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A contribution to video streaming quality and energy efficiency in optical and Ethernet networks

DOCTORAL THESIS

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Abstract

Multimedia content distribution, both over the Internet and over managed networks dedicated exclusively for this purpose, is having a greater share of total traffic and it is raising every day, especially the video streaming service. Therefore, research interest has arisen to maintain and improve the Quality of Service (QoS) in access and backbone networks, and to offer Quality of Experience (QoE) to end users, by adjusting parameters, configurations, protocols and algorithms to the network requirements regarding the video traffic high demand. The latter is carried out taking full advantage of the network resources at core and access networks such as bandwidth and total installed capacity, and at the same time, proposing efficient transportation alternatives for the aggregated video traffic without increasing services costs and alternatives to reduce operational and maintenance expenditures in an efficient manner, such as reducing the energy consumption.

In the first place, this thesis deals with the study of optical switching proposals in backbone networks regarding multimedia content distribution, particularly studying the Optical Bursts Switching (OBS) Networks proposal from the Telematics Engineering point of view. Therefore, architecture, protocols and algorithms of these networks are studied to make contributions to the video traffic evaluation and alternatives to offer QoS and QoE for multimedia traffic. In this sense, this work contributes with proposals for the evaluation of quality through simulation using traces of real video transmissions, and optical burst assembly and scheduling algorithms proposals to offer QoS and QoE to video traffic in the OBS networks.

Secondly, video traffic patterns at the application and network level are studied to characterize them and carry out studies on energy efficiency and energy consumption savings in the network cards used by streaming video servers and clients that implement the IEEE 802.3az Energy Efficient Ethernet (EEE) standard, both in managed networks, such as IPTV that implements RTP video streaming, and unmanaged networks, such as Internet which implements Dynamic Adaptive Streaming over HTTP (DASH) standard for Over The Top (OTT) services. To reach this goal, testbeds are implemented, and experiments are carried out to conclude how video traffic could benefit from the energy efficiency and achieve a large-scale reduction in energy consumption if the EEE standard is implemented word wide in the Ethernet cards of devices involved in video distribution.

The integrating subject that is addressed in this thesis is the distribution of digital video. Therefore, this work also presents a study on the IP networks video transmission encoding and adaptation, presenting aspects regard the video quality evaluation and the network transmission quality assessment. Therefore, methodologies and testbeds are proposed and

implemented that involves programming and execution of simulators, scripts, algorithms, software and, in general, frameworks to carry out studies for video transmissions in telecommunications networks, covering the broad spectrum of Multimedia Communications. This study could be useful to undertake other necessary studies in the rapid evolution of both, multimedia services demand and new content distribution techniques proposals

Resumen

La distribución de contenidos multimedia, tanto a través de Internet como de redes gestionadas dedicadas exclusivamente a este fin, está teniendo una mayor cuota del tráfico total y crece cada día, especialmente el servicio de vídeo streaming. Por ello, ha surgido el interés de investigar para mantener y mejorar la Calidad de Servicio (QoS) en redes de acceso y backbone, y ofrecer Calidad de Experiencia (QoE) a los usuarios finales; mediante el ajuste de parámetros, configuraciones, protocolos y algoritmos a los requerimientos de la red en cuanto a la alta demanda de tráfico de vídeo. Lo anterior aprovechando al máximo los recursos de la red, tanto en el núcleo y las redes de acceso, como el ancho de banda y la capacidad total instalada, y al mismo tiempo, proponiendo alternativas de transporte eficientes para el tráfico de vídeo agregado sin incrementar los costos de los servicios; y alternativas para reducir los costos operativos y gastos de mantenimiento de manera eficiente, como por ejemplo la reducción del consumo de energía.

En primer lugar, esta tesis aborda el estudio de propuestas de conmutación óptica en redes troncales y la distribución de contenidos multimedia, particularmente la propuesta de Red de Conmutación Óptica de Ráfagas (OBS) desde el punto de vista de la Ingeniería Telemática. Por lo tanto, se estudia la arquitectura, protocolos y algoritmos de estas redes para hacer aportes a la evaluación del tráfico de vídeo y alternativas para ofrecer QoS y QoE para el tráfico multimedia. En este sentido, este trabajo contribuye con propuestas para la evaluación de la calidad mediante simulación utilizando trazas de transmisiones de vídeo reales, y propuestas de algoritmos de scheduling y ensamblaje de ráfagas ópticas para ofrecer QoS y QoE al tráfico de vídeo en las redes OBS.

En segundo lugar, se estudian los patrones de tráfico de vídeo a nivel de aplicación y de red para caracterizarlos y realizar estudios de eficiencia energética y ahorro de consumo energético en las tarjetas de red utilizadas por servidores de vídeo en streaming y clientes que implementan el estándar IEEE 802.3az Energy Efficient Ethernet (EEE), tanto en redes gestionadas, como IPTV que implementa transmisión de vídeo RTP, como en redes no gestionadas, como Internet, que implementa el estándar Dynamic Adaptive Streaming over HTTP (DASH) para servicios Over The Top (OTT). Para alcanzar este objetivo, se implementan *testbeds* y se llevan a cabo experimentos para concluir cómo el tráfico de vídeo podría beneficiarse de la eficiencia energética y lograr una reducción a gran escala en el consumo de energía si el estándar EEE se implementa a nivel mundial en las tarjetas Ethernet de los dispositivos involucrados en la distribución de vídeo.

El tema integrador que se aborda en esta tesis es la distribución de vídeo digital. Por lo tanto, este trabajo también presenta un estudio sobre la codificación y adaptación de transmisión de vídeo en redes IP, presentando aspectos relacionados con la evaluación de la calidad de vídeo y la evaluación de la calidad de transmisión de la red. Por ello, se proponen e implementan metodologías y bancos de pruebas que involucran la programación y ejecución de simuladores, scripts, algoritmos, software y, en general, frameworks para realizar estudios de transmisiones de vídeo en redes de telecomunicaciones, cubriendo el amplio espectro de las Comunicaciones Multimedia. Este estudio podría ser de utilidad para emprender otros estudios necesarios ante la rápida evolución tanto de la demanda de servicios multimedia como de las propuestas de nuevas técnicas de distribución de contenidos

Resum

La distribució de continguts multimèdia, a través d'Internet i a través de xarxes gestionades dedicades exclusivament a aquesta finalitat, està tenint una part més gran del trànsit total i augmenta cada dia, especialment el servei de streaming de vídeo. Per tant, ha sorgit l'interés de la recerca per mantenir i millorar la Qualitat de Servei (QoS) en les xarxes d'accés i troncals, i per oferir Qualitat d'Experiència (QoE) als usuaris finals, ajustant paràmetres, configuracions, protocols i algoritmes als requeriments de la xarxa pel que fa a l'alta demanda de trànsit de vídeo. Aquest últim, aprofitant al màxim els recursos de xarxa al nucli i a les xarxes d'accés, com ara l'amplada de banda i la capacitat instal · lada total, i alhora, proposant alternatives de transport eficients per al trànsit agregat de vídeo sense augmentar els costos dels serveis i alternatives per reduir les despeses operatives i de manteniment d'una manera eficient, com ara reduir el consum d'energia.

En primer lloc, aquesta tesi tracta l'estudi de les propostes de commutació òptica en xarxes troncals pel que fa a la distribució de continguts multimèdia, especialment l'estudi de la proposta de xarxes de commutació de ràfegues òptiques (OBS) des del punt de vista de l'Enginyeria Telemàtica. Per tant, s'estudia l'arquitectura, els protocols i els algoritmes d'aquestes xarxes per fer contribucions a l'avaluació del trànsit de vídeo i alternatives per oferir QoS i QoE per al trànsit multimèdia. En aquest sentit, aquest treball contribueix amb propostes per a l'avaluació de la qualitat mitjançant la simulació utilitzant traces de transmissions de vídeo reals, i propostes d'algoritmes de programació i muntatge de ràfegues òptiques per oferir QoS i QoE al trànsit de vídeo a les xarxes OBS.

En segon lloc, s'estudien els patrons de trànsit de vídeo a nivell d'aplicació i xarxa per caracteritzar-los i realitzar estudis sobre eficiència energètica i estalvi energètic en les targetes de xarxa utilitzades pels servidors i clients de vídeo en streaming que implementen l'IEEE 802.3az Energy Efficient Ethernet (EEE) estàndard, tant en xarxes gestionades com IPTV, que implementa la transmissió de vídeo RTP, com en xarxes no gestionades com Internet, que implementa l'estàndard Dynamic Adaptive Streaming sobre HTTP (DASH) per als serveis Over The Top (OTT). Per assolir aquest objectiu, s'implementen bancs de proves i es realitzen experiments per concloure com el trànsit de vídeo podria beneficiar l'eficiència energètica i aconseguir una reducció a gran escala del consum d'energia si s'implementa l'estàndard EEE a les targetes Ethernet dels dispositius implicats en la distribución de vídeo.

El tema integrador que s'aborda en aquesta tesi és la distribució de vídeo digital, per tant, aquest treball també presenta un estudi sobre la codificació i adaptació de la transmissió de vídeo a les xarxes IP, presentant aspectes relacionats amb l'avaluació de la qualitat del vídeo i l'avaluació de la qualitat de la transmissió de la xarxa. Per això, es proposen i s'implementen metodologies i bancs de proves que impliquen la programació i execució de simuladors, scripts, algoritmes, programari i, en general, frameworks per a la realització d'estudis de transmissions de vídeo en xarxes de telecomunicacions, abastant l'ampli espectre de les Comunicacions Multimèdia. Aquest estudi podria ser útil per emprendre altres estudis necessaris en la ràpida evolució tant de la demanda de serveis multimèdia com de les noves propostes de tècniques de distribució de continguts.

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A mis padres, hermanos, familiares y amigos que me motivaron siempre a la dedicación a este trabajo, a Ingrid Juliana que con su amor, compañía y soporte he podido culminar esta meta.

Tito Raúl Vargas Hernández

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Learn from yesterday, live for today, hope for tomorrow.

The important thing is not to stop questioning.

Albert Einstein

To my parents, sister, brother, wife, and daughter.

A mis abuelos y abuelas

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Acronyms

ACK	Acknowledgement
BER	Bit Error Rate
CBR	Constant Bit Rate
CIF	Common Intermediate Format
DASH	Dynamic Adaptive Streaming over HTTP
EEE	Energy Efficient Ethernet
FPS	Frames Per Second
GoP	Group of Pictures
HTTP	Hypertext Transfer Protocol
HVS	Human Visual System
IEEE	International Electrical and Electronic Engineers
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPTV	Internet Protocol Television
ITU	International Telecommunication Union
MOS	Mean Opinion Score
MTU	Maximum Transfer Unit
NS2	Network Simulator 2
OCS	Optical Circuit Switching
OBS	Optical Burst Switching
OPS	Optical Packet Switching
OTT	Over The Top
PSNR	Peak Noise to Signal Ratio
QCIF	Quarter Common Intermediate Format
QoE	Quality of Experience
QoS	Quality of Service
RTP	Real-time Transport Protocol
SNR	Signal to Noise Ratio
SVC	Scalable Video Coding
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
VOD	Video on Demand

Chapter 1
Introduction

This chapter introduces the background and scope of this thesis, explains the problem and research questions that motivated the studies carried out. Furthermore, research objectives, methodology, activities and results are presented. The structure of this dissertation is also described, briefly explaining the contents of the document chapters and relating contributions of this work.

1.1. Background

The research and contributions subject of this work are aligned to video distribution over broadband IP networks, specifically in aspects such as multimedia content adaptation for streaming, simulation and testbeds for video streaming, video traffic characterization and patterns identification, video transmission quality assessments, video aware transmission protocols analysis, and energy efficiency aspects regard video applications traffic patterns at ethernet access networks.

The beginning of the work involves research on the broadband transport networks field, particularly optical communication networks. The nature of such networks is that they are implemented in optical technologies avoiding electrical components and signals. Therefore, Optical Burst Switching (OBS) networks architecture and protocols are studied for being an optical switching technology proposed by the research community as a network paradigm for end-to-end all optical core networks and also for being a well-defined network proposal ideal for academic research and experimentation in managed core networks area. Since video traffic is mainly originated and consumed in access networks crossing managed or unmanaged core networks and considering the initial results and the experience gained, this work ends with video traffic pattern and energy efficiency analysis of the IEEE Energy Efficient Ethernet (EEE) standard at the access network.

During the development of this project, an OBS network study and simulation has been carried out. A set of algorithms and protocols evaluation has been made to determine how the OBS technique offers solutions to the emerging problems caused by increased traffic, specially the multimedia and real time traffic. The study carried out has a great emphasis in the evaluation of the quality of multimedia flows carried over OBS networks. A good quality video streaming services convey Standard Definition (SD), High Definition (HD), Full HD, 4K and 8K videos demanding a great amount of network recourses like bandwidth to offer Quality of Experience (QoE) to the end users. Therefore, high-quality video transmissions require Quality of Services (QoS) network guaranties that the current and future optical networks must satisfy.

Accordingly, an evaluation and simulation framework has been proposed and implemented to assess and contribute to the studies about quality of video traffic transmission over OBS networks. In addition, video traffic generated at the access network segment is analysed. Therefore, it makes possible to assess and propose OBS core network building parts, like algorithms, that could improve QoS guaranties for multimedia traffic flows from different services and application that use this IP transport networks. Particularly, in this work, the methodology and framework defined had made possible to carry out studies and proposals on burst assembly algorithms and scheduling algorithms in OBS networks.

Since this project involves experiments of codification, transmission, tracing, trace driving simulation and analysis of video services, and considering the preliminary results and research collaborations that this work has motivated, an opportunity to apply this experience has been identified. Therefore, simulation environment and testbeds has been proposed and deployed to experiment and evaluate the energy efficiency in the ethernet access network segment where IP Television (IPTV) and Dynamic Adaptive Streaming over HTTP (DASH) applications data are being transported. The later, mainly because IPTV and DASH are technologies wildly implemented, the demand of online streaming platforms as Over The Top (OTT) services based on DASH are increasing, and there is a research interest in proposing mechanism that help energy savings to the wide spread deployments.

1.2. Problem definition

The increasing number of users connected to the Internet through broadband access networks and the increasing traffic from multimedia and real time applications, are causing an increment in the total traffic carried on metropolitan and backbone networks. As massive deployments of access network technologies based on Gigabit Ethernet, Wireless and Optical broadband are implemented, the increased traffic from access networks could adversely affect the cost and quality of the services to the end user. The later could have higher impact if the transport networks architectures based on electronic switching are used. At the transport network, optical burst and packet switching techniques are network paradigms and techniques studied on the road to increase the capacity of metropolitan and backbone segments, reducing operational costs with respect to electronic switching networks. Some other problems arise because of the increment of video traffic demand, for instance the energy consumption at access network segments where servers and end users are, something that would be investigated to applied techniques that impact in energy savings. The aforementioned aspects suggest different issues to be study, with respect to this research project here are some research questions that have guide the work carried out:

- How can video traffic be characterized to evaluate its quality when transmitted over Optical Burst Switched Networks?

- Are there alternatives to propose algorithms that favour the quality of video transmissions over OBS networks?

- What is the impact of RTP/UDP/IP video traffic of the IPTV application when protocols are enabled for energy efficiency in ethernet access networks?

- What is the impact of HTTP/TCP/IP video traffic of the OTT Internet video streaming application when protocols are enabled for energy efficiency in ethernet access networks?

1.3. Research objectives

Considering the context that describes the previous section, below are listed the main and specific objectives of this thesis:

To contribute to the video streaming quality in optical networks and to the energy efficiency in ethernet networks by conducting experiments and analysis about video coding, streaming alternatives and protocols, traffic patterns, quality assessment and energy efficient standards.

To achieve the main objective, specific goals are defined:

O1. To investigate about the end-to-end digital video distribution over broadband networks, integrating research on optical networks, particularly OBS networks, digital video compression techniques, video distribution over IP networks, video QoS and QoE assessments, IPTV and OTT video distribution technics.

O2. To provide tools and methodology for simulation and evaluation of digital video distribution over broadband networks, particular case of OBS networks, allowing the evaluation of quantitative (QoS) and qualitative (QoE) parameters of video distribution.

O3. To contribute to the studies and investigation about QoE and QoS in video streaming services over OBS networks particularly in burst assembly and scheduling algorithms.

O4. To provide tools and methodology to contribute to the studies and investigation about video traffic patterns of IPTV and DASH streaming applications and its corresponding evaluation of the Energy Efficiency in the Ethernet access network to impact on energy savings.

1.4. Research methodology

This project follows a research methodology described in Figure 1.1 showing the main outcomes and activities organized in three main phases that are aligned with the project objectives. Therefore, the development of this project has been carried out in three main phases:

- Phase 1: Study and investigation.
- Phase 2: Experimentation and analysis.
- Phase 3: Contribution and conclusion.

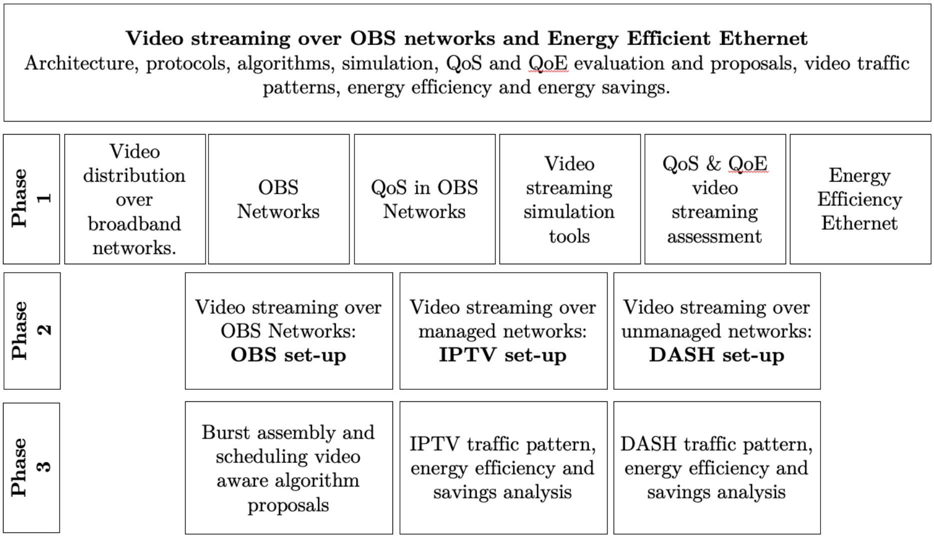


Figure 1.1 Research methodology

The phases and activities are based on theoretical and practical research methods. On one hand, bibliographic review and study about broadband optical networks, video codification and transmission technics, simulation tools, video evaluation tools, video traffic pattern characterization and energy efficiency in networks analysis. On the other hand, contribution in the research area by evaluation, algorithms, methodologies, and testbeds proposals to run experiments, analysis, conclusions to contribute to the research field of this project. The research results have been offered to the community through seminars, conference and journal publications and knowledge transfers activities.

