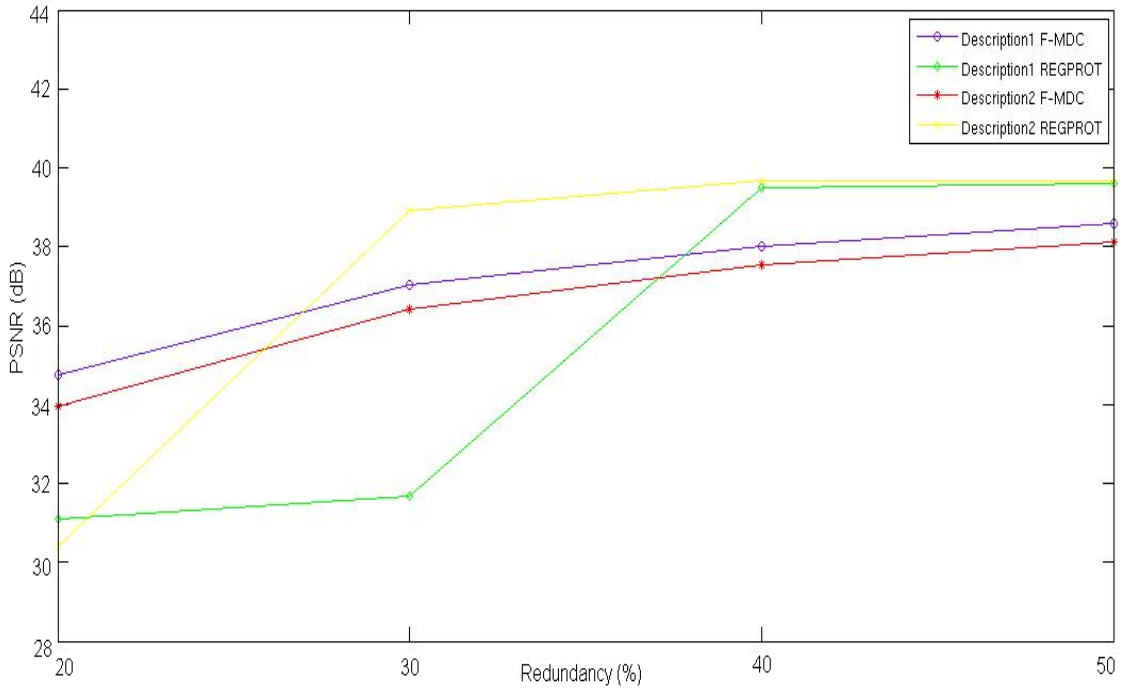

Figure 5. Foreman in format QCIF encoded at 300 Kbps.

In Figure 5, it may be seen that when each description is encoded at 300 kbps, for the first two levels of Redundancy (20%, 30%) in description1, the F-MDC outperforms REGPROT between 3.1 and 5.9 dB, and in the last two Redundancy levels (40%, 50%), the REGPROT outperforms F-MDC between 1 and 1.5 dB. In what refers to description2, the F-MDC outperforms REGPROT in the first Redundancy level (20%) by 3.6 dB, and REGPROT outperforms F-MDC between 1.6 and 2.5 dB in the other levels of Redundancy (30%, 40%, and 50%).

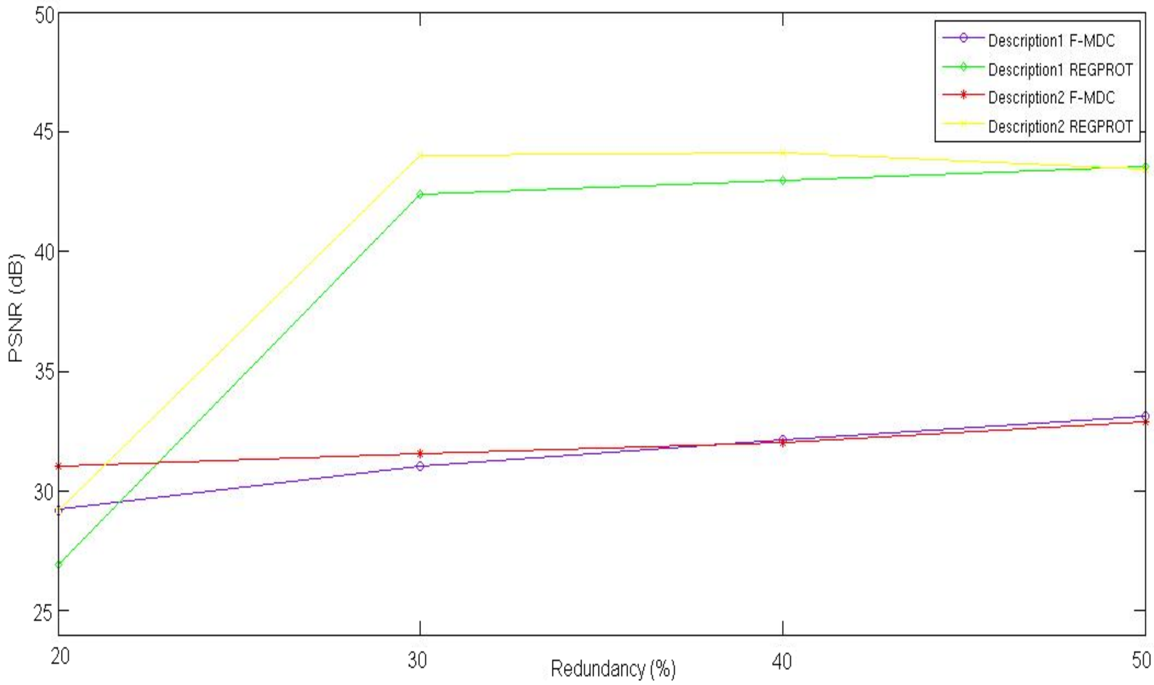

Figure 6. Akiyo in the format QCIF encoded at 50kbps.

In what concerns a low sequence (Akiyo), it may be seen in Figure 6 the PSNR when each description is encoded at 50 kbps. In here, for the first Redundancy level (20%), F-MDC outperforms REGPROT in both descriptions between 1.8 and 2.3 dB, while for the other levels of Redundancy (30%, 40%, 50%) REGPROT outperforms F-MDC in both descriptions, between 10.3 and 11.3 dB in description1 and between 10.5 and 12.1 dB in description2.