1.5. Activities and contributions

In this section it is presented the three research project phases describing the activities carried out on each phase to highligh the corresponding results and contributions of work that are later presented in detail in the following chapters of this document.

Phase 1: Study and investigation

A1.1. Video distribution over broadband networks

In this activity a revision of digital video distribution theory and technics has been carried out about these topics:

- Video compression and codification standards and tools.
- Video streaming over managed networks: IPTV service, RTP/UDP video streaming.
- Video streaming over unmanaged networks: OTT Internet video, HTTP/TCP video streaming
- Dynamic Adaptive Streaming over HTTP, DASH

A1.2. QoS and QoE video streaming assessment

In this activity a study about performance evaluation metrics for video quality and network quality has been carried out reviewing concepts such as:

- QoE evaluation in video streaming scenarios.
- QoS network performance evaluation regard video streaming scenarios.

A1.3. OBS network study

A complete study about OBS networks has been done to know in deep detail this network paradigm in aspects like:

- OBS network architecture.
- OBS protocols and algorithms: Burst assembly, resource reservation, channel scheduling, signalling procedures, content resolution schemes, routing and wavelength assignment.
- QoS strategies in OBS networks.

A1.4. Video streaming evaluation tools and OBS Network simulation tools

Video streaming quality evaluation and simulation tools has been also reviewed and tested for OBS Networks following the next index:

- Simulation tools for OBS networks.
- Video traffic OBS network simulator.
- Simulation tools testing for OBS networks.
- Video traffic over OBS networks analysis.

A1.5. Video streaming over managed and unmanaged networks

Regard video streaming over managed and unmanaged networks in this work has been investigated the following strategies and standards:

- IPTV, RTP/UDP streaming for service provided managed networks.
- DASH, HTTP/TCP streaming for unmanaged networks like Internet.

A1.6. Energy Efficient Ethernet study

Energy Efficient Ethernet standard defined by the IEEE have been also reviewed following the next aspects:

- IEEE 802.3az Energy Efficient Ethernet (EEE) standard definition.
- EEE energy efficiency calculation and analysis.
- EEE energy efficiency regard video burst traffic calculation and analysis.

Phase 2 and 3: Experimental set-up, analysis, and contribution

Next the activates that have been carried out in phases 2 and 3 are grouped to resume the experimental testbeds, the analysis and the corresponding contribution regard the three main testbeds for experimentation namely: video streaming over OBS networks assessment tool set-up, video streaming over managed networks set-up and the video streaming over unmanaged networks set-up.

A2.1. Video streaming over OBS networks assessment tool set-up.

A video streaming over OBS Networks assessment framework has been proposed and applied to research and contribute with the following:

- A study about the effect and optimization of burst assembly algorithms for video traffic transmissions over OBS Networks.
- A proposal about an Adaptive Burst Assembly Algorithm (ABAA) to improve video QoE in OBS networks.
- A scheduling scheme algorithm proposal to provide QoE to video traffic in OBS networks.
- Three knowledge transfer activities of the video streaming evaluation set-up and methodology to research groups by visiting the Navarra Public University (invited), Patras University (Mobility Action), and Essex University (Mobility Action).
- Four knowledge transfer activities of the video streaming evaluation set-up, methodology and results at the BONE European network of excellence summer school and workshops and at the RT-Multilayer and Fierro national thematic networks workshops.

A2.2. Video streaming over managed networks: IPTV set-up to EEE energy efficiency analysis.

To contribute to the analysis of the energy efficiency of the EEE standard a testbed, experiments and analysis have been carried out leading to the following results:

- An IPTV service application testbed.
- An IPTV video traffic profile characterization at application and ethernet level.
- An evaluation and conclusions about of energy efficient and energy savings in video streaming servers implementing the EEE standard.

A2.3. Video streaming over unmanaged networks: DASH set-up to EEE energy efficiency analysis.

In this work Internet video streaming traffic have been also characterized, thus DASH standard has been selected for being widely used by the OTT streaming platforms. This part of the study follows the same methodology of the of the IPTV traffic pattern and EEE energy efficiency analysis leading to next results:

- A DASH testbed.
- A DASH traffic pattern characterization at application and ethernet level.
- An analysis and conclusions about energy efficiency and energy savings on servers and clients of DASH streams implementing the EEE standard.

1.6. Structure of the document and contributions

This document is organized into four chapters that present the work carried out in this doctoral research project

- **Chapter 1 – Introduction.**

  Chapter 1 introduces the background and scope of this thesis, as well as the problem definition and research questions that motivated the study. Furthermore, the research objectives, methodological aspects, and the structure of this thesis are described, as well as a brief description of the contributions of this work. The contributions intend to provide building mechanisms and knowledge for improving video streaming quality over IP networks and to make contributions regard energy efficiency in communications networks.

- **Chapter 2 – Video streaming over broadband networks.**

  The chapter 2 describes video streaming over IP networks explaining video compression and codification, streaming services over managed and unmanaged networks. Concepts QoE and QoS in the context of video streaming are introduced and the video evaluation process that supports the development of this thesis is described. Chapter 2 is aligned with objectives **O1** and **O2**.

- **Chapter 3 – Video streaming services over OBS Networks**

  This chapter presents research results about video distribution over OBS Networks. It starts with an OBS networks overview followed by the presentation of the QoE and QoS video traffic evaluation framework proposed. A study over the burst assembly algorithms in OBS network is presented and its consideration about video transport. There are also presented proposals regard OBS network functionalities, such as adaptive burst assembly and scheduling algorithms to provide QoE to video transport. Chapter 3 presents results related to objectives **O2** and **O3**.

- **Chapter 4 – Video streaming traffic patterns for Energy Efficiency Ethernet analysis.**

  In this chapter the research about video traffic profiles from video streaming services in the study on energy efficiency in the EEE standard in access networks is described. IPTV streaming and DASH streaming applications traffic patterns are characterized and energy efficiency analysis and ethernet energy savings is calculated regard video traffic. Chapter 4 presents results related to **O4.**

Furthermore, this thesis has led to the publication of scientific journal papers, conference presentations and book chapters, mobility actions and collaborations. A detailed list of publications derived from this work is presented in Appendix A.

Chapter 2
Video streaming over broadband networks

In this chapter the main video distribution technics over IP networks are introduced. Video distribution to its destination implies a set of process such as video generation, digitalization, codification, compression, packetization and transmission. At reception side, the video packets are put together to make the process of content presentation: un-compression, decodification and visualization. This video distribution process is explained in this chapter.

Video distribution over IP networks is a challenging mechanism, it demands a lot of networks resources, especially bandwidth to deliver in time and in good terms of video quality. Moreover, evaluating the video transmission over IP networks implies measurements about the service level offered by the network and the level of satisfaction of the end user. Hence, in this chapter is explained the quality metrics for the evaluation of video transmission over IP networks. This metrics are explained under the Quality of Services (QoS) and Quality of Experience (QoS) evaluation concepts.

2.1. Video streaming over broadband IP networks

Transmitting content between different nodes connected through a network could be done in different ways. Usually, the type of content and the underlying network determine the delivery technique. Furthermore, content preparation for transmission and delivery strategies must be analysed to design the appropriate scheme for video transmission over the network. In next subsections, video compression and codification technics are described, as well as an explanation about video distribution technics over IP networks [1].

2.1.1. Compression and codification of video content.

Moving Picture Expert Group (MPEG) has developed compression systems for video distribution, such as MPEG-1, MPEG-2 and MPEG-4. These compression systems enable a number of advanced video services. The MPEG compression standards share the basic compression concepts that are explained below. The difference between each standard lies on the application, quality of the compressed video and the complexity of the encoders and decoders. For instance, MPEG-1 has been used for video storage and reproduction in CD and DVD systems player, it is now considered as a legacy technology. MPEG-2 and MPEG-4 systems are used widely to video broadcast over Digital TV (DTV) networks and video distribution over IP networks respectively. MPEG-4 compression systems have been used in the develop of this research [2].

The purpose of applying compression technics to a video source is to reduce the spatial and temporal redundancy in resulting video sequence, as well as to reduce the amount of information to be storage and transmitted while the image quality is maintained.

The compression algorithms use inter-video frame prediction to reduce the data between a video frame series. It is about differential codification where a video frame is compared with a reference video frame. A video frame is a still image that combine with multiple images form a video sequence. Therefore, the amount of information to transmit is reduced when is used a differential codification between a group of video frames. In inter-video frame prediction, each compressed frame is classified in a type of video frame. *I*, *P*, or *B* are the possible video frame types. Figure 2.1 shows the video frame distribution of a video frames group of a video. It could be seen how the compressed frames are group from different types of video frames depending one on another to conform a compressed image sequence. In the Figure 2.1 it is represented the frame viewing order or played and the corresponding transmission order over the network. Next a brief explanation of each type of video frame is described.

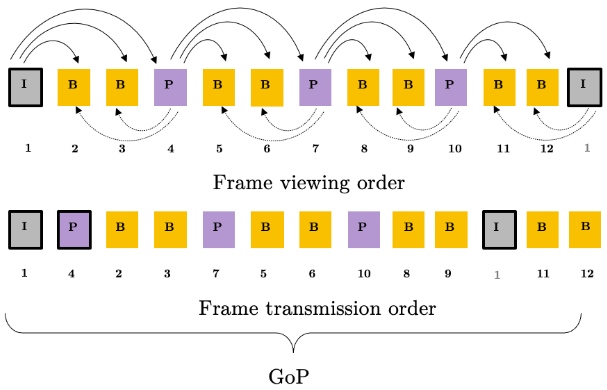


Figure 2.1 Compressed video frame sequence

- The *I* video frame type is a reference frame that could be compressed independently. It is the first video frame in the sequence. The *I* frames are the initial point of new visualizations and synchronization points.
- A *P* video frame, or *predicted* frame, is referred to a previous *I* or other previous *P* frames. This *P* frames requires less bits but are sensible to transmission errors due to its frame dependency.

- A *B* video frame or *bi-predicted* frame makes reference to a previous and subsequent frame, the reference video frames could be *I* or *P*.

After the above brief explanation of the basic video compression concept, compression standards that have been studied in this project are presented next.

2.1.2. MPEG-4 part 2, Advanced Simple Profile

MPEG-4 is a video compression ISO/IEC standard developed by MPEG (Moving Picture Experts Group) [3]. It defines a set of tools to create, represent and distribute individual audio-visual objects. An MPEG video consists of many video sequences, each of which has many groups of pictures in it [4]. Typically, a Group of Pictures (GoP) has a single Intra-frame (*I* frame) and many Predictive frames and Bidirectional frames (*P* frames and *B* frames). Although there is no limit on the size of GoP, values around 15 is common. An MPEG-4 encoder generates these three types of frames. The *I* frames contain information from encoding a still image. *P* frames are encoded from the previous *I* frames or *P* frames. *B* frames are encoded bi-directionally from the preceding and the following *I* and *P* frames. *B* frames achieve the highest compression, which require the lowest transmission bandwidth, but they also take the longest time to encode [5]. Conversely, *I* frames have the lowest compression of the three frame types, but they can be encoded and decoded faster than the other frame types. *I* frames contain by far the most information. Furthermore, losing an *I* frame would cause distortion of all frames in a GoP. A *P* frame loss would only influence the following frames in the GoP after the frame loss and the loss of a *B* frame would not influence any other frame.

MPEG-4 part 2, Advanced Simple Profile (ASP), allows a 50 percent of bandwidth reduction compared to an equivalent video quality of MPEG-2 ASP standard, making the standard for video transport over IP networks before being outstrip by its successors. This standard allowed the video distribution of HD video over Internet and IPTV networks. In MPEG-4 different elements of the video like other audios, on-screen messages, logos and identifiers, could be generated and transmitted as a separate data units of the MPEG-4 stream. The decoder has to assemble in a single video unit to be shown in the user display. MPEG-4 allowed the use of *B* frames based on other *B* frames reducing the overall bandwidth requirements.

2.1.3. MPEG-4 part 10, Advanced Video Coding/H.264

MPEG4 Part 10, also known as MPEG-4 AVC (Advanced Video Coding) or H.264, is a set of compression systems of natural images. It offers efficient compression, same quality level as its predecessor MPEG-4 part 2 but lower encoded bit rate. Here encoders and decoders are more complex. The standard defines many techniques and technologies considering that it is no feasible to implement every feature for every application. Therefore, a set of profiles have been defined for H.264 to be applied in a range of applications. Some profiles are listed below:

- Baseline profile: Represent the lowest processing load, *B* frames are no used. Suitable for videoconference and mobile TV applications.

- Main profile: It is designed for main consumer and broadcast. It adds *B* frames. It requires more encoded and decode processing power and memory.
- Extended profile: Targeted for streaming applications. Supports recovery from data losses.
- High profile: Supports high-quality video storage and broadcast for professional applications.

2.1.4. MPEG-4 part 10, Scalable Video Coding/H.264

Scalable Video Coding (SVC) is an extension of MPEG-4/H.264 AVC. SVC considers a different GoP structure built by hierarchical *B* video frames. Such hierarchical feature of H.264/SVC refers to the fact that *B* video frames can be used to predict other *B* frames, in contrast with the classical H.264/AVC *B* video frame prediction process where a *B* video frame depends on the previous and next *I*/*P* video frames. Essentially, the layer-scalable H.264/SVC codec offers temporal, spatial and quality scalability to perform the decoding process using a single base layer or the base layer plus several enhancement layers (see [6], [7] for further details). This allows to remove some *B* video frames without losing the ability to decode the video. The compressed video frame sequence in H.254 is depicted in Figure 2.2.

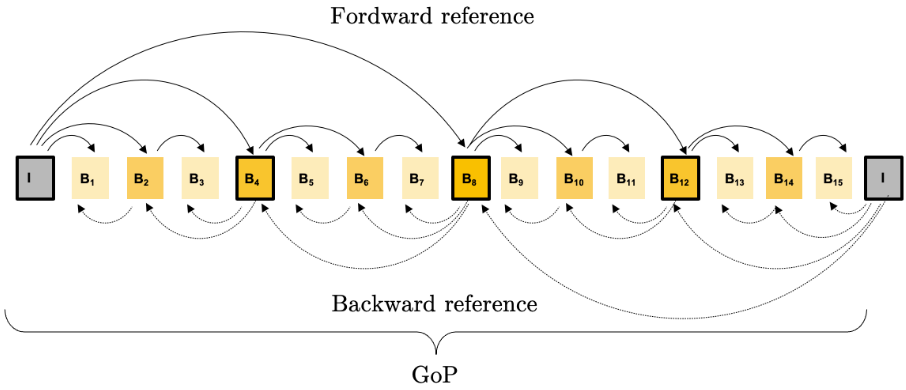


Figure 2.2 Compressed video frame sequence in H.264/SVC

2.1.5. MPEG-H part 2, High Efficiency Video Coding/H.265

H.265 or MPEG-H Part 2 is how it is usually known the video compression standard High Efficiency Video Coding (HEVC) designed in the MPEG-H project. Standardized and approved by ITU-T in 2013. The standard is catalogued as ISO/IEC MPEG-H Part 2, ITU-T H.265. H.265 follow on to H.264 (AVC), delivers significant compression improvement over H.264 and is adopted in the industry. H.265 present up to 40% better compression than H.264 in general use case [9].

2.1.6. MPEG-I part 3, Versatile Video Coding/H.266

H.266 also known as MPEG-I part 3, Versatile Video Coding (VVC) and Future Video Coding (FVC) is a video compression standard by the Joint Video Experts Team (JVET), a joint video expert team of the VCEG working group of ITU-T Study Group 16 and the MPEG working group of ISO/IEC JTC 1. Standard finalized on 6 July 2020 to make 4K broadcast and streaming. H.266/VVC offers 50% better compression than H.265/HEVC [10][11].

2.1.7. Streaming over managed networks, IPTV, RTP/UDP.

IPTV is a video distribution network to deliver broadcast TV channels to consumers over an IP network, a private IP network, like those networks being deployed by telephone companies. Classical broadcast TV channel distribution networks are terrestrial broadcast, CATV and satellite services, such networks do not use IP to deliver the video channels. In IPTV networks customers at home use a set-top-box to take the incoming IPTV stream from the IP network to extract TV channels information and convert it into a video signal to display in a TV set. Features of IPTV are:

- Continuous streams of professionally produced content (such as a TV broadcast network feed).
- Hundreds of 24/7 channels.
- Main video content is in form of TV channels and some video on demand content.
- Uniform content format (all channels typically share one compression method and use roughly the same bit rate).
- Delivered over a private access network, such as a telco digital subscriber line (DSL), Fibre and coaxial access networks (CATV, FTTX) and Ethernet access networks (IP Campus Access Networks).
- Viewed on consumer televisions by way of a set-top box.

Something particular about IPTV is that even though IP is used, the public Internet is not the transport network. IPTV is an IP managed network. The network operator has complete control about it and at the same time usually is the video service provider, therefore the network infrastructure is mainly for TV channels distribution. In this kind of networks, video streaming services uses three protocols of the transport layer, User Datagram Protocol (UDP), Real time Transport Protocol (RTP) and Real-Time Control Protocol (RTCP). UDP is a lower-layer transport protocol while RTP and RTCP are upper-layer transport protocols. UDP is employed because it provides timely delivery of packets. RTP/RTCP [8] provide sequence numbering and time stamping of multimedia packets as well as a feedback channel to obtain reception statistics concerning the received video quality. This pair of protocols are used as the main standard for multimedia streaming over private IP managed networks. The transport protocol used to encapsulate RTP and RTCP packets is UDP. UDP is a simple transport protocol without connection establishment, flow control or packet retransmission; therefore, it suits the requirements of real time applications better than TCP. However, UDP does not guarantee packet delivery.

2.1.8. Streaming over unmanaged networks, Internet, HTTP/TCP

The concept of Internet video is used to talk about video content distribution strategies that used the public Internet to deliver the video to the end user, also known as Over The Top (OTT) streaming services. In this case, the video provider has no control over the network infrastructure, so is it an unmanaged network, the video provider has no control over the video traffic priorities or classification. The usual Internet video set up is a *web server* hosting a website reachable for the video consumer with a standard HTTP request through a web browser. Then the traffic is sent using TCP as a transport protocol from server to the user that is accessing via a software player in a computer, tablet, smartphone, embedded and portable video players or TVs. The website has an index of the video content and all the pieces that conform it, in that way the user selects the content to play. The main characteristics of Internet video are:

- Discrete video content elements, from clips lasting a handful of seconds to full-length movies.
- Millions of content offerings, like YouTube, Netflix, Hulu, HBO MaX.
- Widely varying video content formats, including different types of compression, rights management technologies, image quality (bitrate and resolution).
- Delivered over the public Internet.
- Viewed via software player in computer, tablets, TVs, smartphones.
- Different video distribution technics: video download, progressive download, and streaming.

To overcome the unmanaged nature of the public Internet and its best effort nature to deliver packets, there are a standard and technologies that adapt the video traffic to the variable network conditions, the bandwidth fluctuations and offer a better QoE to the end usar. The standard is called Dynamic Adaptive Streaming over HTTP (DASH) and is explained in the next subsection.