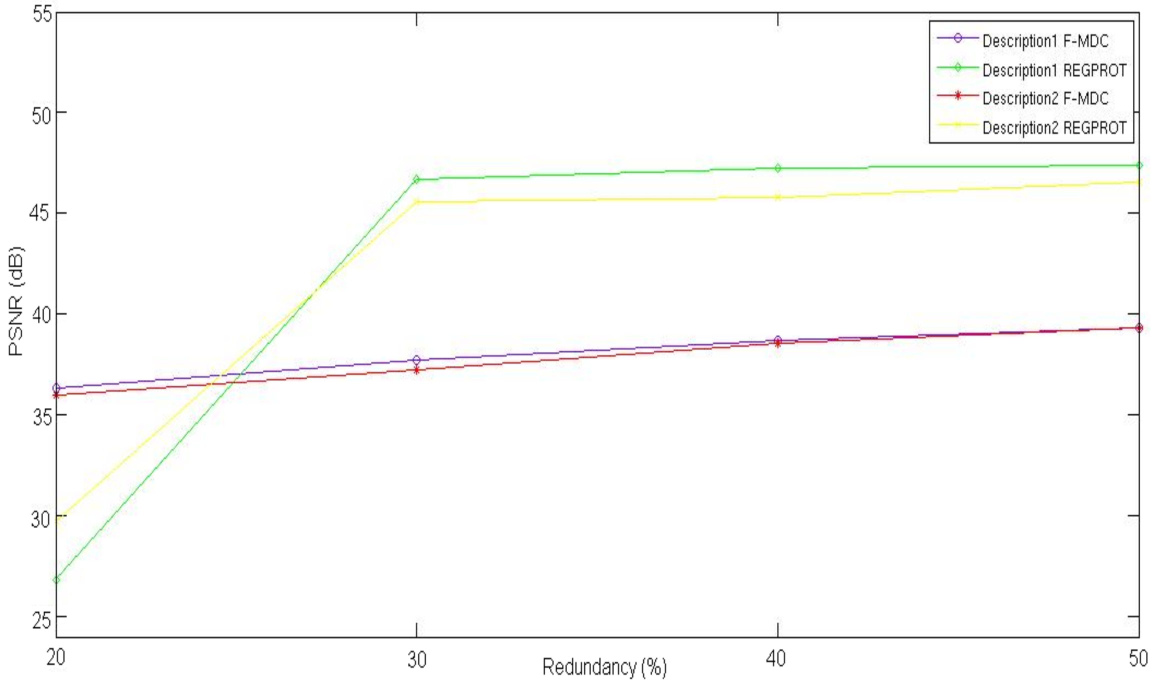

Figure 7. Akiyo in the format QCIF encoded at 100 kbps.

In Figure 7, it may be seen that, when each description is encoded at 100 kbps, for the first Redundancy level (20%) F-MDC outperforms REGPROT between 6.3 and 9.5 dB in both descriptions. It can also be seen that REGPROT outperforms F-MDC between 7.9 and 8.9 dB in description1, and between 7.2 and 8.4 dB in description2, in 30%, 40%, 50% Redundancy levels.

Then the performance of the purposed experimental test bed is tested. The Foreman sequence in QCIF format is encoded with the REGPROT encoder, where 3 spatial and 3 temporal decomposition levels are used for 288 frames at 30 frames per second (fps). Each group of pictures (GOP) has length of 16, and the frames are put into packet size that vary between 1400 and 1432 bytes. The Bitrate of each description (N=2) is set to 100 Kbps.

There are two peers which have the encoded video. Each peer has a bandwidth of about 100 Mbps. In order to determine from which peer the client will download from, the LimeWire [23] P2P application is used.

With the LimeWire application and having the client characteristics to connect as an Ultrapeer [40], it will have access to the information about the uplink bandwidth of the peers, and it will, then, decide where it will download from.

As in [35], we tested the purposed experimental test bed for:

1) different rate of Packet Loss: 3% and 18%;

2) two full play time of video segment: t = 20 s.

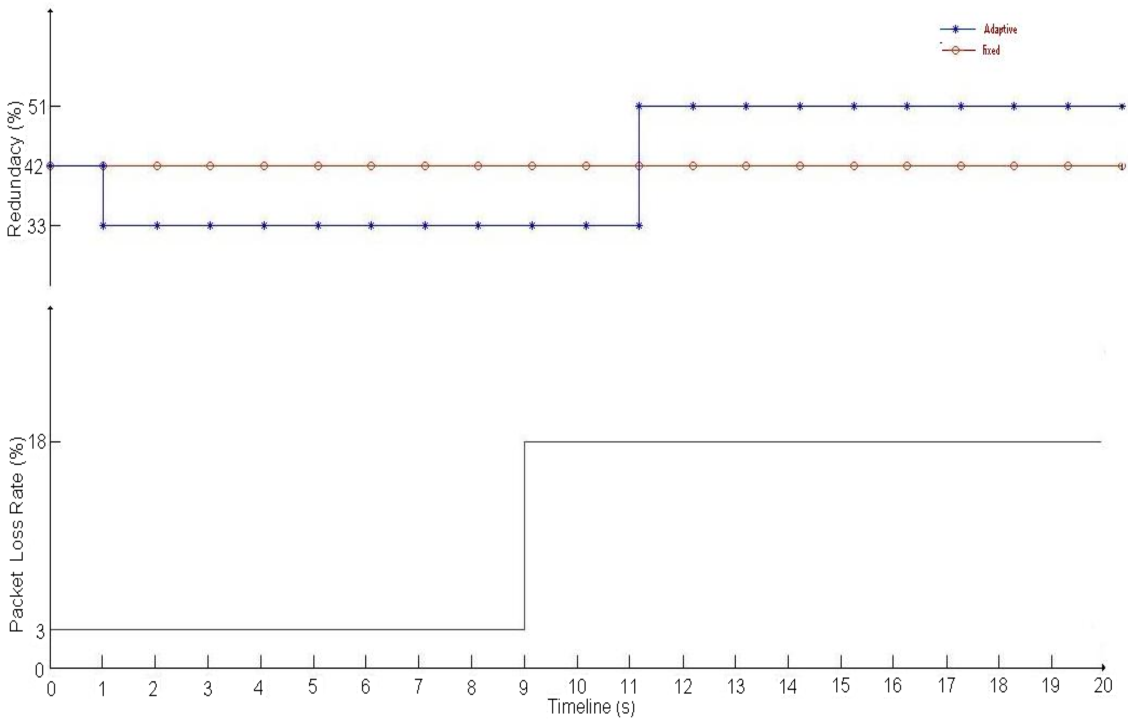

Figure 8. Loss rate and Redundancy adaptation in time.

Figure 8 shows the change of the packet loss rate and the change of Redundancy levels in time. Here, it may be seen that the system was started with an average Redundancy (42%). When one loss period delay t = 1 s is detected, the sender adapts the Redundancy level to 33%, based on the 3% packet loss rate. At time t = 10 s, and after a one loss period delay, the packet loss rate changes to 18% and the sender will adapt the 51% Redundancy. In what refers to the fixed Redundancy level, it was fixed in 42% along the two full play time video segment t = 20 s.