2.1.9. Dynamic Adaptive Streaming over HTTP

DASH is a standard for video transmission through IP networks using HTTP as the application protocol. This transmission technique implies a set of video content of the same video source, each one codified at different quality. Thus, the quality selected for transmission depends on the traffic conditions like congestion and bandwidth. Network conditions are variable during the video transmission, so the quality content to be transmitted is selected according to the network resources available. The video transmission is then adaptable to the network conditions over transmission time. As a result, the video is played without interruption at destination point and network congestion is avoiding [12].

DASH is a widely deployed content distribution standard for unmanaged networks like Internet. This technic implies dynamic changes of quality of the video or audio being transmitted depending on the available bandwidth between the client and server. Thus, the client player requests a specific quality (bitrate, resolution, etc.) of the same content depending on measurements of the available bandwidth and the playback buffer. A specific number of available content qualities is known beforehand by the player prior to starting the streaming process. The dynamic adaptation to the network conditions avoids playing interruptions and video frame losses.

The DASH standard is defined as MPEG-DASH-ISO/IEC 23009 standard [89], it provides the features and framework to offer OTT services through Internet, services like the widely known Netflix, HBO MaX, and so on and, at the same time, enables them to coexist together avoiding congestion using to the fullest the available network resources to the fullest at each specific time. Nowadays, DASH have been implemented by different players and video and audio encoding software such as *Shaka player* [90], *BITMOVIN*, *JWPLAYER* and the *FFmpeg* player and encoding tool.

The DASH application architecture mainly consists of a web server, which is a software entity in charge of serving the content. The video and audio content to be streamed is hosted in this web server and using different files that represent many qualities of the same content using, for instance, MP4 containers. Each quality container file is divided in segment files or, alternatively, one single file with a hint track of segments with a specific time duration. Available qualities, segments and codec parameters are indexed in an XML file called Media Presentation Descriptor (MPD). Additionally, on the other side of this DASH architecture there is the player client, that could be implemented as an independent player or a web application built to run on a web browser. Figure 2.3 presents this DASH architecture.

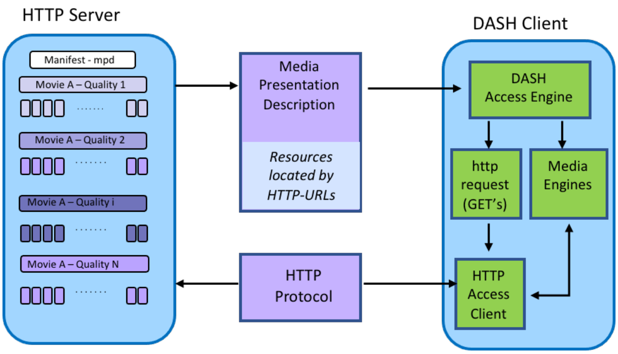


Figure 2.3 DASH application architecture and service architecture

When a DASH streaming session starts, the player first makes an HTTP GET request to download the MPD. After analyzing the contents of the MPD, the player can now perform the following HTTP GET requests to fetch the video and audio container files, specifically, the segments of the selected quality according to the available bandwidth at the exact reproduction time. The player will continue getting segments at the selected quality until the content playback reaches the end. In Figure 2.4, the protocol message exchange in a DASH session is presented. In particular, the HTTP GET request message to download the MPD XML file can be seen, as well as the audio segments in one quality (Q1) and the video segments in qualities Q1 and Q2 and, finally, the corresponding HTTP Response messages.

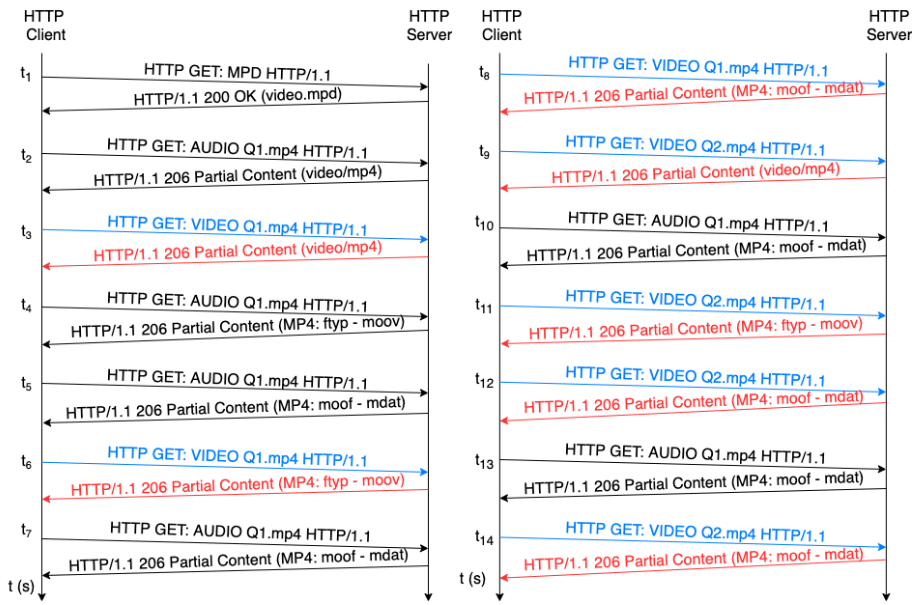


Figure 2.4 DASH session protocol message exchange

2.2. QoS and QoE in video streaming scenarios

To evaluate and measuring quality of communications services and the underlying networks from network performance perspective and the end-user point of view, the concepts of Quality of Service (QoS) and Quality of Experience (QoE) are defined respectively.

2.2.1. QoS video traffic evaluation parameters

On one hand, QoS is a measurement of the network performance when carrying on the service applications traffic, as well as the network techniques and configuration to offer a quality network resources. The QoS is important due to the fact that each service and application have different requirements of network resources, like bandwidth, and different guaranties of delivery of data with bounded time, tolerant or not to packets loses or variable bitrate. Therefore, offering QoS implies network configuration and measurements supporting by algorithms and protocols.

Techniques to offer QoS for video traffic are defined by priority polices for video packets and resource reservations, this kind of technics are difficult to implement in unmanaged networks where the video content provider has no control about the network configuration and resource reservations.

Performance metric to assess the QoS offered by the network for video traffic usually are:

- Delay of video packets of video frames.

- Throughput of video packets of video frames.
- Loss rate of video packets of video frames
- Jitter of video packets of video frames.

This performance metrics joint with quality perceived metrics leads a complete analysis on the video streaming service.

2.2.2. QoE video evaluation parameters

On the other hand, QoE is a measurement of the services from the final user perspective, the grade of satisfaction in terms of quality that the user has about the service. Also, it is a mechanism to verify if the network is satisfying the needs of the service and application. [13]

QoE measurements could be subjective and objective. The subjective measurements imply that people/customers evaluate the services. The objective assessments attempt to generate models that compute the QoE avoiding the help of end-users but using programs and algorithms to automate the process.

Therefore, is important to evaluate and propose techniques that improve QoE/QoS for video transport. Next it is presented y more detail the QoE video evaluation parameters

A video quality measurement must be based on the perceived quality of the actual video being received by the users of the digital video system. Such perception-based evaluation is appropriate, as the subjective impression of the user is what only counts. This intuitive impression of a human user watching a video is achieve by subjective quality metrics. These subjective metrics provide most information, but their determination is costly as real humans have to watch videos, making them highly time consuming and requiring high manpower and special equipment.

The expensive and complex subjective tests are often not affordable. Also, many tasks in industry and research require automated methods to evaluate video quality, for instance after codifications and transmissions. That is why objective metrics have been developed to emulate the quality impression of the human visual system.

- **Mean Opinion Score (MOS) and Peak Signal-to-Noise Ratio (PSNR):**

Mean Opinion Score (MOS) is the human impression of the video quality, which is given on a scale from 5 to 1 [14], being a subjective quality metric. Scale 5 corresponds to the best quality perceived and scale 1 is the worst quality perceived. In order to obtain the MOS in video evaluation process, each image that conform the video has to be evaluated computationally. Therefore, an objective quality metric that maps to the MOS is used, the Peak-Signal to Noise Ratio (PSNR), that has shown to result in a good correlation with the subjective quality perception [14]. The PSNR results are mapped to a corresponding MOS value using an equivalence table, as shown in Table 2.1.

The PSNR compares the maximum possible signal energy to the noise energy, which is mathematically equivalent to the root mean squared error [15]. The Equation 2.1 shows the definition of the PSNR.

Table 2.1 PSNR to MOS equivalence

PSNR [dB]	MOS
> 37	5 (Excellent)
31 – 37	4 (Good)
25 – 31	3 (Fair)
20 – 25	2 (Poor)
< 20	1 (Bad)

$$PSNR = 10log_{10}\ [255/MSE]$$

Equation 2.1

$$MSE = \frac{1}{MN}\sum_{m=0}^{M-1}\sum_{n=0}^{N-1}[I_{org}(m,n) - I_{dec}(m,n)]^2$$

Where I_{org} is the original image and I_{dec} is the decoded image; M, N represent the size of the image; and MSE is the Mean Square Error.

While the PSNR does not directly correspond to the MOS, it is used the heuristic mapping of PSNR to MOS (subjective quality) as explained above. To evaluate the impact of the network on the video quality, it is necessary to compare the received video (possibly distorted) with the actually sent video. The sent video is already distorted by the encoding process and this distortion cannot be avoided when striving for acceptable bit rates in video streams, still the video could be distorted more because losses of frame information when is streamed through the network.

Hence, there is a computational approximation of the subjective human impression of every single frame. Based on this frame by frame MOS calculation, also it is defined a metric which reflects the user impression of the entire received video. A simple approach to extend a frame-wise metric like the PSNR to a metric for an entire video would be to calculate the average of all PSNR values of all frames.

- **Structural Similarity Index Measure (SSIM)**

The Structural Similarity Index Measure (SSIM) it is an objective metric of greater complexity than the PSNR whose objective is to better adapt to the subjective quality of the image. For this, a different weighting of the distortions produced in luminance, contrast and image structure is carried out, giving greater importance to those that most affect subjective quality such as changes in structure and less to those that have to do with luminance [11][16]. With this, SSIM is a metric that does not compare the pixels of the image, if not the elements perceived by the human being. The expressions that allow obtaining the SSIM are presented below.

$$SSIM(x,y) = [\mathrm{l(x,y)}]^{\propto} \cdot [\mathrm{c(x,y)}]^{\beta} \cdot [\mathrm{s(x,y)}]^{\gamma}$$

$$l(x,y) = \frac{2\mu_x\mu_y + \ C_1}{\mu_x^2 + \mu_y^2 + C_2'}$$

$$c(x,y) = \frac{2\sigma_x\sigma_y + \ C_2}{\sigma_x^2 + \sigma_y^2 + C_2'}$$

$$s(x,y) = \frac{\sigma_{xy} + \ C_3}{\sigma_{xy} + C_3'}$$

$$C_3 = \frac{C_2}{2}$$

If $\alpha = \beta = \gamma = 1$ then

$$SSIM(x,y) = \frac{(2\mu_x\mu_y + \ C_1)(2\sigma_x + \ C_2)}{(\mu_x^2 + \mu_y^2 + C_1)(\sigma_x^2 + \sigma_y^2 + C_2)}$$

$$SSIM_{ij} = W_Y \cdot SSIM_{IJ}^{Y} + W_{Cb} \cdot SSIM_{IJ}^{Cb} + W_{Cr} \cdot SSIM_{IJ}^{Cr}$$

Where:

- α, β, γ are the importance coefficients.
- μ_x is the mean value of the luminance sample of the image x.
- μ_y is the mean value of the luminance sample of the image y.
- σ_x is the standard deviation of the luminance sample of image x.
- σ_y is the standard deviation of the luminance sample of image y.
- σ_{xy} is the covariance of the luminance of the two images x and y.
- C_1,C_2 are the stabilization coefficients
- W_Y is the weight of luminance
- W_{Cb} is the weight of the chrominance of blue.
- W_{Cr} is the weight of the chrominance of red.

SSIM values can vary between 0 (least similarity) and 1 (maximum similarity and same image).

- **Video Multi-method Assessment Fusion (VMAF)**

Video Multi-method Assessment Fusion (VMAF) is a video quality metric developed by Netflix whose objective is to predict the subjective quality of a video by combining elementary quality measurement metrics. The Machine-Learning Support Vector Machine (SVM) algorithm is used to merge all these metrics [11] [17]. The algorithm assigns weights to each elementary metric so that the final metric obtained maintains the strengths of the individual metrics. The machine-learning model is trained with MOS (Mean opinion scores) obtained with the subjective experiment NFLX Video Dataset.

In the NFLX Video Dataset experiment, Netflix generated a set of reference videos, encoded and distorted in order to cover animation scenarios, exterior and interior, with objects, with people in motion, etc. Through a subjective evaluation with real users, the mean MOS for each video was determined.

The current version of VMAF 2.3.1 uses the following quality assessment metrics:

- **Visual Information Fidelity (VIF):** It is an image quality metric based on the quality perceived by the human being that in general offers better correlation with the subjective evaluation than the PSNR.
- **Detail Loss Metric (DLM):** It is a metric that consists of the measurement of the loss of detail that affects the visualization of the video and the redundancy that can capture the user's attention.
- **Movement:** Measurement of the temporal characteristics of the video in which the temporal difference between adjacent frames is obtained. To do this, the difference in absolute value of the luminance component is calculated.

VMAF ranges from 0 to 100, with 0 being the minimum value (worst quality) and 100 being the maximum value (best quality).

Table 2.2 presents the MOS, PSNR, SSIM and VMAF interpreting scores and equivalences.

Table 2.2 MOS, PSNR, SSIM and VMAF interpreting scores.

	MOS	PSNR [dB]	SSIM	VMAF
Excellent	5	> 37	.99+	80 – 100
Good	4	31 – 37	.95 - .99	60 – 80
Fair	3	25 – 31	.88 - .98	40 – 60
Poor	2	20 – 25	.50 - .88	20 – 40
Bad	1	< 20	< .5	< 20

2.3. Chapter Summary

In this chapter has been presented the theory about video streaming over broadband IP networks regard this project, presenting the compression and codification standards and the streaming techniques over managed and unmanaged networks. The DASH streaming standard is also explained. It has been also presented concepts about the QoS and QoE for video streaming. The concepts presented in this chapter are fundamentals on which this study has been based regard multimedia networking and it is referred from the next chapters of this document.

Chapter 3

Video Streaming Services over OBS Networks

Optical fibre technologies are being used in transport networks, mainly at metropolitan and long distance network segment. In addition of that, optical communications are highly being used in access networks. Brand new optical networks deployments tend to involve all optical communication technologies avoiding optical/electrical media conversions. All optical networks, where the transactions are end-to-end at optical domain, enable better fibre optics bandwidth utilization, lower power consumption and reduce operational cost. Such benefits allow greater client satisfaction.

In optical communication, a driver towards greater fibre optic bandwidth utilization has been the Wavelength Division Multiplexing (WDM). WDM enables to carry information in multiple channels through one fibre optic. For instance, Dense WDM (DWDM) divides the total bandwidth per fibre (Tbps) in individual channels (Gbps) per optical wavelength. In DWDM it is possible to have 192 channels at 10 Gbps per fibre. The industry has reached 400 Gpbs specified by the IEEE Std 802.3bs-2017 using WDM to transmit over eight different wavelengths at 50 Gbps and reach up to 10 km. The Ethernet 200 Gbps standard specify four lanes of 50 Gbps over on or four single-mode fibres.

Using WDM at transport links, flexible-switching technologies should be used at inter-connection nodes avoiding optical/electrical conversion at each hop. There are three switching technologies: Optical Circuit Switching (OCS), Optical Burst Switching (OBS) and Optical Packet Switching (OPS). Depending on the switching technology used, the basic switching unit are light-paths/circuits/wavelengths in OCS, optical packets in OPS and optical bursts in OBS (burst is a group of optical packets).

The switching technologies mention above differ from one another in the switching unit (light-path, optical burst, optical packet), the switching speed (minutes-hours, milliseconds, nanoseconds), implementation cost and market availability (high, low, low). OCS has been widely deployed at transport optical networks, while OPS and OBS has been implemented in research and development testbeds from network equipment prototypes and fully functional optical networks testbeds.

Table 3.1 Switching technologies comparative

Optical Switching	Bandwidth usage	Latency	Optical Buffering	PROC/SYNC overhead	Adaptability
Circuits OCS	Low	High	Non Required	Low	Low
Burst OBS	High	Low	Non Required	Low	High
Packets OPS	High	Low	Required	High	High

In this context, there is a motivation to build all optical transport network using either OPS or OBS. Therefore, the transmission of information in an optical burst mode has been studied and implemented based on OPS and OBS network architectures proposals. Mayor concepts and technology surround OPS and OBS are then implemented to the optical transport network.

This chapter presents work done at the initial stages of this research project by presenting the overview carried out about the architecture, algorithms, protocols and QoS proposed to OBS networks. Following the OBS network theory, it is presented a simulation tools review for OBS regard this work and the alternative to evaluate video traffic over OBS networks under a OBS network topologies, video traffic sources and algorithms. The QoS and QoE video traffic over OBS evaluation methodology and setup is presented as a methodology that have allowed to undertake the experiments that lead to the results presented in this chapter. That results are the effect and optimization of the burst assembly algorithms for video traffic in OBS and the alternatives to provide QoS and QoE to the video traffic by contributing with proposal about an adaptative burst assembly algorithm and a preemptive scheduling algorithm for OBS networks.

3.1. Optical Burst Switching Networks

Optical Burst Switching networking has been investigated since Qiao [18] propose this switching paradigm as a path towards the building of all-optical networks. Different proposals have been presented in many publications in order to construct an optical transport network based in OBS [18][19][20]. The proposals focus on architectures, protocols, algorithms and physical elements of the OBS network, for instance, important building blocks of an entire OBS network, such as routing algorithms, switching strategies, scheduling algorithms, signalling protocols, control plane, energy efficiency, resilience, QoS, and so on. In spite of these vast efforts, there is not a final OBS Network Architecture defined and there are important implementation aspects to deal with.

Therefore, studies about common application traffic over OBS networks over a partial architecture are being held in order to test proposals already published and to contribute to the definition of the OBS network architecture and protocols.

The main aspects of the OBS network's structure are presented in this subsection as a review. Also, there is a special attention in elements regard to the QoS mechanism for real time and video traffic. More details about OBS networks could be found in a wide range of research literature that is cited through the development of the document.

3.1.1. OBS network architecture

OBS Networks are all optical transport networks were the switching basic unit, at optical nodes, are optical bursts. An optical burst is a group of multiple data and voice packets. Then, each node must forward optically burst between input and output ports, through an internal optical switching fabric, to guide the optical burst from source to destination across the network. Figure 3.1 presents the OBS network topology.

The optical bursts are generated at edge nodes of the OBS Network. Data from clients are aggregated at the ingress node (an edge node) conforming an optical burst to be transmitted. Therefore, the data is forwarded through the network in an optical burst, without media conversion, to the egress node where the burst is disassembled. The data aggregated in each individual burst have the same edge node destination. OBS Network architectural and functional features are presented below:

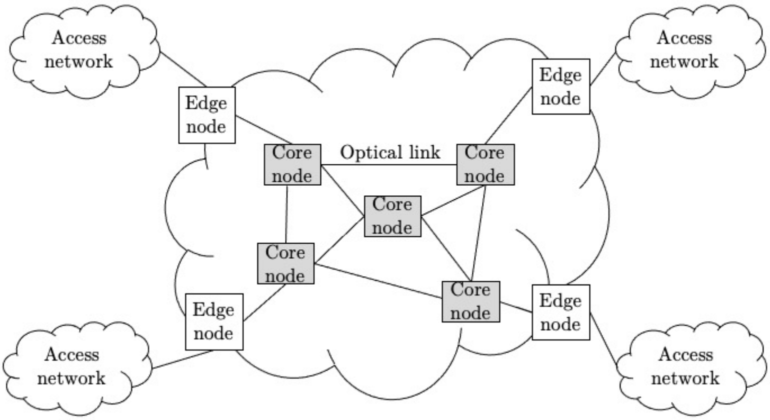


Figure 3.1 OBS network architecture

- Client data from metropolitan or access networks are assembled into burst and disassembly in individual packets at the edge nodes.
- Data having the same egress node destination is aggregated into the same burst. The assembly unit is called *burstifier*.
- Optical burst based statistical multiplexing happens at OBS network core, the core nodes are optical cross connectors (OXC).

- A Burst Control Packet (BCP) is generated for each burst. The BCP or Control Header Packet (CHP) carries signalling information, is sent before the burst to make in advance resource reservations.
- Optical bursts are transmitted using many optical channels and the optical control message are transmitted in a few optical channels. Signalling is out of band.
- Control messages need optical to electrical media conversions at core nodes control units, while the optical burst remains at optical domain at the core.
- The control message goes before the optical burst through a control channel to make resource reservations down the path. The time between the control message and the corresponding burst is called *offset time*.
- The control plane could be electronic/optical while the data plane is all optical.
- Each core node must run scheduling algorithms to make channels reservations for forwarding.
- The routing schemes could be static or dynamic based in shortest path first.

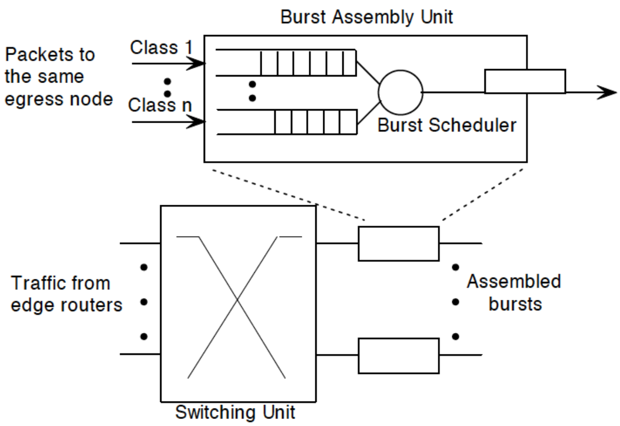


Figure 3.2 Edge node

3.1.2. Burst assembly in OBS networks

At the edge of the OBS Network, ingress nodes as a function shapes data coming from client access networks into optical burst to be send to the core OBS network. Basic communication units from access networks, mainly electrical packets are assembled into burst and transmitted as optical bursts. Here at ingress nodes medium conversion take place, from electrical to optical domain.

The data arriving at the ingress node are classified and ordered to be assembled into a burst based on their destination and class of service if QoS policy is possible depending on the algorithm, there could be an assembly queue for each class of traffic. A *classifier* and *burstifier* at ingress node a) that creates the optical burst and its corresponding control packet, b) sets up the offset time between the optical burst and optical control packet, c) schedules the optical burst transmission at output links, d) sends the optical burst and control packet down to the OBS core network. The burst assembly and disassembly process could be seen at Figure 3.3 and Figure 3.4.

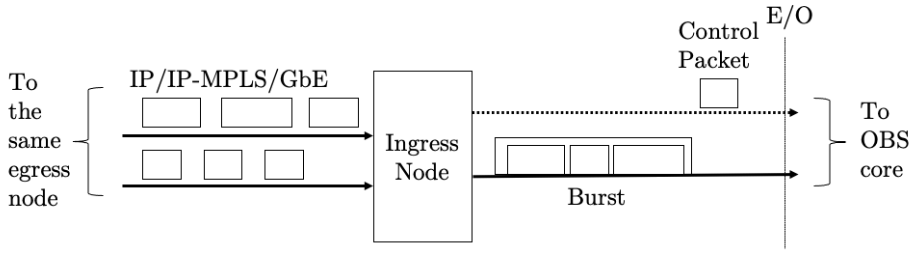


Figure 3.3 Burst assembly process

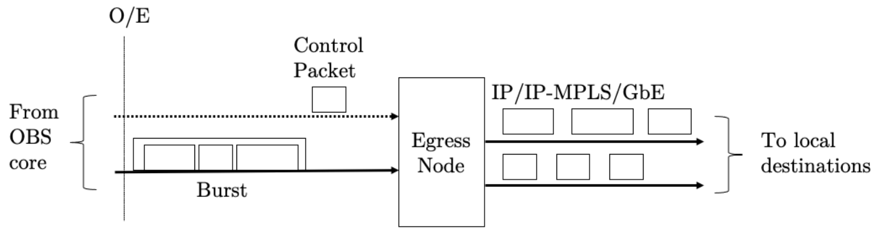


Figure 3.4 Burst disassembly process

The aggregation process is supported by burst assembly algorithms, where each assembly algorithm performs this task with different strategies. There are optical burst assembly algorithms alternatives mainly a) length-based burst assembly, b) time-based burst assembly, c) hybrid burst assembly time-length and d) adaptive burst assembly.

- **Length-based burst assembly algorithm**: In the length-based burst assembly technique the algorithm works as follows: first, the packets coming from different sources are buffered into assembly queues. Then, based in the algorithm criteria, packets in the assembly queues are assembled into burst. The algorithm criteria consider the size of the assembly queue to conform the burst. When the assembly queue reaches a size fixed threshold (Kbytes), the burst is forwarded to an output queue, and immediately a Burst Header Packet (BHP) is sent to do the reservation in the down link to the corresponding burst. Once the reservation is made the burst is send through the OBS core network, all optically, to the egress edge node. It is clear that, because of the assembly process, the packets could experiment an extra delay. The delay could be variable depending on the amount of traffic load and the variability could produce delay not acceptable in real time and video traffic transmissions. In Figure 3.5 is a representation of an example where incoming UDP packets arriving to the ingress node are grouped to conform a burst with same length.

- **Time-based aggregation strategy:** In this alternative, the burst formation time is constant. It is equal to the timeout established in the burst assembly algorithm. The burst assembly queue is equipped with a time counter. When a packet arrives at a burst assembly queue, the time counter is started. Then, incoming IP packets (UDP packets from video frames) are buffered in the burst assembly queue until the timeout is reached. Immediately a burst is created with the contents of the burst assembly queue, and it is queued for transmission on the data channel. The time counter is restarted to zero and remains in this value until the next packet arrives to the queue. The assembly strategy must guarantee no segmentation of IP packets when the timeout is triggered. Figure 3.6 shows that generated burst could have different lengths depending on the size and number of the incoming data during the assembly time.

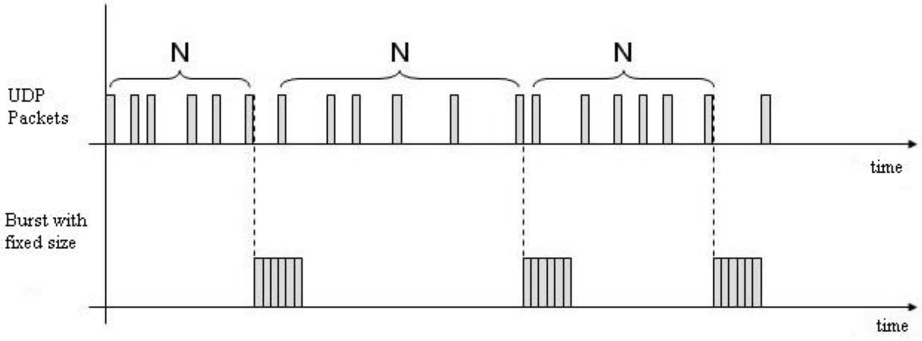


Figure 3.5 Length-based burst assembly process

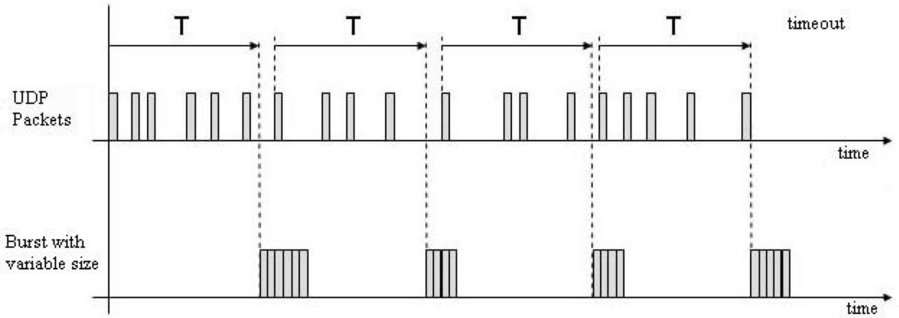


Figure 3.6 Time-based burst assembly process

- **Hybrid burst assembly algorithm:** The hybrid burst assembly process is being proposed in order to use a time or length based decision to finalise the burst assembly process. On one hand, a time threshold to guarantee an optimal delay. On the other hand, a length threshold to have burst length into controlled values [21][22].

- **Adaptive burst assembly algorithms:** Adaptive burst assembly algorithms generate each burst in a more controlled manner, trying to be adaptive to the actual traffic load. They adapt its operation to the packets arrival rate at the ingress node or other changing parameters. Therefore, the length and time thresholds are adaptive to have a great deal in terms of delay and size of each burst.

  There are some examples of adaptive burst assembly algorithms, such as the Adaptive Assembly Period (AAP) [23], Dynamic Time Burst Assembly (DT-BA) [24], Enhanced Burst Assembly [25], Adaptive Burst assembly based in Buffers Overflow [26], TCP Window-based burst assembly scheme [27], assembly algorithms with Burst Length Prediction [28].

  In AAP for each new burst a time threshold is calculated dynamically. Each new time threshold is calculated for each burst assembly queue considering the average burst length of the previous bursts and the available bandwidth.

  DT-BA aims to minimize the energy consumption and offers delay guaranties. The algorithm computes the assembly time threshold according to the queue state and a traffic baseline. When the assembly process takes place, the algorithm measures the energy consumption. If this measure begins to rise, the algorithm reduces the time threshold to reduce the queue size.

  The Enhanced Burst Assembly algorithm makes prediction of traffic to compute a minimum burst length and set the appropriate offset time. In this case, when burst assembly process starts, the burst length is known.

  In the Adaptive Burst Assembly based on Buffers Overflow, the traffic mean and variance are calculated to a determinate queue size and buffers overflow estimation. This information is used through a feedback control loop to dynamically adapt the time assembly threshold. The control loop permits to maintain a bounded delay at the assembly queue.

  The Assembly Algorithm with Burst Length Prediction is proposed to improve a time-based burst assembly algorithm, because in this strategy the time of burst assembly process is known, not the burst length. The burst length depends on the traffic load. With burst length prediction resource reservation are made before the assembly process ends and the BHP is sent with a greater offset time.

3.1.3. Resources reservation and signalling in OBS networks

When the optical burst is ready to be sent through the OBS network, the optical path must be reserved from ingress node to the egress node, that implies making channel reservation at each interconnection link and switching resources assignment at each hop. Remind that, the optical burst is send in one go, end to end, at speed of light in the optical fibre, therefore, at each hop is not possible to have any kind off buffering. Later, in this section, it is presented alternatives to buffering. To deal with the reservation task there are a signalling strategic that are supported by the control message that is send out before the optical burst. That lead the development of resources reservation protocols and strategies based on the signalling strategies like *tell-and-wait* and *tell-and-go* [18]. The signalling strategies for OBS are presented below, after a short explanation about the a) tell-and-wait (TAW) and c) tell-and-go (TAG) signalling concepts.

- **Tell-and-wait:** In TAW when an ingress node is ready to send out an optical burst, the node tries to make the resource reservations (bandwidth, optical channel, switching elements) all the way down to the egress node sending a short reservation message. Each node along the path receives the reservation message and make the corresponding reservations. When the resources reservations are done along the path, an acknowledgment message is sent back to the ingress node. Then, the ingress could transmit the optical burst. If the reservation fails at a core node because the lack of resources, a negative acknowledgment (NACK) is sent back to the ingress node freeing resources. Then, the ingress node starts the reservation process after a back-off time [29].

- **Tell-and-go:** When TAG signalling strategy is used, the ingress node starts the optical burst transmission along the path without a previous resource reservations confirmation. The BCP is supposed to have make the resource reservation. If there is no resources availability in a core node, the optical burst must be delayed or dropped in a core node. In the case of a dropped optical burst, the ingress node is informed by the NACK message to retry retransmission after a back off time [18].

After the explanation of TAW and TAG signalling strategies next there are a brief exposition of resource reservation protocols used in OBS networks, the signalling protocols are based in tell-and-go scheme.

- **Just-in-time (JIT):** JIT makes possible perform resource reservation in advance. Each node is configured when a control message arrives that is sent before the optical burst, with an offset time. The protocol is called JIT because the recourse reservation is made just before the optical burst arrives [30]. In this protocol there are two control messages, a SETUP message and a RELEASE message, the second one is sent back to the ingress node realising resources. The offset time is enough to make all the reservations before the burst is forwarded in each core node.

  Some implementations of JIT could use a central scheduler, where each control messages are sent to that central control unit. The scheduler informs to each node about the resource reservation needed. To have better scalability, distributed schedulers are implemented on each node. The schedulers are time synchronized and have the global state information about the links and nodes, but the reservation is made when the control message arrive.

- **Just-enough-time (JET):** JET protocol is the implementation of JIT hop-by-hop basis. Each control message carries information about the time reservations and makes an anticipated reservation for the incoming burst, but only for the exact time needed, so it is no need for a RELEASE control message [18].

3.1.4. Channel scheduling in OBS networks

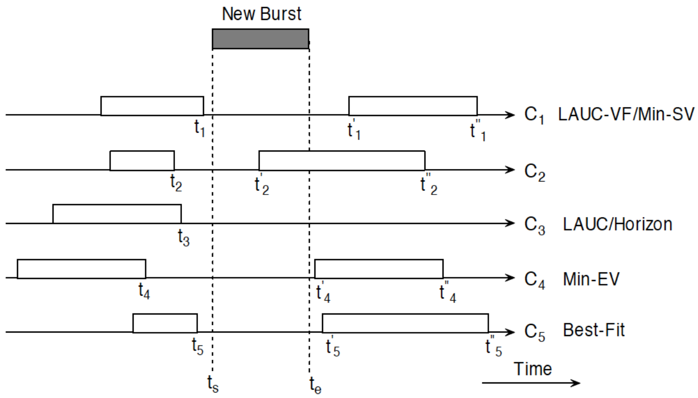
In electronic routing and switching when contentions happen between packets at each node, it could be solved by electronic buffering. Not the case for optical burst contention at core nodes in OBS networks. Contention resolution have to be done in a different manner. Therefore, there are proposal to make a very precise scheduling on each core node to avoid burst contentions that could be difficult to solved. Nevertheless, there are some contention resolution alternatives that could be implemented to reduce the burst drops.

In the literature, there are some relevant scheduling algorithms for OBS networks. Before explaining them, there is a concept called *Scheduling Horizon* that must be mentioned. If it is possible to use Wavelength Conversion (WC) the optical burst could be forwarded and transmitted over multiple wavelengths over multiple optical fibres. The scheduler could choose the channel taking into account previous reservations, so the *Scheduling Horizon* is "the latest time that a lambda channel is scheduled to be used". Below there are the relevant scheduling algorithms in OBS networks.

- Latest Available Unscheduled Channel algorithm, LAUC/Horizon: Using the LAUC/Horizon a unique *horizon* is maintained for each lambda channel. When a new burst reservation is needed, only the channels with a *scheduled horizon* that is before the time when the burst is supposed to arrive are considered available. Each time a new reservation is made, the scheduled horizon is updated. The main idea of the algorithms is to reduce the time gaps/voids between reservation.

- Latest Available Unscheduled Channel algorithm with Void Filling LAUC–VF: There is a possibility to used time gaps/voids between already scheduled optical burst. LAUC–VF considers the scheduled horizon and the complete scheduling window for each channel at node to decide if the new optical burst has to be scheduled after a horizon or in a gap/void. This algorithm has implementation variants: LAUC–VF with a) Min-SV (Minimum Starting Void), b) Min-EV (Minimum Ending Void) and c) Best Fit.

Figure 3.7 is a representation of available channels (C_1 to C_5) at one core node to make reservations to forward and transmit an incoming burst. There are reservations made for burst that are coming indicating the time needed to maintain the reservations. If a new reservation is needed for a new incoming burst the channel scheduling algorithm make the reservation base on the horizon criteria and a strategie to reduce the time void/gaps without reservations. In the Figure 3.7 the new burst will be scheduled for instance in C_3 if LAUC/Horizon is implemented or C_5 if best-fit is implemented.



source: [18]

Figure 3.7 Scheduling algorithms in OBS networks

3.1.5. Contention resolution schemes in OBS networks

In an OBS network, where optical bursts are sent end-to-end through the network, usually contention for resources and burst drops happens because of different aspect, for instance: a) there is no optical buffers at core nodes, b) operational modes of the signalling protocols like JET where the burst is sent without resource reservation confirmation. To solve the possible burst contentions and avoid burst drops there are some techniques that could be applied:

- **Deflection routing:** A burst is sent out over a non-preferred channel/time like a) alternative lambda: the burst is sent out over a non-content lambda, b) alternative port/fibre: the burst is sent out over an alternative route with its corresponding burst, c) burst delaying using Fibre Delay Lines.

- **Dropping:** A solution to a contention is to drop one of the burst in contention. The algorithm decides which burst should drop to not generate so many losses of data or waste of resource already used.

- **Preemption:** There is a possibility to cancel a channel reservation to grant resources to a new contention burst based on priority policy or traffic profile. The high priority burst preempt a low priority burst.

- **Burst splitting:** Another way to solve contention is to split the burst in multiple bursts and then deflect, drop or preempt the resulting burst. Implement a contention scheme depends on the burst assembly algorithm, scheduling algorithm, routing scheme selected to the operational OBS network.

3.1.6. Routing and wavelength assignment in OBS networks

In OBS networks the routing process decide the set of links and nodes that will be used in the transmission of an optical burst from the edge node to the core nodes and to the egress edge node. The routing process is supported by the signalling and scheduling mechanisms. The routing process could be static or dynamic. On one hand, when static routing is used the routes are calculated when a OBS networks is starting the operation and stay static under operation. If there are a link failure, the static routing matrix is computed again. On the other hand, dynamic used dynamic parameters that are updated during operation. Therefore, the routing reacts to the network state and to network congestion when traffic engineering elements are used.