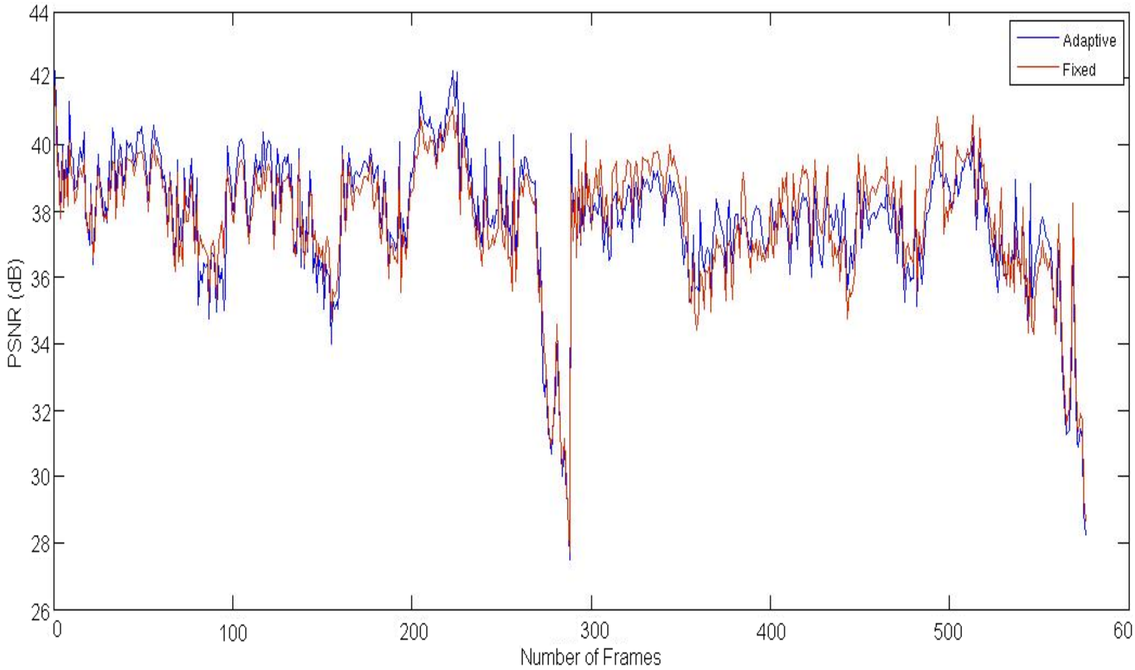

Figure 9. PSNR for every frame in the fixed and adaptive system.

It may be seen in Figure 9, that in the purposed experimental test bed the adaptive system outperforms the fixed one by 1.09 dB in the first half, and the fixed system outperforms the adaptive in the second half by 0.87 dB of the test.

PacketsDump software [41] is used to determine the Packet Loss rate as well the number of packets per second sent containing GOPs (see Figure 10).

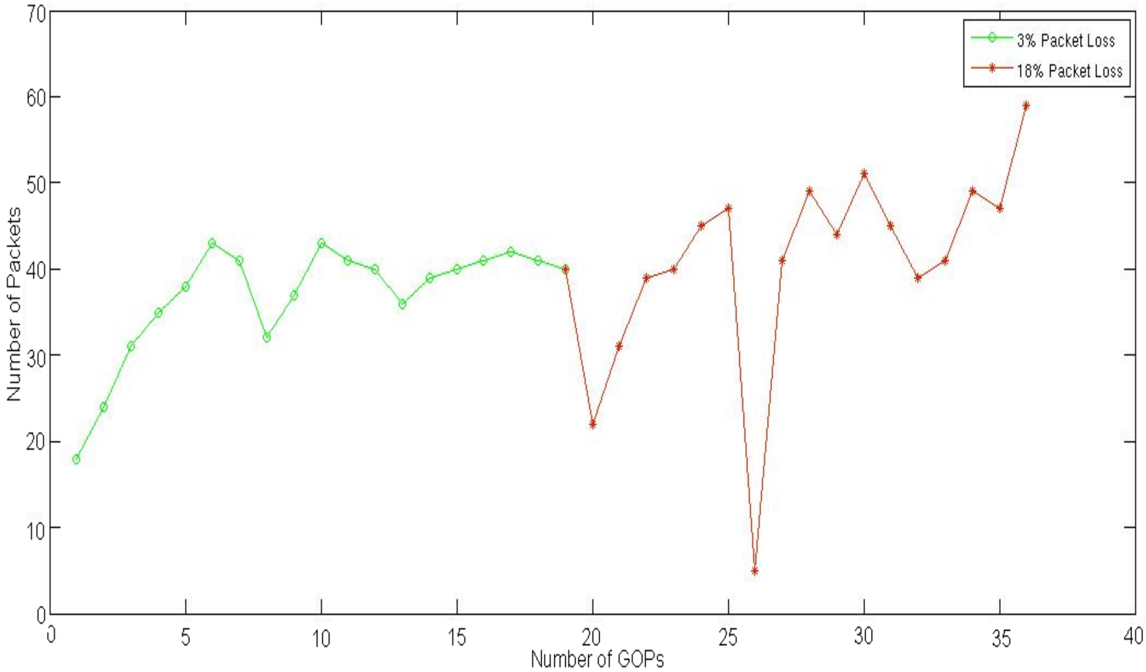

Figure 10. Number of Packets per second that each GOP is sent.

The Packet Loss rate was found by analyzing the output of the trace file given by the PacketsDump software and the number of packets per second that each GOP is sent by analyzing the statistics window of the PacketsDump software.

As it may be seen in Figure 11, even with 18% of packet loss there is always, at least, one description that is received.

In this way, the video quality will not be compromised, because as it has been told before losing a single description will have limited impact on the video quality.

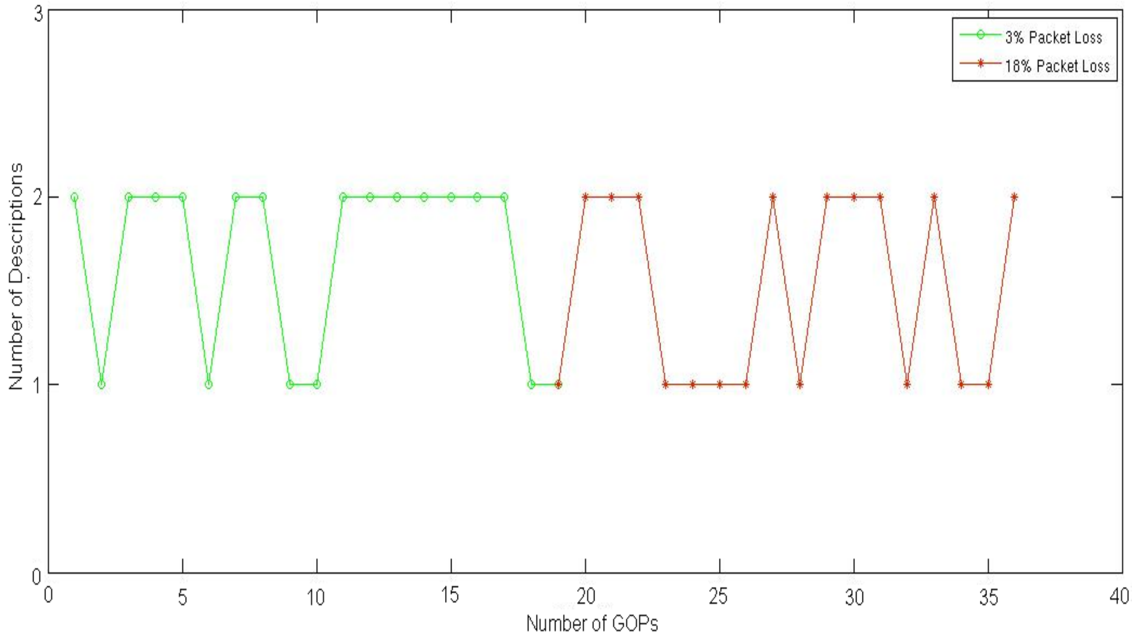

Figure 11. Number of descriptions received for each GOP.