In OBS networks studies, usually it is used a hop by hop routing process. That is, each node decides which will be the next hop. The main advantage of this routing strategies is flexibility because each node makes real time decisions. In OBS networks that implement signalling protocols without confirmations, like JET, the offset information is known in advance. This offset information is used by each node to makes decisions, then there is the possibility of make the routing decision and update the offset time [31].

Other proposal takes into account source routing or centralized routing schemes. In the former, the ingress node computes the overall path like other connection oriented networks such as MPLS or GMPLS [32]. In the latter, could be implemented a specific element to compute the routes. The IETF has define a Path Computation Element [33] to be used in optical switching networks (WSON) [34]. This routing approach has been proposed to an OBS network variant called WR-OBS, Wavelength Routed Optical Burst-Switched Network [35].

3.1.7. Quality of Service in OBS networks

Provide Quality of Service in OBS networks is one of the most challenging problems because the lack of buffering in the core nodes. Burst cannot be stored, so contention resolution schemes are difficult to implement. Added to this, assembly and scheduling algorithms that are chosen for its small computational complexity has best effort nature, and they do not support QoS service differentiation.

There are solutions to contribute to the QoS provisioning in the OBS network based on its basic architecture explained before during this chapter. The main proposals tackle OBS network specific building blocks, either burst assembly algorithms, scheduling algorithms, contention resolution schemes, and so on [20]. There is not an entire OBS network architecture with QoS as a whole. The QoS in OBS networks proposals listed next are alternatives studied during the development of this research that where contrasted to formulate the alternatives for QoS in OBS proposed.

- Offset increased mechanism.
- Differentiated burst assembly.
- Burst cloning.
- Differentiated routing.
- Differentiated wavelength assignment.

- Burst pre-emption of scheduled burst.

For instance, Prioritized JET (P-JET) [36][37] implements strict priorities by differentiating the offset time of priority classes, but with a big delay overhead; and PLAUC-VF uses burst pre-emption adding flexibility to the burst scheduling process. It allows the re-arrangement of already scheduled bursts in order to accommodate reservation requests which would be otherwise blocked, and at the same time it makes QoS differentiation possible [38].

The OBS network architecture, protocols, and the QoS in OBS networks proposal explain in this section, are alternatives studied during the development of this research, the related bibliography explain in more detail each aspect of the OBS research efforts.

Having studied the OBS network proposal and the video streaming over IP networks, activities of this project follow the route to evaluate and configure the quality of video traffic over OBS networks and then to make contributions to QoS in OBS networks. Therefore, next subsection present tools, strategies and the corresponding contributions which have been reached.

3.2. QoE & QoS video traffic over OBS network evaluation setup

In this subsection, the proposed and employed evaluation framework of video transmissions over OBS networks for this research project is outlined. First at all, summarizing the investigation carried out about simulation in OBS networks, then explaining the video traffic over OBS simulation integration and evaluation process defined. Ending this subjection, the simulation scenarios for video over OBS employed in this project is presented.

3.2.1. Simulation tools for OBS networks

Different models for analysis and simulation of OBS networks have been proposed and published in several articles. Each model is oriented to a particular network architectural aspect to be evaluated to study the behaviour of an OBS network. Either traffic analysis, QoS, queuing behaviour, burst drop policies and retransmission, burst assembly algorithms, scheduling algorithms, contention resolution, blocking probability, among other network functional aspects.

In [39] is presented a brief summary of different types of analytical models proposed for OBS networks, both simple models to observe traffic in a single link between two nodes with Poisson and Pareto traffic distribution, and such as more detailed simulation models. Also [39] emphasize that most of the analytical work done use Poisson distribution arrivals models, as well as traffic generated by Markov processes in the case of edge node analysis.

The main contribution in [39] is a detailed event simulation model of the burst assembly process of Ethernet and Internet traffic. The model takes into account the self-similar traffic by means of synthetic traffic sources to find optical burst statistical properties at the core of the OBS Network. The simulation model is a detailed and realistic event driven simulator and a OBS core traffic model. The OBS networks scenario is a Cost 239 European Network topology (connecting 11 European cities)[40]; great amount of traffic sources with variable IP packet size, scheduling algorithm with wavelength conversion (WC); and Fibre delay lines (FDLs) and QoS by mains of a controlled BHP offset-time.

The above mentioned OBS event driven simulator is built over the J-sim framework [40] developing the packet (burst) and optical links and a complete set of models such as: data traffic sources, burst assembly, edge and core nodes. The simulation process is controlled by a script where the topological OBS network structure, propagation delays, routing tables and OBS network models are set it up. Is important add that results in [39] conclude that the performance properties of the a large OBS network confirm what other researchers have found using analytical models and using event driven simulation method is better for large complex systems with self-similar traffic arrival models because with analytical models is hard to reason.

Event driven simulation has been used in some other contributions [41],[42],[43],[44] that propose analytical models. Those studies are based in the OBS network simulator develop by the Optical Internet Research Centre, OIRC. The OIRC OBS network simulator is based on NS-2 adding OBS network modules. The OIRC OBS network simulator is called OBS-NS-2. This simulator has the following characteristics: a self-similar traffic module, burst differentiation controlled by offset-time, fibre delay lines modules, LAUC-VF scheduling, burst buffering electronic at edge, queue control to the assembly process, time-slotted OBS module, UDP agents to transport CBR, exponential and Pareto traffic, NAM objects to animate the simulation, among others. Basically, OBS-NS-2 is an upgrade over an OBS-NS developed by DAWN at Meryland University.

Other work done in OBS networks use event driven simulation using the OPNET simulator. For instance, a simulation model to evaluate the performance and optimization of the optical burst size carried out by ARC Special Research Centre for Ultra-Broadband Information Networks at Melbourne University. This model is a tool to study about performance metrics such as blocking probability, network capacity utilization and end-to-end delay. This model has been used to propose and develop wavelength assignment technics to improve QoS in IP over WDM networks [45].

In order to complement the study of the OBS network simulations tools there is an interesting paper [46] that reviews simulation models that could be consulted. Due to the wide range of simulation models, it is important to define the OBS network aspect to be evaluated and make an adequate decision about the simulation tools needed in particular research.

For the purpose and objectives of this research proyect, the OIRC OBS-NS2 has been selected due to its characteristic explained before. It makes possible to evaluate different types of traffic like TCP and the UDP agents. This characteristic makes possible the video traffic evaluation by adding some modules, these are explained later in this chapter.

In the evaluation process with this simulator is possible to work with variables like burst time, size, scheduling algorithms. The OBS-NS-2 presents a set of output statistic supported by result traces that include measurements of optical burst, burst header packets and per flow statistics. An important aspect is that the OBS-NS-2 is based on the Network Simulator 2 that is open source and have a wide community working on it around the globe.

3.2.2. Video traffic OBS network simulator

The OBS-NS-2 network simulator offers support for the evaluation of IP traffic over OBS. In order to evaluate in a more complete manner the OBS network including video traffic analysis over OBS and for the purpose of these research, it has been investigated the integration of video traffic source modules to the OBS-NS-2. The objective of the research is to evaluate QoE of actual video transmission over the OBS network simulated. A video module developed for NS-2 compatible with an QoE evaluation tools has been incorporated to the OBS-NS-2.

The quality video evaluation tools investigated and incorporated in the set-up are the ones under the EvalVid proposal [15]. The EvalVid tools are a set of programs that enable to evaluate the video quality at destination of a video that is transmitted through a communications network, real network set-up or simulated network. Evalvid is used to evaluate video degradation after video coding process and/or video transmission process. The idea is to evaluate the quality of videos perceived by users. The Evalvid tools allows to measure both, QoS network parameters like burst losses, delay, throughput and jitter, and objective QoE parameters such as PSNR and MOS. QoS and QoE metric are explained in section **Error! Reference source not found.**. The video codecs s upported to evaluation are H.263, MPEG-4 and MPEG-4 (H.264). The video traffic evaluation over OBS network set-up proposed and used in this research is presented in Figure 3.8.

3.2.3. Video transmission evaluation methodology and set-up

In this subsection is outlined the methodology proposed and followed to evaluate the video streaming over OBS networks, to make video transmission quality assessments and to propose and evaluate concepts and algorithms for burst assembly and scheduling.

The Figure 3.8 shows the building blocks of the evaluation framework grouped by main functions: video codification streaming and tracing, video streaming over OBS network simulation, video reception and decodification from result traces, and the video quality assessment. Next are related steps and task of this evaluation framework showing the manner how the evaluation experiments were carried out.

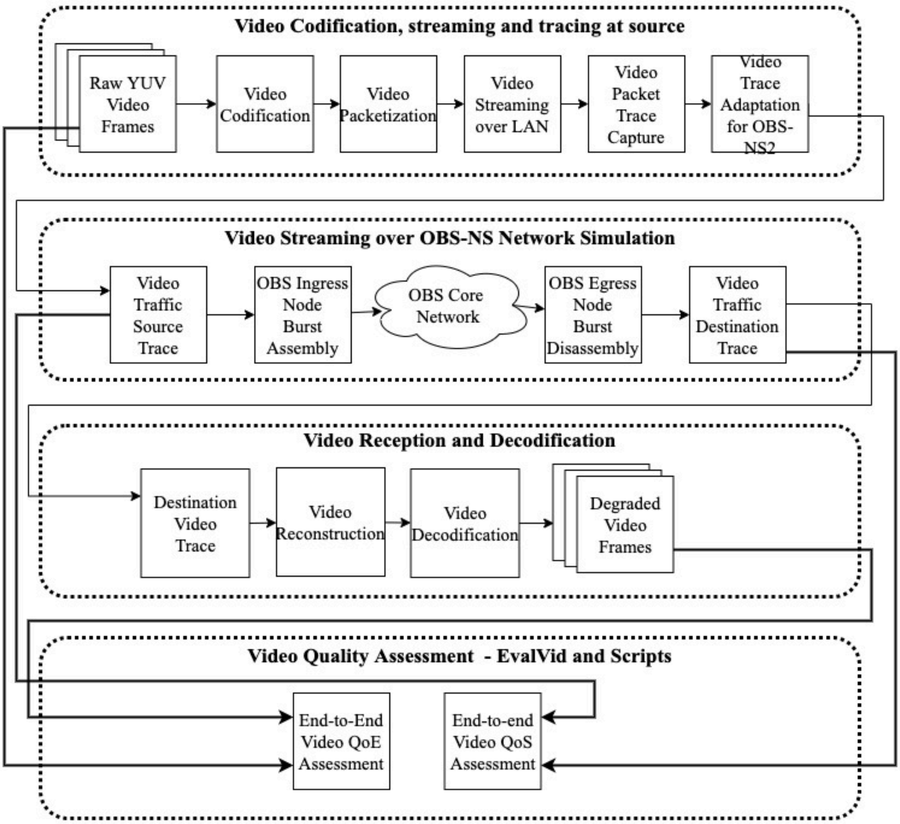


Figure 3.8 Video traffic evaluation over OBS network set-up

- **Definition of a OBS network simulation scenario:** First at all, the OBS network topology is designed for evaluation. Fixed and variable parameters of the OBS networks such as burst assembly, signalling, scheduling, traffic source, video traffic sources are assigned.

- **Video content generation:** Then, video content is prepared for simulations and test. Video sources are capture, codified, and compressed. The source video content is then adapted for transmission over IP networks, adding fragmentation and packetization information.

- **Video trace generation:** Video content is transmitted over an IP network. At the same time capturing video packets traffic driven by the video source. The setup of the video transmission is made over a real network. The result of this step is the video traffic trace containing detailed information about the video packets and characteristics at different network layers such as application, transport and network.

- **Video trace adaptation:** Video traffic traces generated from the real video transmissions are then adapted as video traffic source in the OBS-NS2 simulations. At this point there is interface between the actual video traffic and the OBS simulated network scenario in the OBS-NS2 simulator.

- **OBS-NS2 simulator adaptation:** In order to the OBS-NS2 simulator to accept trace driven video traffic source from a codified video transmission the simulator has also been adapted. Therefore, a video packet has been defined and programmed at the OBS-NS2 simulator.

- **OBS network topology building:** The OBS network topology, protocols, and algorithms to be evaluated are built. The definition includes all the OBS network scenario, traffic sources including the video traffic source.

- **Experiments by simulation:** The simulations are then executed considering the defined scenario. The simulation is automated by means of simulation scripts and operation system scripts that have been also programmed.

- **Results compilation and analysis:** After simulation process, results from traces are processed with a set of scripts to extract results. Traces are exported to traces prepared to the QoE/QoS measurements using the Evalvid tools.

- **Video reconstruction:** From traces at destination after the simulation, it is possible to reconstruct the video transmitted with losses. After reconstruction, the video is decode to obtain a video near the original. In that way, it is possible to assess the video quality at reception compere with the original video at sender side.

- **QoS measurements:** QoS metric are obtained from the simulation results, delay, losses, throughput and jitter. A set of scripts and programs are defined to measure and results plotting.

- **QoE measurements:** PSNR and MOS measurements are done comparing the video before transmission and the video after the transmission using the video recovered at reception.

- **Results presentation:** Results traces of the QoS and QoE are analysed and presented on graphics. Each process implies the development of automation scripts and the use of different software tools.

3.2.4. Simulation scenarios for video over OBS networks

Setting up the complete framework to evaluate video traffic transmissions over OBS networks in terms of QoE and QoS explained before, implies different stage of research and implementation that have been carried out. Starting with the evaluation of the OBS network with TCP traffic over OBS, setting up OBS network modules, traffic sources, result collectors, OBS network topologies, and OBS algorithms and protocols. When this first set up has been implemented the final stage has been to stablish the video content generator, the video source traffic modules, the video over OBS trace modules to be incorporated. Next are explained the basic elements of the simulation scenarios and are showed some example results from the preliminary evaluations. The next subsections present evaluations and contribution that has been developed using this framework.

- **OBS network topology**

In this research has been reviewed the OBS network topologies to be used in the simulation scenarios. OBS networks are proposed to be implemented at transport segment of an end-to-end optical network. The OBS network topologies could be found in Metropolitan Optical Networks proposals with a relationship with the application of OCS and OPS. Here there are presented the OBS Network topologies selected:

- A simple OBS network with 6 nodes: It is a simple mesh network used mainly for illustrative purposes and to obtain numerical results of complex problems. The network has 6 nodes, 8 bidirectional links and 1.533 mean distance between nodes.

- Interconnected ring: It is a metropolitan telecommunication network composed by four rings. The rings share links between the nodes 0 and 6, each the rings are interconnected through the 0, 5, 6, 12 nodes. It is typical topology that shows how it is nowadays interconnected the optical backbone networks that are multiple rings SDH/SONET and OTN. This topology has 15 nodes, 21 bidirectional links and 2419 Km mean distance between nodes.

- Backbone NSF Network: Since 1984 the National Science Foundation (NFS) has been designing a backbone high speed network to replace ARPANET and to be at disposal of all research community in USA. This is the most used topology in Optical WDM studies. The NFSNET has 14 nodes, 21 bidirectional links, 2142 Km mean distance between nodes.

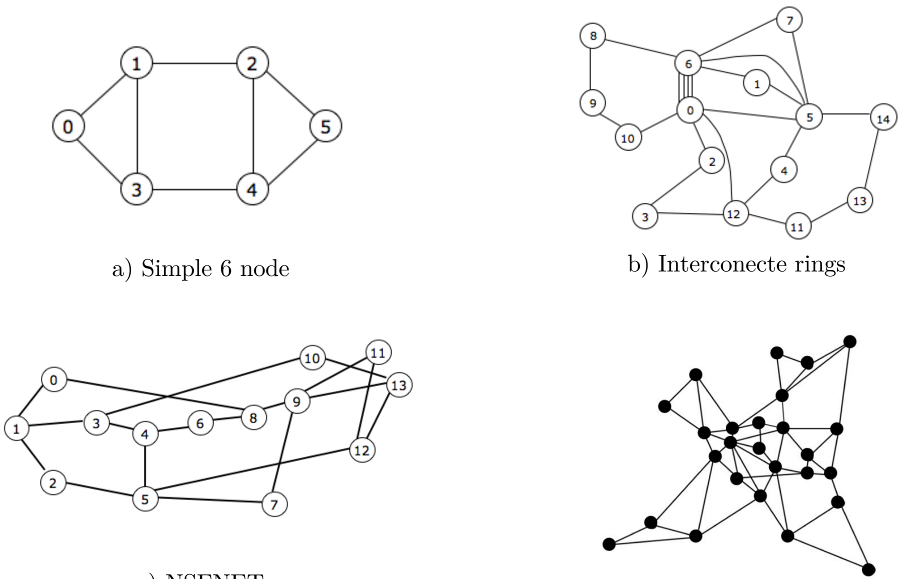
- PAN-European network topology (Cost 266): European network defined under the COST projects. The network has 26 nodes.

The most used OBS network topology in the research literature has been the NSFNET network topology because it has an adequate number of nodes and links and is similar to a real WDM optical transport network. NSFNET OBS characteristics are showed in Figure 3.9 c) and Table 3.2.

The NFSNET network model connect cities across the USA, each link has different length, so the propagation delay is different in each link, Table 3.3 shows the delay information.

Table 3.2 NSFNET OBS network

Network parameter	Value
Edge nodes	14
Core nodes	14
Channel bandwidth	10 Gpbs
Link delay by default	1 ms
Control channels per data channel	1
Data channel per link	64



a) Simple 6 node

b) Interconecte rings

c) NSFNET

d) PAN-European COST 266

Figure 3.9 OBS network topologies

Table 3.3 NSFNET OBS network propagation delay

Link	Delay - ms	Link	Delay – ms	Link	Delay – ms
Node 0 – Node 1	7	Node 3 – Node 10	18	Node 8 – Node 9	5
Node 0 – Node 2	10	Node 4 – Node 5	8	Node 8 – node 11	2
Node 0 – Node 7	16	Node 4 – Node 6	4	Node 8 – Node 12	2
Node 1 – Node 2	4	Node 5 – Node 9	7	Node 10 – Node 11	4
Node 1 – Node 3	5	Node 5 – Node 13	12	Node 10 – Node 12	5
Node 2 – Node 5	12	Node 6 – Node 7	5	Node 11 – Node 13	2
Node 3 – Node 4	4	Node 7 – Node 8	5	Node 12 – Node 13	1

- **Video traffic sources**

To complete the scenario for simulation to evaluate video traffic over OBS networks the video traffic sources has been generated as a traffic traces from actual video transmissions of sample video. This allow to build the video at reception that has been transmitted over the simulated OBS network making possible the objective and subjective video quality evaluation. Below is explained the video content generation, and the NS2 video trace adaptation to finally rebuild the video at reception:

- From CIF.264 to YUV: Selection of a source sample videos in YUV format [47] or CIF.264 to YUV video conversion using the ffmpeg tool.

- From YUV to MPEG4(m4v): Video codification in MPEG4 standard using ffmpeg.