4.4.2 MDC With Four Descriptions

In this section, the proposed experimental test bed will be tested for four descriptions. There will be four peers that have the encoded video. Each peer has a bandwidth of about 100 Mbps. The LimeWire P2P application is used, in order to allow the user to see the uplink bandwidth of the peers and to choose from which peer he/she wishes to download from. The tests will be made for:

1. 3% and 8% of packet loss

2. for two full play time of video segment: t = 20 s.

The Foreman sequence in common intermediate format (CIF) format was encoded with 2 spatial and 2 temporal levels for 144 frames at 15 fps. Each GOP has a length of 16 and the frames are put into packet sizes that vary between 1400 and 1432 bytes.

In order to identify which is the best way to improve the video quality in the receiver side when we face a scenario of packet loss, three experiences were made:

1. Varying the Redundancy on the fly - In here, the system was started with average Redundancy (46%). When one loss delay t=1sec is detected, the sender adapts the Redundancy level to 31%, based on the 3% of packet loss. At time t = 10 s, and after the detection of the delay t = 1 s, the packet rate loss changes to 18%, so the sender will adapt the system into 43% (see Figure 12).

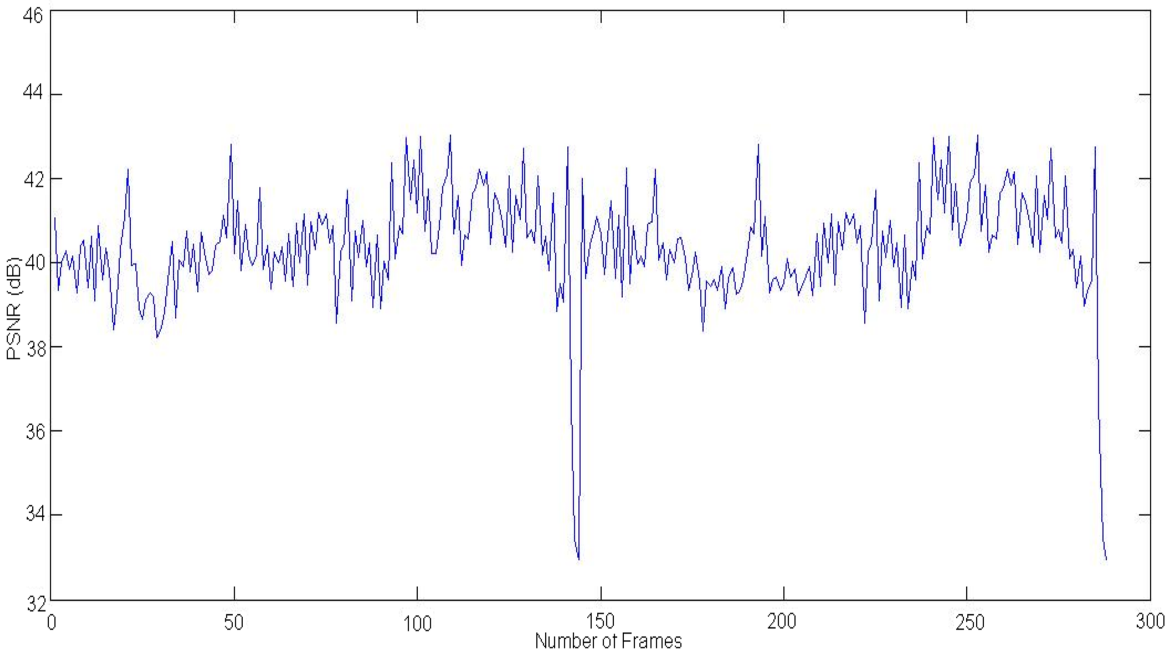

Figure 12. PSNR for every frame in the Redundancy adaptive system.

2. Varying the Bitrate on the fly - for this experience the system started with each GOP encoded at 1532 Kbps of Bitrate. Each time that one loss delay t = 1 s is detected; the sender adapts the

GOP Bitrate to 1500 kbps, based on the 3% of packet loss. For the second play time of the video, and after the delay, the sender adapts the GOP Bitrate for 1700 kbps to 18% of packet loss (see Figure 13).

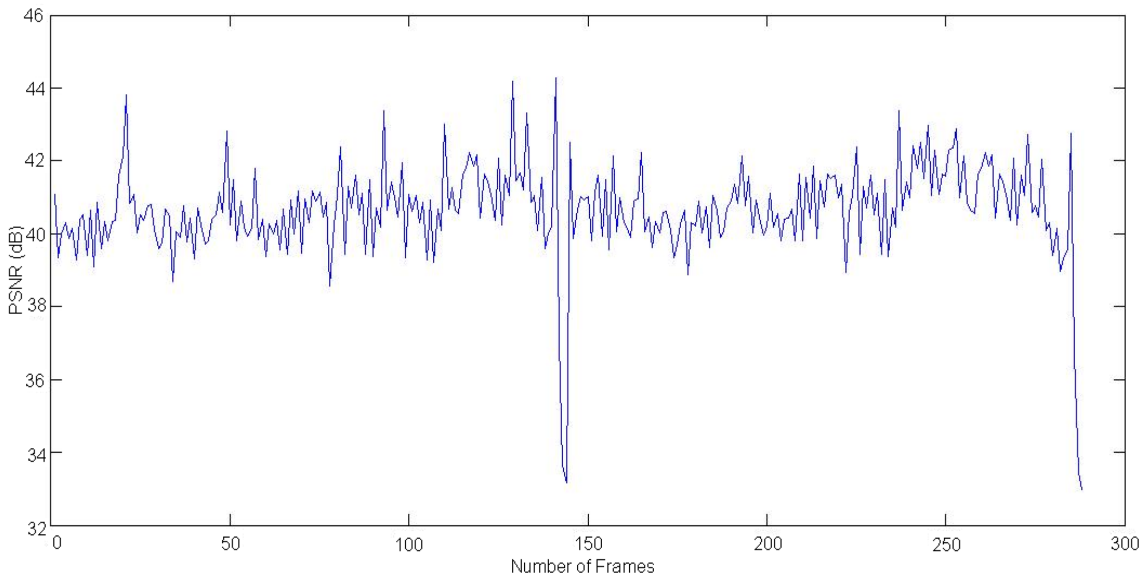

Figure 13. PSNR for every frame in the Bitrate adaptive system.

3. Varying the Redundancy and the Bitrate on the fly - In here, the two experiences mentioned above are joined. In this way, the system starts with an average Redundancy (46%) in which each GOP is encoded at 1532 kbps. When one delay is detected, the sender adapts the Redundancy level to 31% and each GOP is encoded at 1500 Kbps, based on the 3% of packet loss. Then, when the next delay is detected, the packet loss rate changes to 18%, so the sender adapts the Redundancy level to 43% and encodes each GOP at 1700 kbps (see Figure 14).

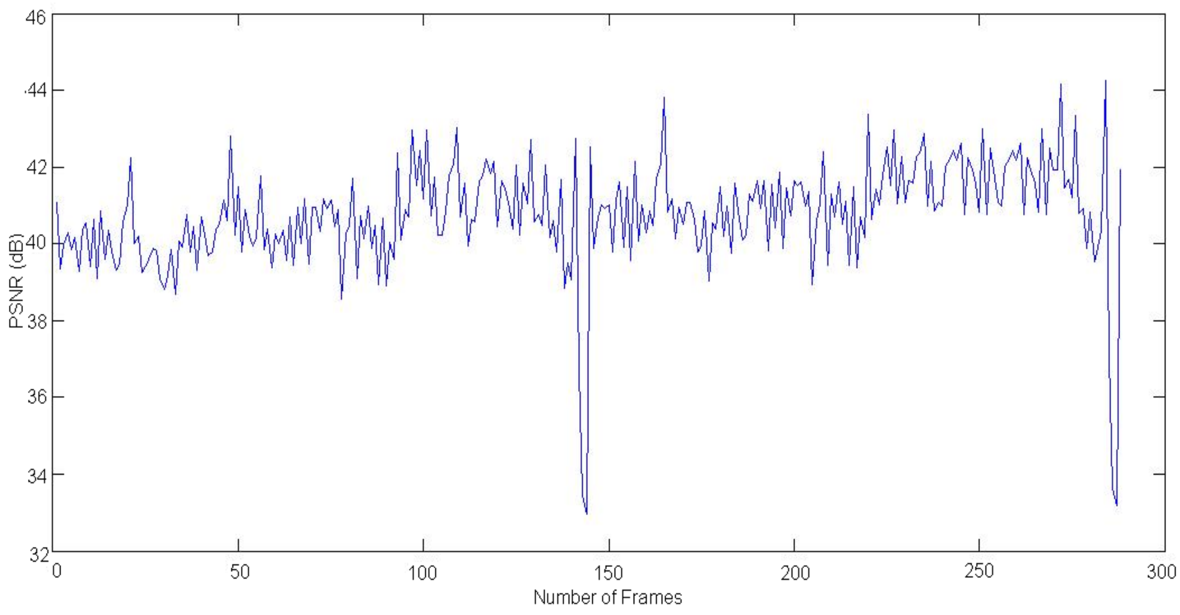

Figure 14. PSNR for every frame in the Redundacy plus Bitrate adaptive system.

In the following, we present a study regarding the varying of the Redundancy versus the varying of the Bitrate on the fly.

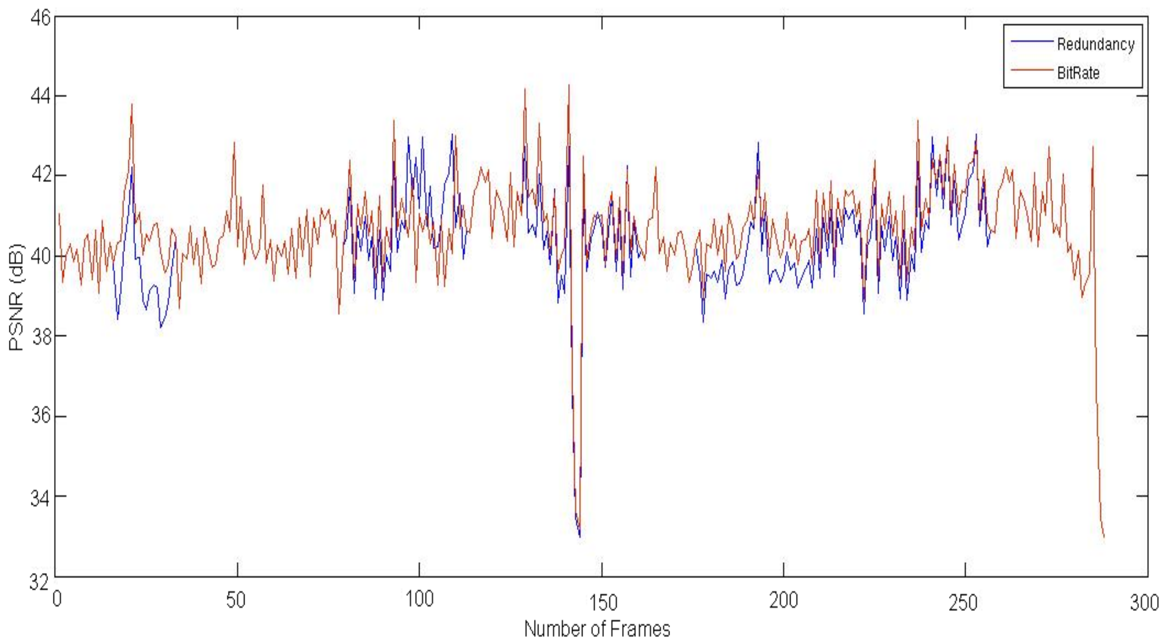

Figure 15. PSNR for the Redundancy adaptive system versus Bitrate adaptive system.

As we can see in Figure 15, for the first half when the packet loss rate is of 3%, the adaptive Bitrate system has a gain of about 1.57 dB when compared to the adaptive Redundancy system. In what refers to the second half, when the packet loss rate is set to 18% the Bitrate adaptive system also outperforms the Redundancy adaptive system with a gain of 1 dB.

In this way, changing the Bitrate on the fly is more effective than changing the Redundancy on the fly.

In the following, we present a study regarding the varying of the Redundancy versus the varying of the Bitrate on the fly versus the varying of the Redundancy plus Bitrate on the fly.