- MPEG4 Video (m4v) to MP4 video container (mp4) with RTP adaptation: The MPEG4 video is prepared to be transport by RTP/UDP packets using the MP4Box Evalvil tool.

- MP4 video to video trace: MP4 video is dumped into a real IP network, transmitted to a destination. The video UDP packets are captured, and the video trace and the NS-2 compatible video trace is generated. Trace in packet mode, showing the information about each UDP packet, or the trace in video frame mode with represented information about each video frame information. This is accomplished through the mp4trace Evalvid tool.

- **OBS network algorithms and protocols**

  The OBS network topology used in the simulation has been the 14 nodes NSFNET and the traffic source has used the video traces explained before. In terms of OBS network has been taking into account the length based and time based burst assembly algorithms, JET signalling, LAUC–VF scheduling algorithm.

At this stage of the development of this project an evaluation framework is proposed to assess video traffic transmissions over OBS networks. Thus, making possible the evaluation and contribution to QoS and QoE alternatives to video traffic by means of an adaptive burst assembly algorithm and preventive scheduling algorithms proposals.

The simulation environment has been set-up at laboratories of the Communication Multimedia Research Group at iTeam Research Institute of the Universitat Politècnica València and the UNITEL Research Group at Universidad Santo Tomás de Bucaramanga. Later in the next subsection video over OBS simulations and the burst assembly and scheduling algorithms proposal is presented.

3.3. Effect and optimization of burst assembly algorithms for video traffic transmissions over OBS networks

This subsection describes the use of the evaluation framework and a first contribution to the evaluation of the QoS and QoE to video traffic over OBS considering the burst assembly process.

OBS has been proposed as a switching paradigm for the Internet backbone for the next generation all-optical Internet [18]. Due to the increase of traffic coming from applications such as Video on Demand, Voice over IP, Cloud services, online gaming, that have delay constrains and demand a lot of bandwidth, it is important the evaluation of the behaviour of real time traffic over OBS networks. Through the evaluation it is possible to determinate the convenience of assembling into burst, that kind of traffic and to determinate if the optical burst switching proposal is suitable to carry the growing multimedia traffic in the Internet backbone.

In this step of the research, evaluation of the effects due to the aggregation process at the edge node of the OBS network is done using two different burst assembly algorithms, length-based burst assembly algorithm [20] and time-based burst algorithm [50]. The focus lies in finding if these two basic assembly strategies fulfil quality video transmission requirements like frame delay and frame losses.

There are research studies about the impact of burstification process that expose problems that result from the assembly process. An example of such problems is related to TCP performance [51][52]. The studies have been concluded that the burst assembly process introduce an unpredictable delay, that challenges the window mechanism used by TCP protocol for congestion control [53]. The extra delay is interpreted by TCP congestion control as a network congestion situation when the congestion is not really happening. In [27] has been proposed a new TCP version that could take advantage of the broadband optical networks, that is, using large congestion windows to deal with the low throughput in the TCP classic transmissions. Then, there are studies in TCP over OBS, but performance evaluation of real-time traffic over OBS has not been thoroughly studied. In this work, the focus lies on the problems resulting from an assembly process at the edge node of the OBS network when digital video traffic flows are the main load and the protocol used to carry the video frames information is the User Datagram Protocol (UDP).

The remainder of this section is organized as follow: subsection 3.3.1 explains the simulation scenario, traffic characteristics, network topology, and presents the evaluation metrics used in this evaluation. In subsection 3.3.2 results obtained by simulation of the video quality transmission using a length-based burst assembly strategy are explained. Subsection 3.3.3 presents the results of the evaluation using the time-based assembly algorithm as a burst aggregation scheme. Finally, subsection 4 presents the conclusions of this study.

3.3.1. Simulation scenario

In the study of the behaviour of the video traffic over OBS networks that implements either, length-based burst assembly algorithm or time-based burst assembly strategy, it is used the National Science Foundation OBS network topology (NSFNET).

The NSFNET OBS network topology presents the following architecture: 14 nodes, 8 of the nodes are edge nodes and 6 are core nodes. The data links between nodes are bidirectional at 40 Gbps with two channels available per link. One wavelength is dedicated to data channel and the other one to control channel. In this topology, each data link presents different propagation delays that represent different distance between nodes. See Figure 3.10

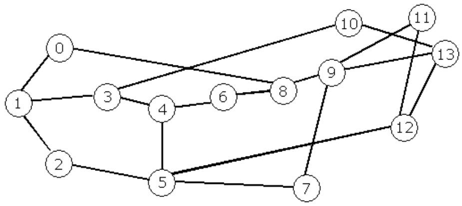


Figure 3.10 NSFNET 14 nodes

Two simulation scenarios are implemented, the first one to evaluate a length-based burst assembly algorithm and the second one to evaluate the time-based burst assembly algorithm.

As a traffic source, it was used video traces from video samples using in video codification evaluation [47]. The videos are codified before transmission using the MPEG-4 standard and then hinting to send in UDP packets with a maximum packet size of 1024 bytes.

The simulation process, evaluation and results were obtained using the video evaluation framework proposed in section 3.2.3, NS2, OBS module, video evaluation tools and algorithms for results analysis.

With each burst assembly scheme, different simulations are done varying the number of traffic sources, 10, 50 and 100 video flows to make an evaluation that take into account the amount of video traffic. Other parameter that is dynamically modified through the simulation is the maximum burst size that could be reached. So, it is possible to find the optimal burst size that assures quality video transmissions, in the case of length-based burst assembly strategy. Results with maximum burst size between 20 Kbytes to 700 Kbytes are obtained and are presenting in the next section. In the case of time-based burst assembly algorithm, the parameters variables in each simulation are presented in detail in subsection 3.3.3.

The video evaluation process of video streaming over OBS networks carry out in this work is as follow: The reference video, one original YUV file (raw video), is encoded with an open source MPEG-4 encoder, in this case the *FFmpeg* program, which in turn generates an encoded video stream. The encoded video stream is read by the video sender to generate a trace file to be used as a traffic source of the OBS ns-2 simulator. The results produced by the video sender are two trace files containing information about every frame in the video file and every packet generated for transmissions.

After finishing the simulation, the effect of streaming video over OBS could be found in the captured streaming client log files which are generated by OBS ns-2. The log file contains information such as timestamp, size and identity of each video frame. The traces and log files are used by the *Evaluate Trace* from Evalvid tool set to generate possible corrupted video files as a result of transmissions over OBS. Here, the actual calculation of frame losses and delay takes place. The corrupted video file is needed by Peak Signal to Noise Ratio (PSNR) Evaliv program to evaluate the video quality.

3.3.2. Length-Based burst assembly algorithm

In this subsection, it is presented results obtained by simulation, measurements of digital video quality metrics of the end-to-end video transmissions over OBS networks using a length-based burst assembly algorithm explained in subsection 3.1.2. The main analysis focuses on the effects of the assembly process at the edge node using a length-based burst assembly scheme. In first place, it is presented measures of the PSNR per frame, then it is exposed a delay problem analysis and finally results from frame losses are presented.

A. Peak Signal to Noise Ratio (PSNR)

As mentioned before, the PSNR has a higher correlation with the subjective quality perception; therefore, this parameter is chosen to extract information about the video quality that could be perceived at reception side by the end user. Three different scenarios were simulated, where the difference between each scenario was the number of active video flows. In each simulation, the variable parameter was the maximum burst size to conform the burst.

In Figure 3.11 to Figure 3.13 are represented the average PSNR and the standard deviation resulting in each case of number of traffic video flows. As it could be observed, depending on the number of video flows there is an optimal burst size to get in reception high quality videos. The values of PSNR between 35 and 40 dB equivalent to the highest value in the MOS scale could be reached with maximum burst size higher than 60, 280, 540 Kbytes in the cases of 10, 50 and 100 video flows respectively. The longer the number of video flow the longer of the optimal burst size to get high quality videos. In the case of small burst size the standard deviation is high and could be reduced increasing the maximum burst size.

The above mentioned implies that the maximum burst size should be dynamically established depending on the video traffic flows. If the algorithm has a fix value for the maximum burst size, when the traffic increases the quality of some videos could decrease, even to the lowest quality value.

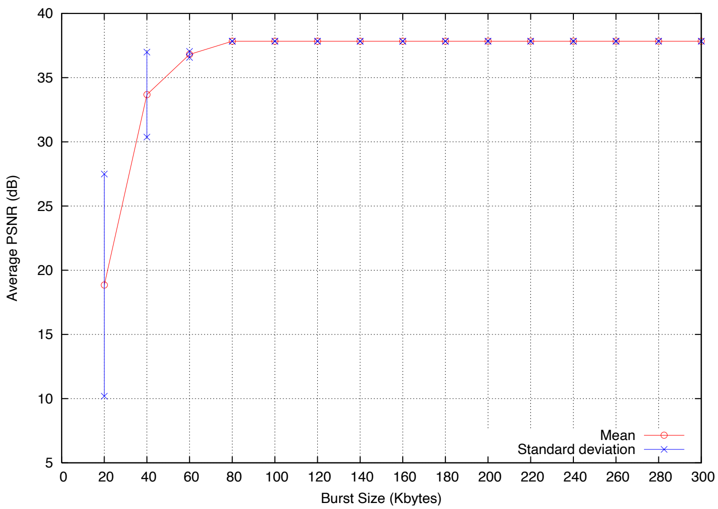


Figure 3.11 Average PSNR and standard deviation, 10 video flows.

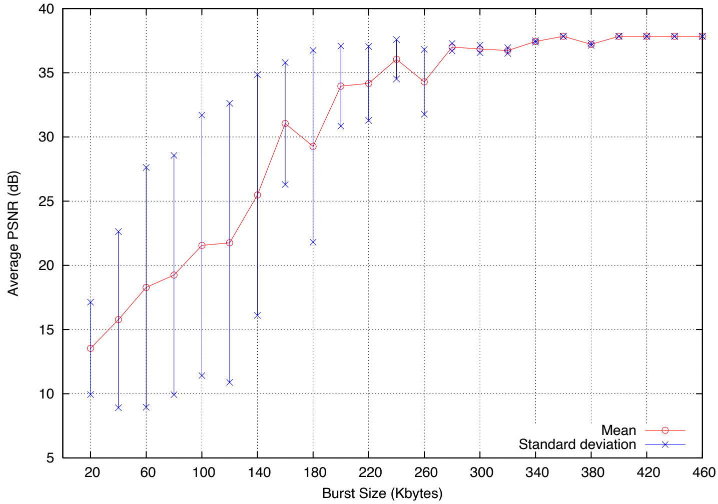


Figure 3.12 Average PSNR and standard deviation, 50 video flows

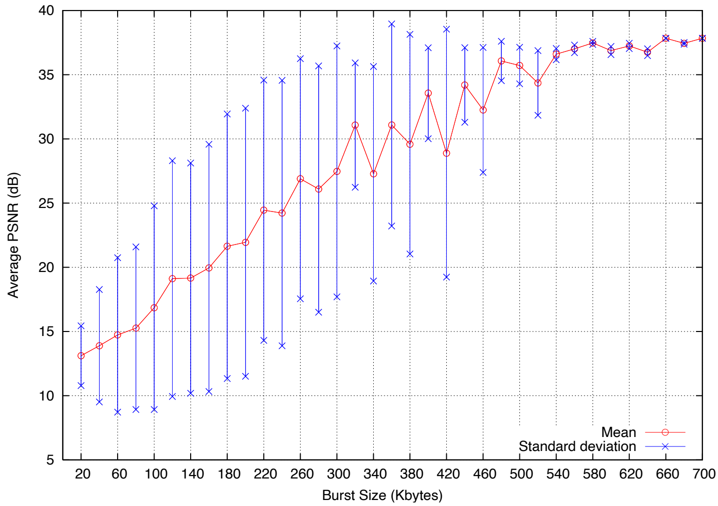


Figure 3.13 Average PSNR and standard deviation, 100 video flows

B. Frame delay

It is possible to get high quality videos with a large burst size, as explained before. But, in the other hand large burst size causes delay increases. To show that, several simulations with the same cases of video flows were made. In each case, the burst sizes that were analysed and showed are those that resulted in the previous simulation with high quality videos.

To show the problem of the increased delay causes by long burst size Figure 3.14 to Figure 3.16 represent the cumulative distribution function (CDF) of the frame end-to-end delay for the three different cases of number of video flows. The CDF was calculated taking into account all video sessions per simulation. Thus, the delay increases when the maximum burst size is increased to guarantee good video transmissions and each video flow could experiment different end-to-end frame delay.

With the evaluation it was found out that length-based burst assembly algorithm, additionally presents problems when the video flows are transmitting its final video frames because the maximum burst size is never reached. Consequently, the last burst will never send. To solve that, a timer that limits the time of assembling was fixed. That value, in the assembly scheme evaluated, is not dynamically established producing extra delay to the last video frames.

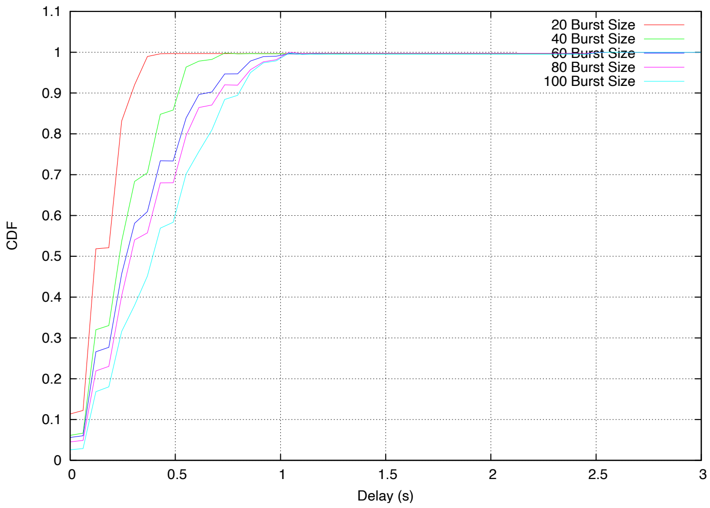


Figure 3.14 Frame end-to-end delay. Case 10 video flows

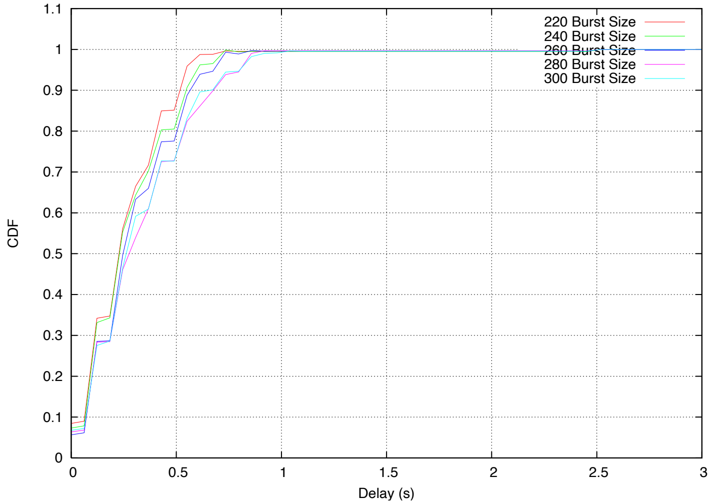


Figure 3.15 Frame end-to-end delay. Case 50 video flows

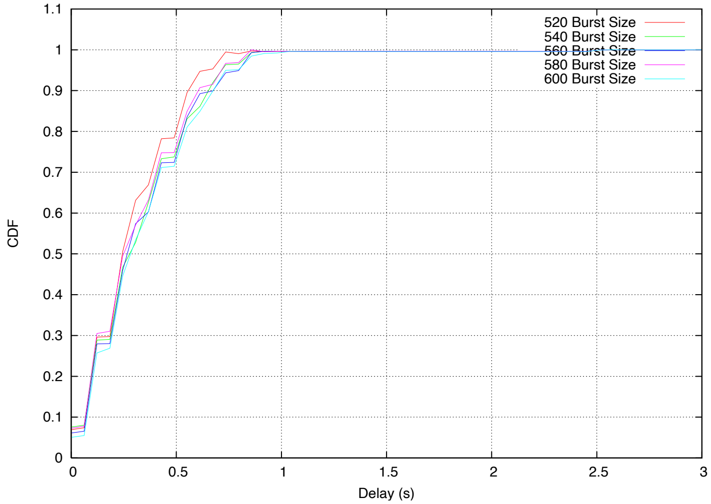


Figure 3.16 Frame end-to-end delay. Case 100 video flows

C. Frame losses

The frame losses in video transmissions are high with small burst size. Incrementing the burst size, the frame losses tend to decrease. Simultaneously, the delay increments as explained before. In the cases of 50 and 100 video flows the frames losses tend to stabilize with small values of burst size, but the standard deviation is not small until a large value of burst size. Therefore, some videos are received in good conditions and others are received with poor quality. Reason why, with a large number of video flows is needed a large burst size. In Figure 3.17 to Figure 3.19 is shown such behaviour.

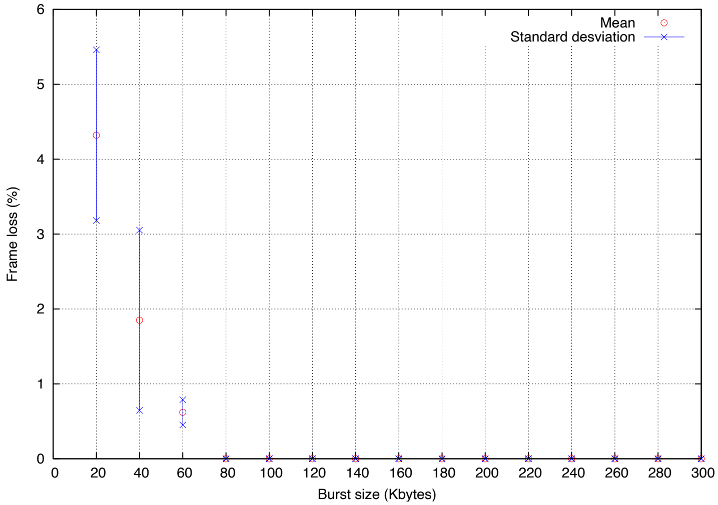


Figure 3.17 **Average frame loss. Case 10 video flows**

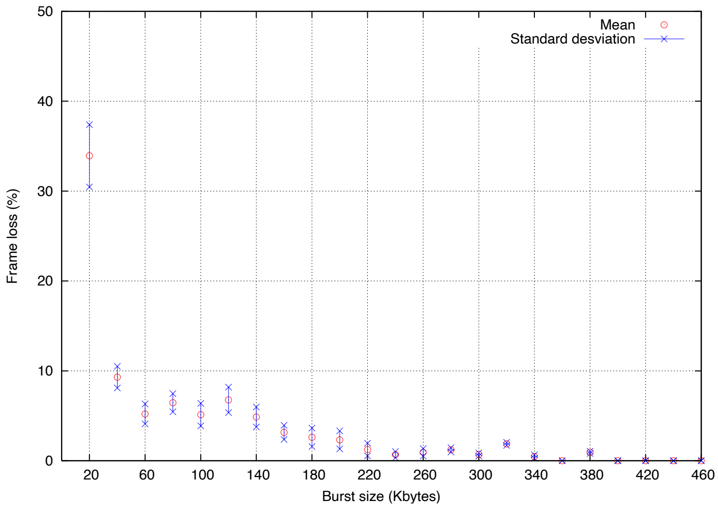


Figure 3.18 **Average frame loss. Case 50 video flows**

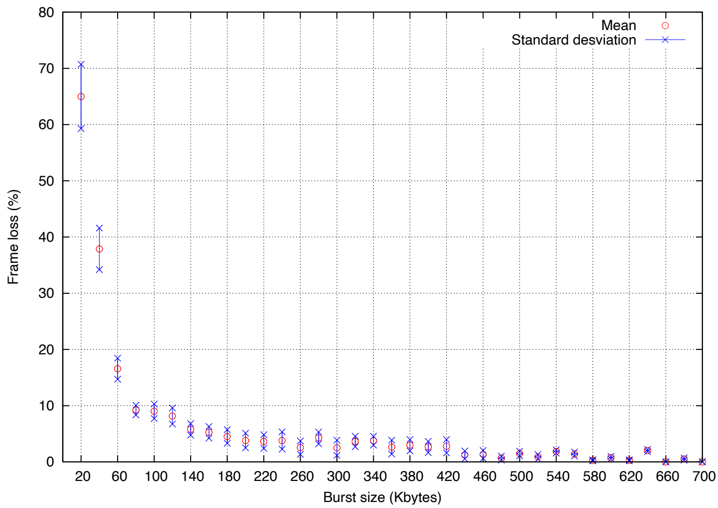


Figure 3.19 Average frame loss. Case 100 video flows.