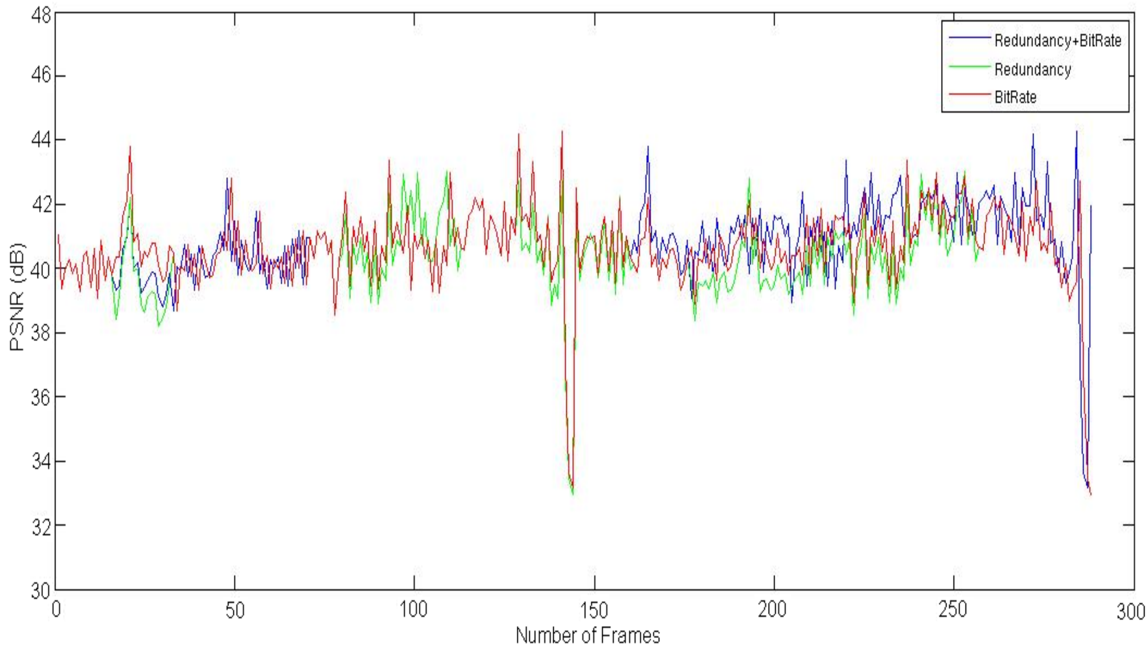

Figure 16. Redundacy plus Bitrate adaptive system versus Redundancy adaptive system versus Bitrate adaptive system.

For the first half of the experience, when there is a 3% of packet loss rate, it is possible to see that both adaptive systems Redundancy plus Bitrate and Bitrate have the same performance, and that both outperform Redundancy one at 1.57 dB. For the second half of the experience the adaptive system Redundancy plus Bitrate outperforms the Bitrate adaptive system having a gain of 1.95 dB and outperforms the Redundancy adaptive system in about 2.57 dB (see Figure 16).

In this way, changing the Bitrate on the fly is better, than only changing the Redundacy. Nevertheless, when the changing of the Bitrate on the fly is compared with changing the Bitrate and the Redundancy on the fly, it may be seen, in Figure 4, that only for the second half (18% of packet loss) the system that changes both Redundancy and Bitrate outperforms the one that only changes the Bitrate on the fly. So, when there is a large packet loss the system that should be used when there is N=4 descriptions is the Redundancy plus Bitrate adaptive one.

4.4.3 MDC With Eight Descriptions

In this part, the purposed experimental test bed will be tested for eight descriptions. The video encoded will be distributed into six peers. Each peer has a bandwidth of about 100 Mbps. The LimeWire P2P application is used, in order to allow the user to see the uplink bandwidth of the peers and choose from which peer he wishes to download from. The tests will be made for:

1. 3% and 8% of packet loss

2. for two full play time of video segment: t = 20s.

The Foreman sequence in CIF format was encoded with 2 spatial and 2 temporal levels for 144 frames at 15 fps. Each GOP has a length of 16 and the frames are put into packet sizes that vary between 1400 and 1432 bytes.

In order to identify which is the best way to improve the video quality in the receiver side when a scenario of packet loss is presented, three experiences were made:

1. Varying the Redundancy on the fly – For this, the system was started with an average Redundancy (46%). When one loss delay t = 1 s is detected, the sender adapts the Redundancy level to 31%, based on the 3% of packet loss. At time t = 10 s, and after the detection of the delay t = 1 s, the packet rate loss changes to 18%, so the sender will adapt the system into 43% (see Figure 17).

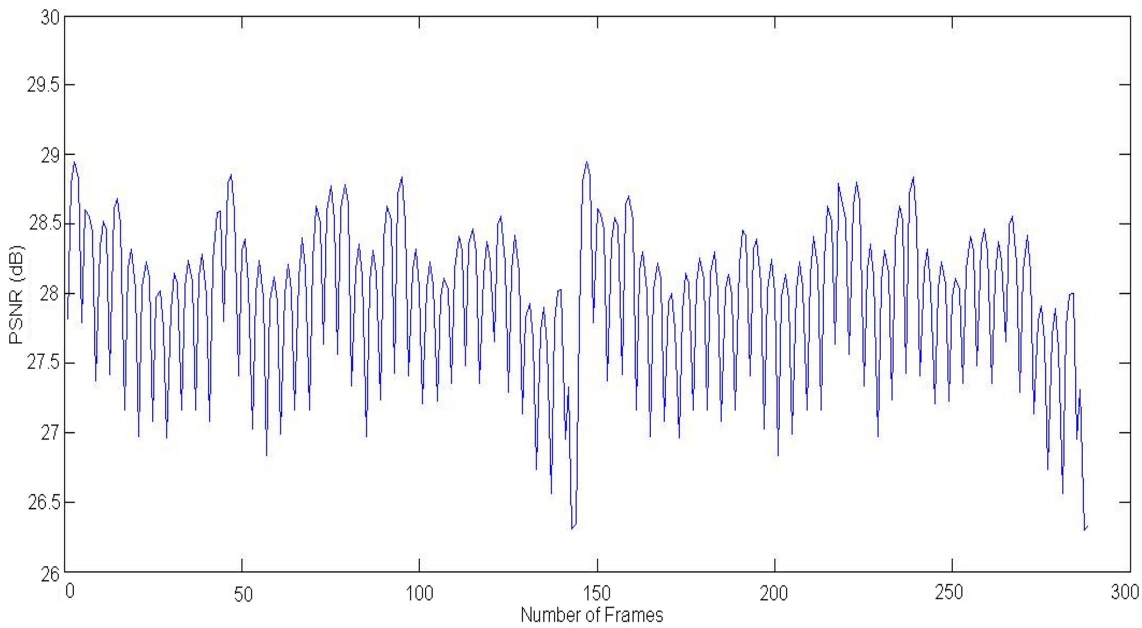

Figure 17. PSNR for every frame in the Redundancy adaptive system.

2. Varying the Bitrate on the fly - for this experience, the system started with each GOP encoded at 1532 Kbps of Bitrate. Each time that one loss delay t = 1 s is detected; the sender adapts the GOP Bitrate to 1500 kbps, based on the 3% of packet loss.

In what concerns the second part of the video, the sender adapts the GOP Bitrate for 1700 kbps for 18% of packet loss after the delay (see Figure 18).