In the next subsection, evaluation of video traffic transmissions over OBS networks using a time-based assembly algorithm is present. The evaluation is made taking into account the results obtained in the evaluation with the length-based burst assembly algorithm to conclude if one of them improves the quality of the video transmissions.

3.3.3. Time-Based burst assembly algorithm

In this subsection, it is presented results obtained by simulation of the performance evaluation metrics of digital video quality in end-to-end video transmissions over OBS networks. The main analysis focuses on the effects of the assembly process at the edge node using a time-based burst assembly strategy, this assembled scheme it is explained in subsection 3.1.2. In first place, it is presented results of the PSNR per frame and frame losses. Then, it is presented results of the frame end-to-end delay and burst size distribution using time-based scheme.

In the simulations that have been carried out to assess the time-based burst assembly strategy the timeout of the burstification process was fixed to 1ms, 10 ms and 100 ms. Each timeout was evaluated with different number of video traffic flows, 10, 50 and 100 video flows.

A. Peak Signal to Noise Ratio (PSNR) and frame losses

When evaluating video transmission with a time-based burst assembly strategy implemented, it has found that time based scheme improves the length-based strategy in terms of PSNR and frame losses. The average PSNR values obtained were constants for all videos in each case of traffic load with the different timeouts evaluated. The average PSNR of the video frames values were 37.17 dB. This PSNR values indicates high quality video transmissions.

The frames losses are related with PSNR metric. With high frame losses the average PSNR decrease and its standard deviation increase. High PSNR values indicate low frame losses. In all scenarios simulated the video transmissions have presented no frames losses. It is because, the timeouts simulated were calculated so that the burst assembly unit could assemble burst with sizes among the previous obtained optimal burst size value in the length-based scheme analysis, and also trying to get optimal burst size in any case of traffic load. The average burst size that could be reached with different timeouts (1 ms, 10 ms and 100 ms) are presented later in this section. Next it is presented the video frame delay results.

B. Frame delay

In the length-based burst assembly strategy large bursts do not degrade the video quality at reception side. But, waiting until the assembly queues reach the burst size threshold causes frame delay increases, and with optimal burst sizes the frame delay increase more. More than 70 % of the video frames suffer different delay values, up to 1 second, only few video frames have delay values less than 500 ms. Even, the video frames at the end of the video reach a 2.5 sec delay. The 2.5 seconds, the security timeout, guarantee burst transmission when the size threshold is not reach.

Implementing a time-based burst assembly strategy, the frame end-to-end delay is guaranteed. The delay through the entire path (propagation delay, burst switching times) without the burst assembly time is approximately 0.027 s. Figure 3.20 represent the CDF of the frame end-to-end delay. It shows the delay improvement using time-based strategy. As it could be seen in the figures, the frame delay varies from time propagation plus burst timeout, (for the frame packets that arrive at the beginning of the burst formation) to a value near the propagation time (for UDP packets at the burst tail). For instance, in the case of 100 ms timeout the frame delay value is in the range of 0.027s to 0.127 s.

Figure 3.20 to Figure 3.23 represent the frame delay CDF and PDF. They show the frame end-to-end delay in the three different scenarios simulated. The improvement in terms of PSNR, frame losses and end-to-end frame delay using a time-based assembly strategy is due mainly to:

- The burst timeout guarantees limited delay.
- The burst timeout permits the formation of burst with optimal sizes to carry out UDP packets of the variable size video frames when streaming digital video. Minimizing frames loses and maintaining constant the PSNR level.

It is important the exanimation of the burst size resulting in each case of number of video flows in order to contrast the burst sizes reaches with the time-based algorithm with the optimal burst size in the case of length-based strategy. The next section presents the burst size study.

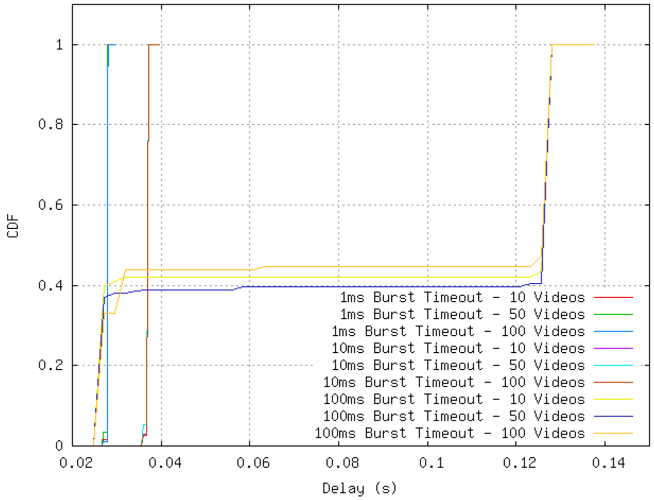


Figure 3.20 CDF of the frame end-to-end delay, assembly times 1ms, 10ms and 100ms. Cases 10, 50, 100 video flows.

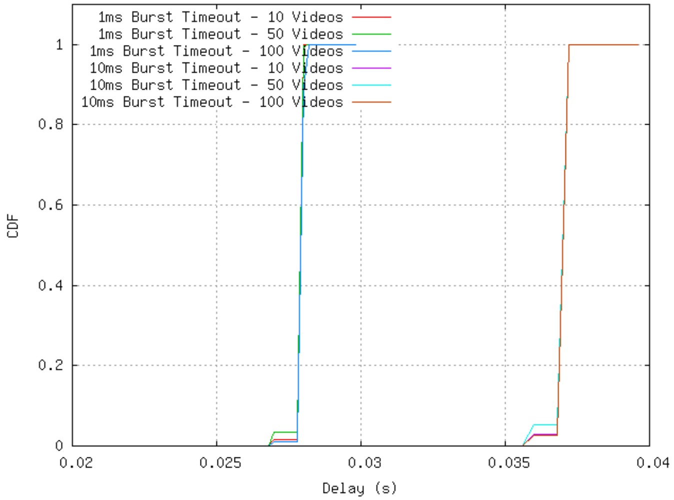


Figure 3.21 CDF of the frame end-to-end delay, assembly times 1ms, 10ms. Cases 10, 50, 100 video flows

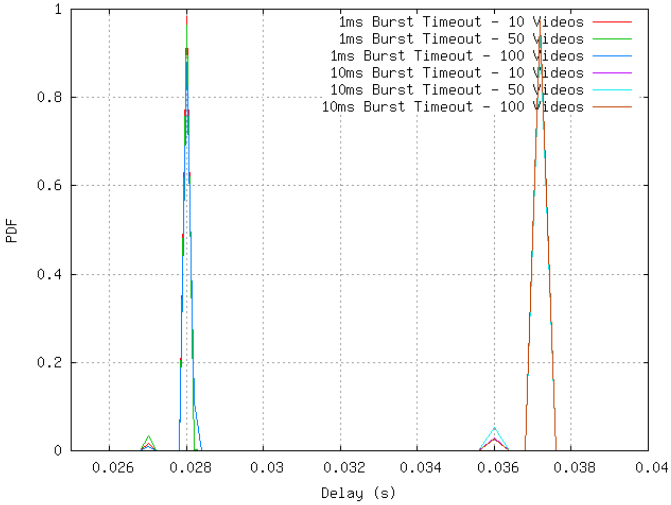


Figure 3.22 PDF of the frame end-to-end delay, assembly times 1ms and 10ms. Cases 10, 50, 100 videos flows.

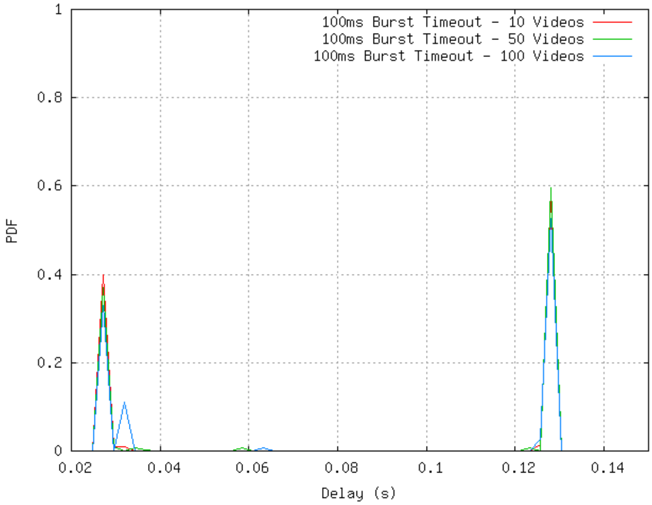


Figure 3.23 PDF of the frame end-to-end delay, assembly time 100ms. Cases 10, 50, 100 video flows.

C. Burst size

Video frames have variable size. I frames have the lowest compression of the three frame types. I frames contain by far the most information and the first I frame of the encoded video is the largest I frame in all the compressed video. Furthermore, losing one I frame will cause distortion of all frames in a GoP. When burst size threshold is not large enough to carry the largest I frame, many UDP packets are dropped during the burst assembly process; resulting in decode videos with low quality in some frames sequences. This is the case when is used length-based burst assembly strategy with fixed maximum burst size and variable video traffic because many UDP packets of the large I frames could be dropped and the I frame could not be decoded.

When a time-based assembly strategy is used, there is no burst size limitation. The largest first I frames are not dropped because they always fit in a burst. This characteristic allows the formation of burst with a size needed to carry UDP packets from frames with variable size. Therefore, using a time-based burst assembly strategy frame losses are very low and the videos in reception have high PSNR values. The end-to-end delay is guaranteed.

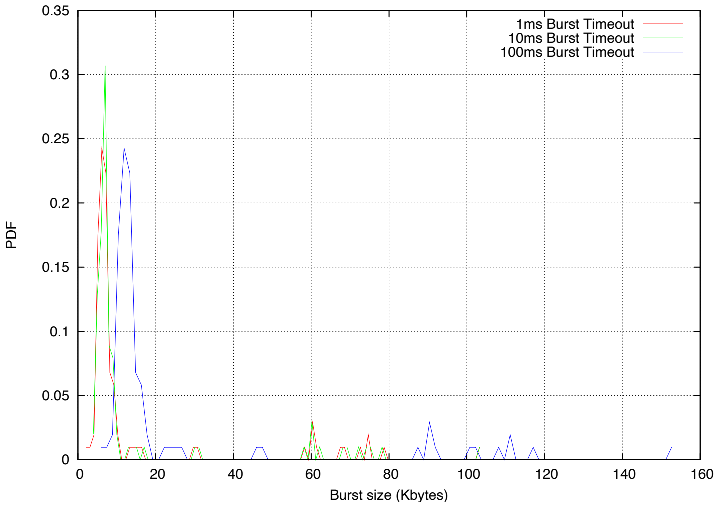


Figure 3.24 PDF of the burst size with assembly times: 1ms, 10ms and 100ms. Case: 10 video flows.

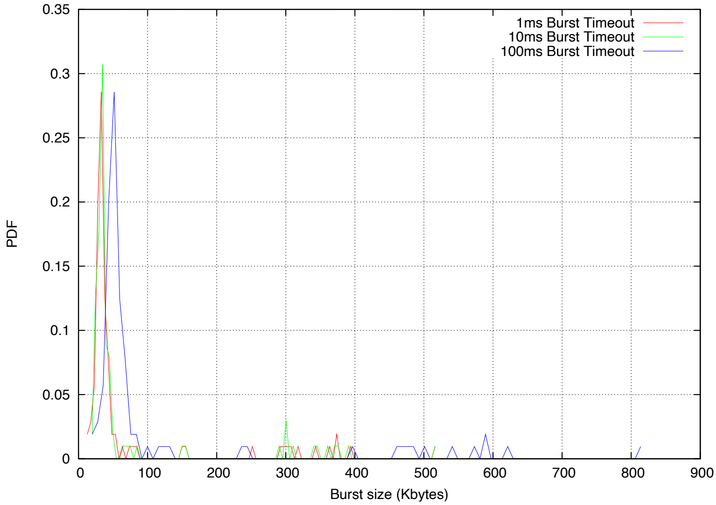


Figure 3.25 PDF of the burst size with assembly times: 1ms, 10ms and 100ms. Case: 50 video flows

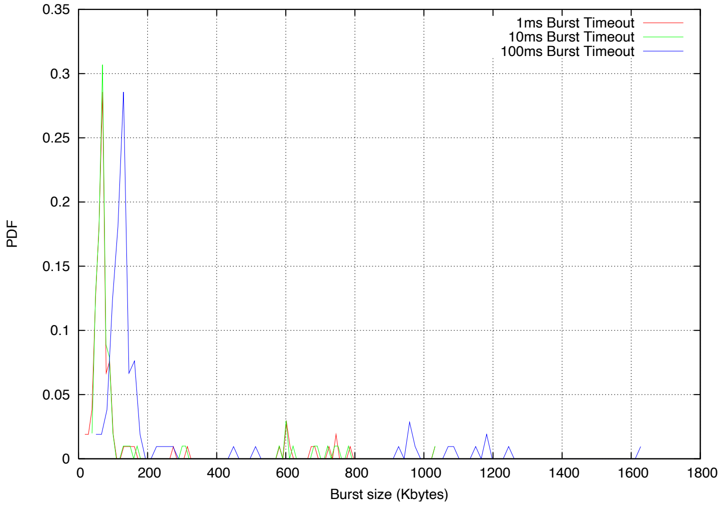


Figure 3.26 PDF of the burst size with assembly times: 1ms, 10ms and 100ms. Case: 100 video flows

Figure 3.24 to Figure 3.26 present the PDF of the burst sizes for different number of video flows. Each figure shows results with three timeouts. In the three figures it is observed that the burst sizes reached are spread over a broad range. The results indicate that the I frames cause the transmission of the large burst. The number of large burst is less than the number of small burst. But in the case of video flows the large bursts carry the information of the important I frames needed to decode correctly the received video.

In this section, it was presented a study of video transmission over OBS networks. Both, the length-based assembly algorithm and the time-based burst assembly algorithm were evaluated to find out its optimal configuration for digital video transmissions. The basic operation of the two assembly strategies was presented.

The evaluation has been made through event driven simulation, using the ns2 implementing an OBS module. The complete simulation environment to evaluate video traffic transmissions over OBS networks has been set up adding capacity to use video traces as a traffic source and using some software tools to assess the video quality at receiver side.

With the evaluation, it has been found that there is a drawback using length-based burst assembly algorithms for video traffic transmissions. To guarantee high quality videos at reception is needed large burst size thresholds, but at the same time increasing the end-to-end delay. Using a length-based burst assembly algorithm in conditions of variable traffic load, it is difficult to establish a priori an optimal burst size that must be set up. So, the basic operation of the algorithm could be modified to smooth the effects in the extra delay. The burst size threshold must be configurable on-line through measurements of the video traffic load to take into account the difference of the optimal burst size for a few video flows to a several video flows.

Additionally, it is necessary to implement mechanisms to detect when the video transmission is near to finalized. Thus, a timer could expire and the burst would be ready to send before reach the maximum burst size. Implementing such mechanisms could minimize the extra delay for the final video frames. Such assembly algorithm should be implementing in a video traffic assembly queue.

In the case of time-based assembly scheme, it has been showed that the delay requirements are fulfilled and quality video transmissions could be obtained. It is due to the flexible burst size that could carry UDP packets from frames with variable size. The drawback here is the variable burst sizes that are generated. This variability could cause problems when a QoS mechanism is implementing at the core, e.g., scheduling differentiation mechanism.

These experiments presented in this subjetction, about burst assembly algorithms, have been an initial study of UDP video streaming over OBS networks. Making possible to show some performance metrics to evaluate video transmissions; such metrics that could be taking into account in the design of burst assembly strategies. Video streaming using hybrid assembly techniques, burst assembly schemes proposed for improve TCP throughput [57], as well as scheduling algorithms could be studied. It is important to evaluate those algorithms in order define the OBS architecture and protocols suitable for multimedia and at the same time for non real-time data traffic to the all optical internet backbone. So that, it will be possible take advantage of the high capacity of the OBS networks. In the next section is presented a burst assembly proposal that take into account results and conclusions of this research stage.

3.4. Adaptive burst assembly algorithm to improve video QoE

The transport of conventional video streaming flows over OBS networks is subject to potential edge node congestion, according of findings presented in the previous section. This phenomenon could cause a severe video quality degradation, especially if static burst assembly schemes is in place. In this subsection is proposed an adaptive burst assembly algorithm inspired in the TCP-friendly strategy. The proposal modifies a burst threshold taking into account UDP packet drop rate at the burst assembly queue in the ingress node. Simulation results show that the proposed technique effectively reduces video frame loss and helps to maintain constant the video quality with variable traffic video flows.

User Datagram Protocol (UDP) is a protocol generally used in streaming media. Real-time video and audio streaming protocols are designed to handle occasional packet losses. Through the years, UDP traffic has been increasing since Internet broadband access has been gaining a lot of subscribers demanding broadband services such as video on demand, video conference, audio and video podcasts, on-line gaming, and other media applications increasing the data traffic and UDP traffic. On the other hand, the huge data traffic in Internet is being carried by optical core networks and new switching proposals have been investigated in the last years in order to build all optical networks, as mentioned in previous sections.