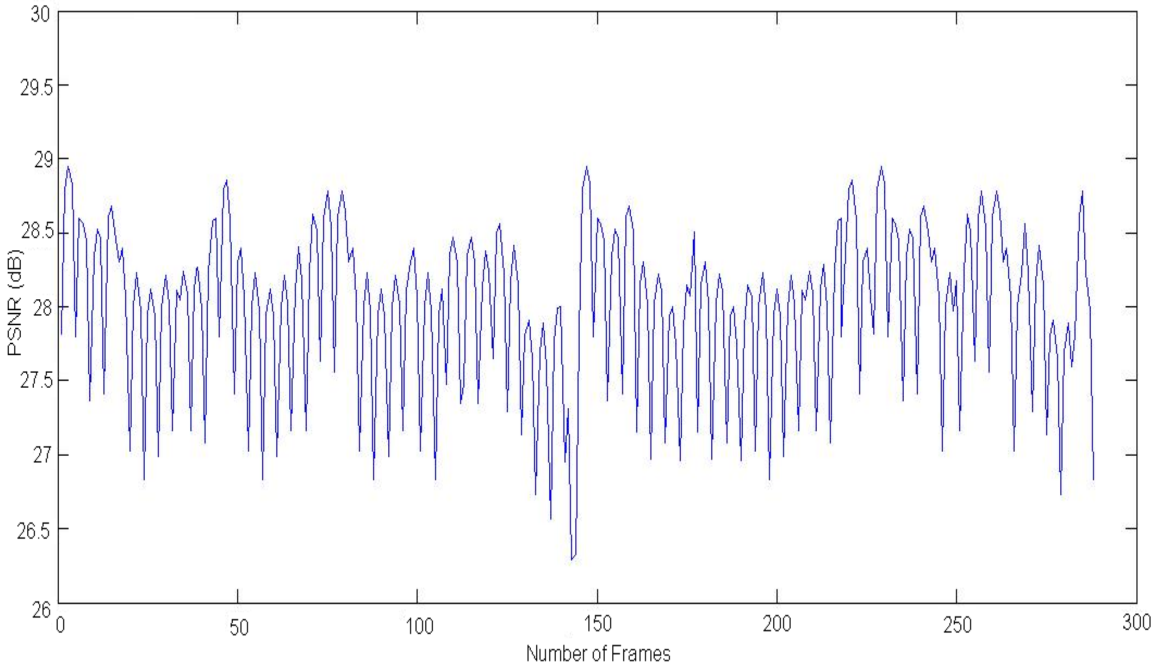

Figure 18. PSNR for every frame in the Bitrate adaptive system.

3. Varying the Redundancy and the Bitrate on the fly - concerning this aspect, the two experiences mentioned above are joined. In this way, the system starts with an average Redundancy (46%) in which each GOP is encoded at 1532 kbps. If a delay is detected, the sender will adapts the Redundancy level to 31% and each GOP will be encoded at 1500 Kbps, based on the 3% of packet loss. Then, the packet loss rate will change to 18%, if other delay comes to be detected. Therefore, the Redundancy level will be adapted to 43% and will encode each GOP at 1700 kbps by the sender (see Figure 19).

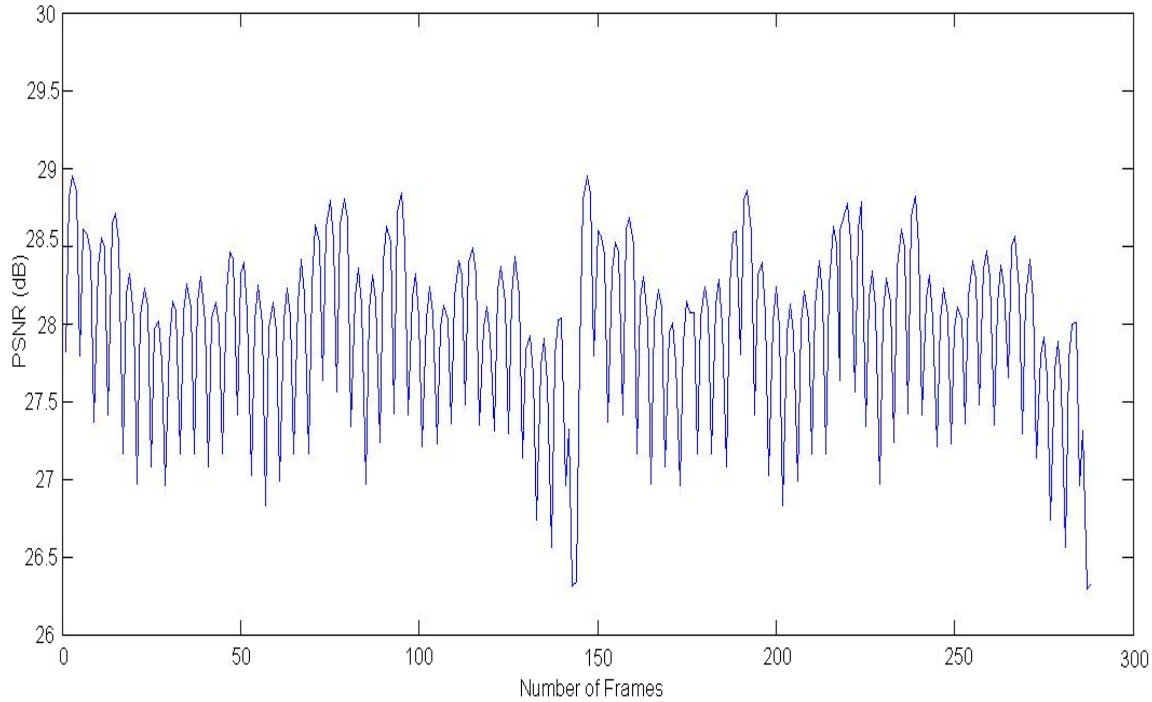

Figure 19. PSNR for every frame in the Redundacy plus Bitrate adaptive system.

In the following, we present a study regarding the varying of the Redundancy versus the varying of the Bitrate on the fly.

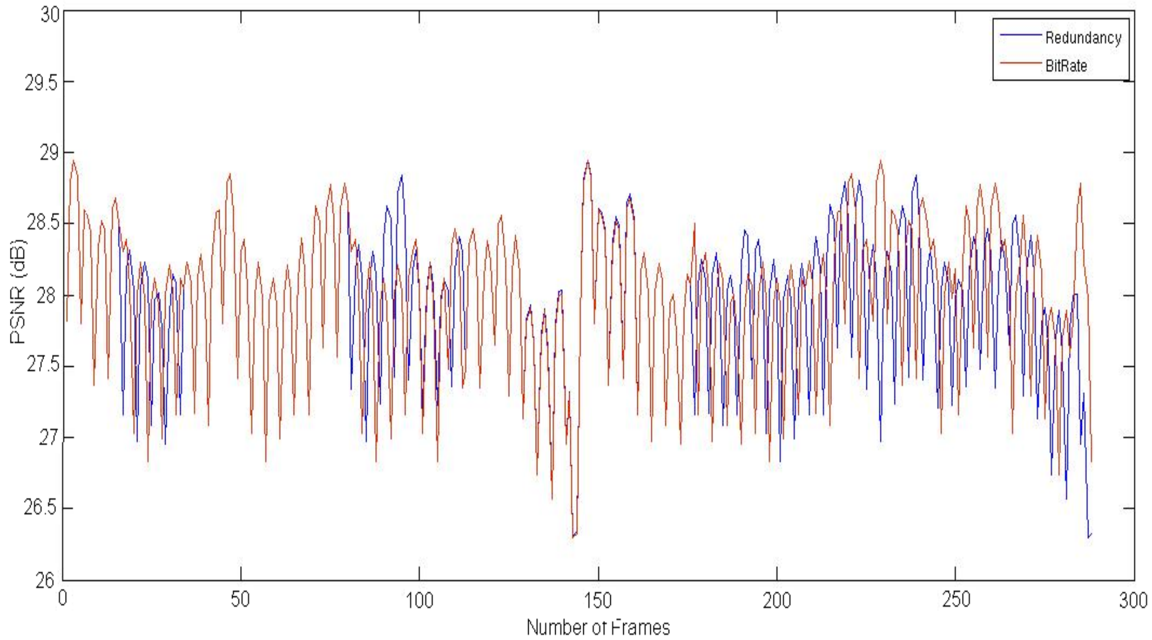

Figure 20. PSNR for the Redundancy adaptive system versus Bitrate adaptive system**.**

As may be seen in Figure 20, when we deal with eight descriptions, in the first half when packet loss rate is of 3%, though in some frames the Bitrate adaptive system equals the Redundancy adaptive system, for other frames the Bitrate adaptive system has a gain of 0.44 dB in terms of PSNR.

In what refers to the second half when the packet loss rate is set in 18% the Bitrate adaptive system outperforms the Redundancy adaptive system with a gain of 1.97 dB.

In this way, changing the Bitrate on the fly is more effective than changing the Redundancy on the fly.

In the following, we present a study regarding the varying of the Redundancy versus the varying of the Bitrate on the fly versus the varying of the Redundancy plus Bitrate on the fly.

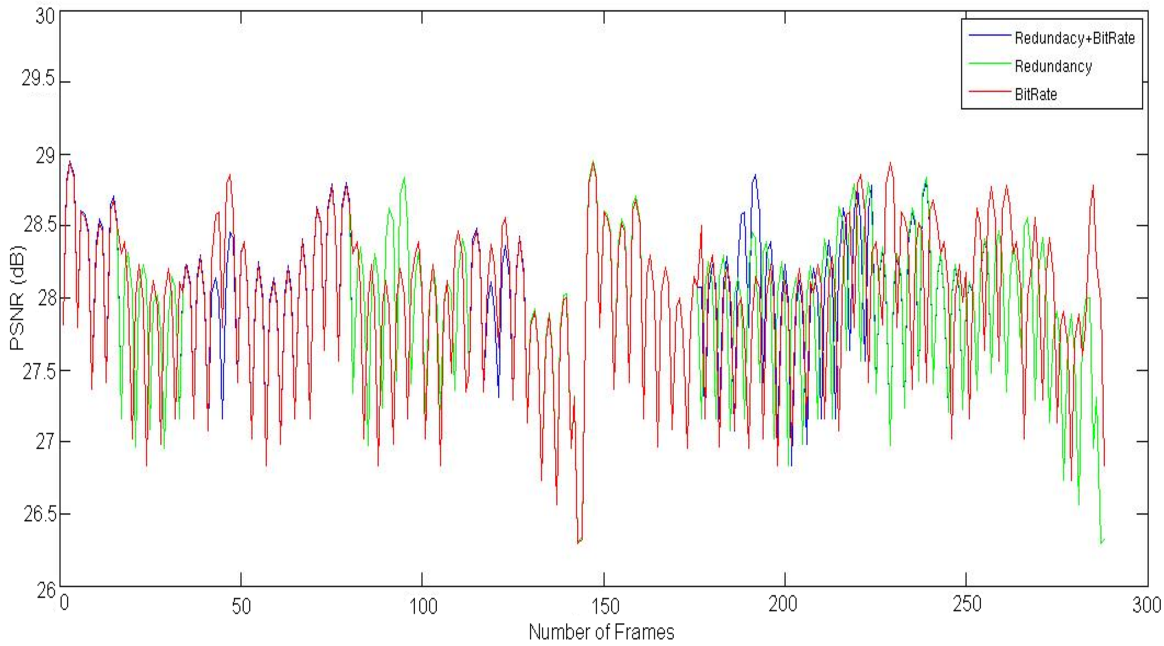

Figure 21. Redundacy plus Bitrate adaptive system versus Redundancy adaptive system versus Bitrate adaptive system.

For the first half of the experience, when there is a 3% of packet loss rate it may be seen that both adaptive systems, Redundancy plus Bitrate and Bitrate, have the same performance, and that both outperform Redundancy one by 0.44 dB. For the second half of the experience the adaptive system Redundancy plus Bitrate outperforms the Bitrate adaptive system, having a gain of 0.98 dB and outperforming the Redundancy adaptive system in about 0.36 dB.

Nevertheless, as it may be seen in Figure 21, the Regprot encoder has a low performance when decoding the GOP for eight descriptions, it can also be seen that changing the Bitrate on the fly is better, than only changing the Redundacy. However, when the changing of the Bitrate on the fly is compared with changing of the Bitrate and the Redundancy on the fly, it can be checked, in Figure 21, that only for the second half (18% of packet loss) the system, which changes both Redundancy and Bitrate, outperforms the one that only changes the Bitrate on the fly. So, if there is a large packet loss, the system that shall be used when there is N = 8 descriptions is the Redundancy plus Bitrate adaptive one.

4.5 Conclusions

In this chapter, a P2P experimental test bed is proposed, as well as, a brief description of the REGPROT encoder, which is based on MDBA described in chapter three, used to split the video file into multiple descriptions.

It was shown that MDC is a good option when one deals with P2P networks, since even with a high rate of packet loss there is always at least one description that is received. Thus, it was also showed that MDC is more effective when varying both the Redundancy and the Bitrate on the fly.

Chapter 5

Conclusions

5.1 Main Conclusions

In this dissertation, a P2P experimental test bed is proposed, in order to show that MDC scheme is a good option for P2P streaming of video, since even at a high rate packet loss one description is received, which grants the reconstruction of the signal.

In chapter four, it is shown that the MDC scheme (REGPROT) outperforms the most efficient, as far as I know, multiple descriptions coding presented in the literature, which is Flexible Multiple Description Coding [35]. Besides that, it is also proved that distributing Redundancy efficiently among the descriptions makes a stable video quality reachable. In spite of the fact of being a good technique, it was also noticed that when the Bitrate is distributed among the descriptions, a better video quality is achieved than the one got with the distributing of the Redundancy.

In the end of the experiences, a new idea came, and it was proved that even though the Bitrate distributing is a good technique, because it outperforms the Redundancy distributing, when the packet loss rate is too high the better thing to do is to distribute the Redundancy and the Bitrate among the descriptions.

5.2 Future Work

To conclude this dissertation, it remains to suggest future research directions that may result from this research work:

1) Study of the performance and robustness of the proposed scheme for media streaming over peer-to-peer networks using multiple description coding in an open large scale platform, such as the PlanetLab [42].

2) Study of the influence of the structure degree of the P2P system in the robustness of the proposed scheme, namely the use of structured systems based on distributed hash tables (DHT), which support efficient and precise routing but may present a small level of resilience in highly dynamic environments with peers entering or leaving the network versus the use of unstructured systems which present a high resilience in dynamic environments, but with a limited search efficiency.

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