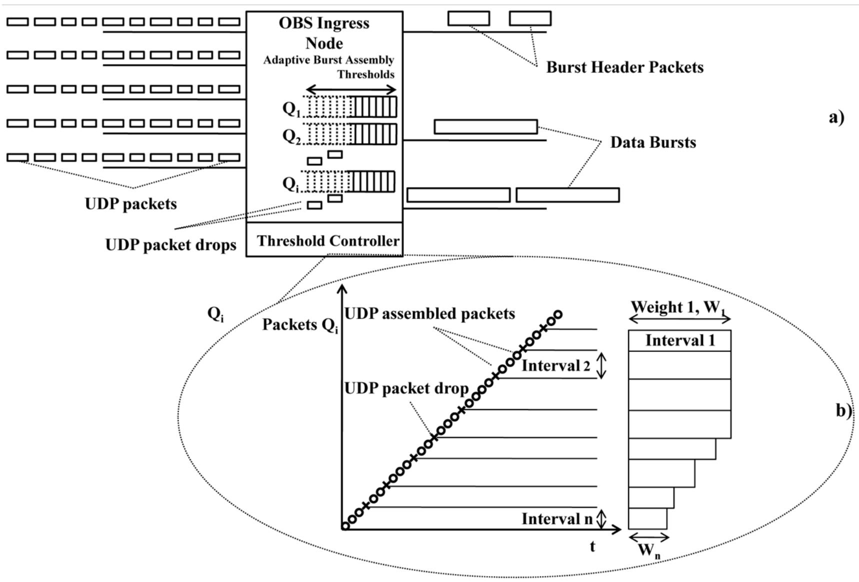


Figure 3.27 Adaptive burst assembly algorithm at ingress node

As previously mentioned, an OBS network is formed by edge nodes interconnected by core nodes through Weavelength-Division Multiplexed (WDM) optical links. In the ingress nodes of the OBS network packets are buffered in assembly queues and then packed into a burst based in a burst assembly criterion. Figure 3.27 (a) shows the assembly process in the ingress node. The bursts generated are optically switched by the core nodes to reach the egress node following a path previously reserved by a Burst Header Packet (BHP) previously sent over a common signalling channel at an offset time in advance. The burst size or the burst time depends on the burst assembly algorithm used in the ingress node.

Different burst assembly techniques could be implemented in the ingress node: length-based, length-based and hybrid schemes. Using a length-based burst assembly technique, when a constant burst size threshold is reached, the incoming packets buffered in the assembly queue are assembled into a burst that becomes eligible for transmission. Then, the burst is forwarded to an output buffer and the BHP is sent to make the reservation in the down link for the corresponding burst. In the time-based aggregation strategy, the burst formation time is constant and it is established beforehand. When a packet arrives at a burst assembly queue, a timer is started. Then, incoming packets are buffered in the burst assembly queue until the timeout is reached. Then, the burst is created with the content of the burst assembly queue, and the burst is scheduled for transmission. The timer is reset and remains in this value until the next packet arrives to the queue. Hybrid schemes use time and size thresholds, where the burst is assembled when any of the thresholds is reached.

In OBS networks, there are two types of congestion: edge congestion and path congestion [60]. In the former, congestion occurs at ingress and/or egress edge nodes due to the regulation imposed by burst assembly and output scheduling, whereas in the latter, congestion occurs in the core due to contention of bursts coming from different ingress nodes. In this subsection, the focus lies on edge congestion, incoming packets, arriving in the ingress node to form data bursts arrive at a much higher arrival rate than the ingress node can deal with and eventually the assembly buffer overflows cause packet drops.

Most previous work deals with assembly strategies to improve the performance of TCP [61][62] and, conversely, on TCP congestion control mechanisms optimal for a given assembly mechanism. However, less studies have been made about transport of UDP real-time traffic, which would complete the picture. For its importance, the focus in this study lies on proposing mechanism to improve video traffic QoE. Here is proposed an adaptive burst assembly scheme that adjusts the burst size or time threshold. Making possible fewer packets drops at the edge node to guarantee high quality video transmission when variations in the number of active video flows occur.

In this section is presented a mechanism where the burst assembly unit at the ingress node adjusts the burst size threshold as a function of the *loss event rate*. A loss event consists of one or more UDP packets dropped in the burst assembly queue at the edge node within a period of time between the moment when the first packet that starts the assembly process arrives until the burst is assembled. Figure 3.27 (b) presents the adaptive assembly process at the ingress node. In the next subsections is explained the adaptive scheme proposed following by a simulation evaluation where a length-based and the adaptive burst assembly schemes are compared.

3.4.1. Adaptive Burst Assembly Algorithm - ABAA

The aim of the algorithm is establishing an optimal burst size threshold during the burstification process when the number of active video flows could increase or decrease. Thus, it generates bursts long enough to carry all the necessary UDP packets with video frame information, avoiding UDP packets drops at the burst assembly queue. In order to adapt dynamically the burst size threshold value, the algorithm takes into account UDP packet drops at the ingress node to decide if the burst size threshold value has to be increased or decreased.

The algorithm requires computing the *loss event rate*, *p*. This process is accomplished at the OBS ingress node. The *loss event rate* consists of several UDP packets that are dropped at the ingress node in the assembly process of each burst. As already discussed, packet drops are due to peaks of traffic when there are many active video flows, causing burst assembly queue overflow. The *loss event rate* is calculated based in the *Average Loss Interval method* [63] adapted to OBS.

In this context, a *Loss Interval* is defined as the number of UDP packets correctly assembled and transmitted for each burst generated between loss events, see Figure 3.27 (b). The *average loss interval method* computes a weighted average of the loss rate over the last n loss intervals, with equal weights on each of the most recent *n/2* intervals. So the *Average Loss Interval* $\hat{s}(1,n)$ could be calculated as follow:

$$\hat{s}(1,n) = \frac{\sum_{1=i}^{n} w_i s_i}{\sum_{1=i}^{n} w_i}$$ **Equation 3.1**

for weights w_i:

$$w_i = 1, \qquad 1 \leq i \leq n/2$$

and

$$w_i = 1 - \frac{1 - \frac{n}{2}}{\frac{n}{2} + 1}, \qquad n/2 \leq i \leq n$$

and in the case of ABAA algorithm for an OBS network s_i is defined as:

$$s_i = \frac{x_i}{x_i + y_i}$$

where x_i is the number of UDP packets dropped and y_i represents the number of UDP packets sent in a burst.

In [63] is shown that n = 8 present a balance between resilience to noise and a fast response to changes in the network conditions. Once the *Average Loss Interval* is calculated, the loss event rate could be obtained as follow:

$$p = 1 - \hat{s}(1,n), \qquad 0 \leq \hat{s}(1,n) \leq 1$$ **Equation 3.2**

The *loss event rate* is the parameter that controls the burst size threshold. If the *loss event rate increase*, the burst size threshold has to be incremented, otherwise if the *loss event rate* decreases the burst size threshold has to be decreased. The idea is to have a trade-off between an optimal burst size and the demanding traffic that need to be assembly into burst, with a bounded delay.

3.4.2. Experimental evaluation

The evaluation of video streaming over OBS networks has been made first using the static length-based burst assembly algorithm and then the proposed adaptive algorithm in order to assess the two algorithms and find the advantages of using the ABAA algorithm for video transport. Through simulations with a static length-based burst assembly algorithm, it has been found that the quality of the received videos gets worse when the number of active video flows increase and at the same time the burst size threshold remains static. Figure 3.28 shows video frame losses, as it could be observed frame loss grows when video traffic grows if the burst size threshold value is constant, for instance 100 Kbytes. Also, the Figure 3.28 shows that there is a threshold value at which, depending on the number of active video flows, higher frame losses can be avoided. These valid burst size thresholds are 80, 280 and 580 Kbytes for 10, 50 and 100 video streams, so it is necessary that the burst size threshold be adaptive. However, as shown in the Figure 3.29, there is a trade-off between the burst size threshold and the number of video flows to have a constant frame delay and a lower frame loss. If it is used as optimal burst size the thresholds presented in the Figure 3.28, 80, 280 and 580 Kbytes, the frame losses are lower and it is possible to guarantee that 70 % of the frames have 500 ms average delay in each case of number of video flows.

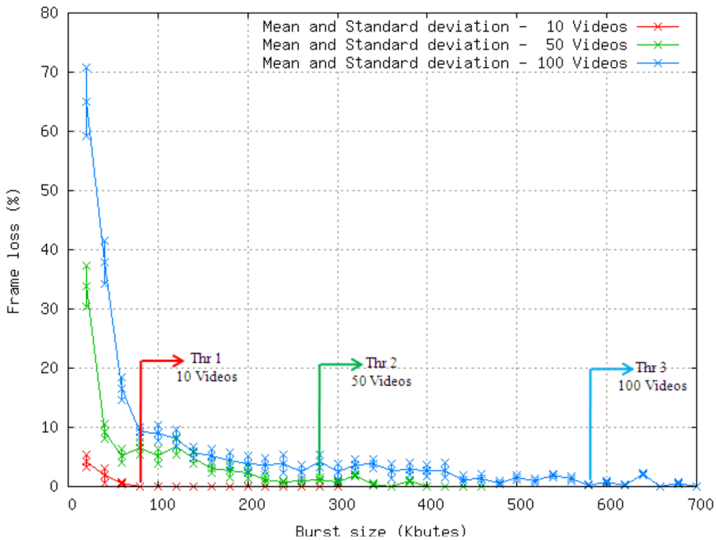


Figure 3.28 Average frame loss using static length-based assembly for different cases of number of video flows and burst size threshold values

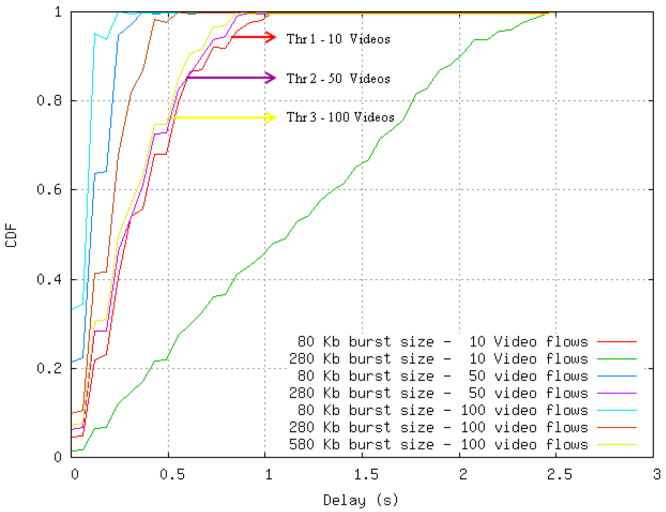


Figure 3.29 Static length-based assembly frame delay CDF 10, 50, 100 video flows.

Using optimal burst size thresholds, 80, 280 and 580 Kbytes it has been done the evaluation using the ABAA algorithm. The optimal burst size threshold is used by ABAA algorithm to set up the burst size threshold value when is necessary to increase or decrease de burst size in response to changes in the loss event rate.

The simulation shows that the ABAA algorithm outperforms burst-size-triggered assembly in terms of video quality of the received video. Figure 3.30 presents the Peak Signal to Noise Ratio (PSNR) as a measure of video quality. These valid burst size thresholds are 80, 280 and 580 Kbytes for 10, 50 and 100 video streams, so it is necessary that the burst size threshold be adaptive. However, as shown in the Figure 3.29, there is a trade-off between the burst size threshold and the number of video flows to have a constant frame delay and a lower frame loss. If it is used as optimal burst size the thresholds presented in the Figure 3.28, 80, 280 and 580 Kbytes, the frame losses are lower and it is possible to guarantee that 70 % of the frames have 500 ms average delay in each case of number of video flows.

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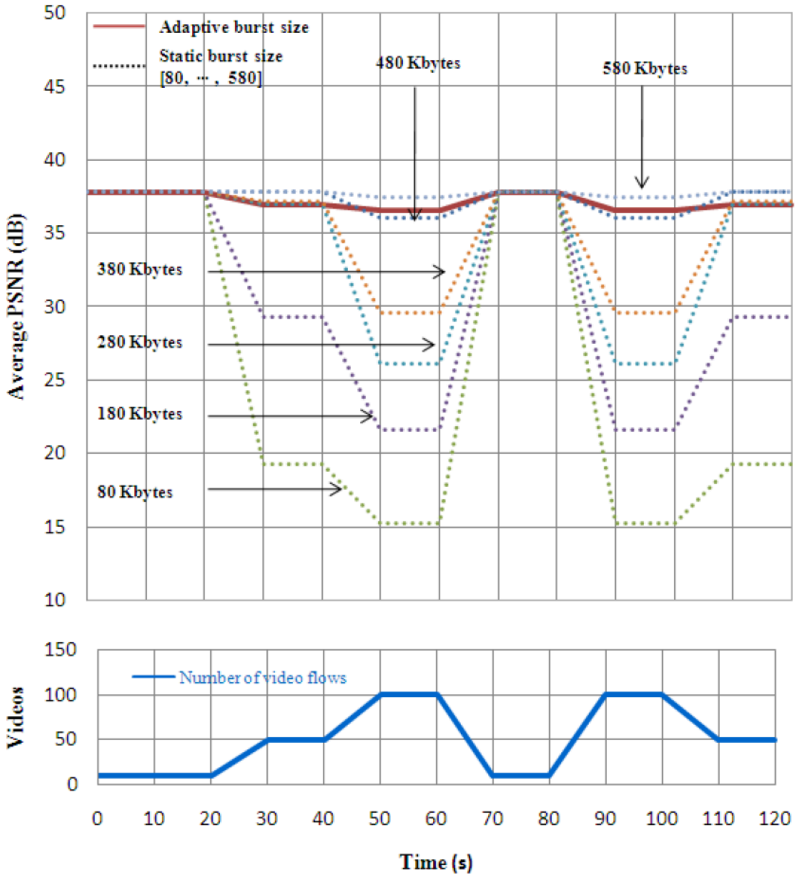


Figure 3.30 Average PSNR for different burst size thresholds and Average PSNR using ABAA

Figure 3.30 compares the two algorithms evaluated. As it is presented the static length-based burst assembly algorithm causes lower PSNR value when the number of video flows increases or decrease. Good quality videos are possible with a static 580 Kbytes burst size threshold, but the frame delay is unacceptable when there are few video flows. On the other hand, ABAA maintain the video quality (PSNR value) constant, when the number of video flows varies, adjusting the burst size threshold and at the same time the frame delay is guaranteed.

Adaptive burst assembly algorithm (ABAA) is a technique that pays attention to the loss event rate of UDP packets at the ingress node to dynamically set up the burst size threshold to maintain the video quality while keeping the end-to-end delay within a desirable bound. This technique improves the static nature of the burst sizes in the length-based burst assembly technique. The algorithm controls the burst size transmitted through the OBS core network generating burst sizes with the same length for those periods of time where the number of video flows remains constant making easy the implementation of the scheduling algorithm in the core nodes and making possible the implementation of end-to-end QoS differentiation mechanism for video streaming, as well as maintaining a delimited delay.

3.5. Scheduling scheme to provide QoE to video traffic

Offer QoE to final video transmissions customer through QoS implementation in the OBS core transport network has to do with the different efforts done in each building part of the OBS network. Above in section 3.4, an adaptive burst assembly scheme at edge has been presented to provide QoS at the ingress node, below in this section is presented a contribution to offer QoS focusing on the scheduling process at the OBS core nodes.

The methodology and set-up to evaluate the video transmission over OBS networks explained in subsection 3.2.3 has been implemented in this part of the study, adding simulation modules for internet traffic and the scheduling algorithm. This part of work has been carried out by the support of a collaboration strategy under the European Network of Excellence BONE: Building the Optical Network in Europe in a mobility action to the Research Academic Computer Technology Institute University of Patras, Greece [66].

In OBS-related bibliography, LAUC-VF is the scheduling algorithm of choice for OBS networks, due to its good average case performance and relatively small computational complexity, but it does not support QoS differentiation as explained in section 3.1.4. QoS provisioning in one-way OBS is a challenging problem, due to lack of buffering at the core. LAUC-VF has no information about future bursts, so inevitably makes non-optimal decisions regarding channel selection and burst dropping [65]. Two very effective techniques proposed in bibliography for strict priorities in OBS networks are PJET and PLAUC-VF [36]. PJET implements strict priorities by differentiating the offset time of priority classes, but with a big delay overhead. PLAUC-VF uses burst preemption, which is a technique that adds flexibility. It allows the re-arrangement of already scheduled bursts to accommodate reservation requests which would be otherwise blocked, and at the same time it makes QoS differentiation possible. In what follows in this section, it is explained a multi-class preemptive scheduling scheme that supports QoS differentiation.

3.5.1. QoS preemtive scheduling scheme

Research on preemptive scheduling algorithms has focused on multi-class schemes with strict priorities in order to achieve QoS guarantees and class separation. In the proposed scheme, it does not enforce strict priorities between classes, but rather employ a probabilistic preemption policy to decide whether to preempt a burst or not. It is used a variation of the *Preemption Drop Policy* [67] (PDP) to build a flexible preemption scheme, where the relative burst size is considered in the preemption decision, as well as their priority class. In an OBS node employing PDP, a free wavelength can be reserved by a burst belonging to any service class but if all wavelengths are occupied, preemptions can be performed, according to a predefined policy. The preemption policy is parameterized by the definition of *P*, which expresses the probability of a successful preemption. That is, whenever a reservation request for a new burst arrives, this new burst has a probability *P* to preempt an already scheduled overlapping burst.

Preemption probability (P) as defined in the proposed preemption policy depends on the relative significance of two bursts competing for the same resources. The significance of a burst depends on its length (a large burst has a more significant impact on throughput than a smaller one when dropped) and the priority of the service class it belongs to. So, the probability of a high priority (class-1) burst preempting a low priority (class-0) burst depends partly on their relative burst size, and partly on a constant preemption probability, P_0. P_0 parameter leads to class separation, and provides performance guarantees to high priority bursts, regardless of their size. It must be added here that, a class-0 burst never preempts a class 1 burst, whereas between two bursts of the same service class the smaller one is always preempted. The definition of *P*, given that B_{new}, CoS_{B_new} are the size and class identifier of the incoming burst respectively and B_{old}, CoS_{Bold} is the size and class identifier of an already scheduled overlapping burst. Preemption probability *P* is defined as follows:

$$P = \left\{ \begin{array}{cc} P_0 + (1 - P_0).\dfrac{B_{new}}{B_{old}}, & CoS_{B_{new}} > CoS_{B_{old}} \\ 1, & B_{new} > B_{old} \\ 0, & B_{new} < B_{old} \end{array} \right\} \quad \textbf{Equation 3.3}$$

3.5.2. Preemptive LAUC-VF based channel scheduling

Here is presented the proposed variation of LAUC-VF, termed as PLAUC-VF-ext, that employs preemptions based on the proposed *preemption drop policy* aforementioned. In first place, the LAUC-VF operation is presented.

In LAUC-VF, the scheduler keeps track of the free periods in each channel. To keep computational complexity low, the scheduler only stores two time intervals: a) the semi-unbounded interval *(sched_end,∞)* as well as b) the void period *(void_start,void_end)*, between the two most recent burst reservations. Thus, LAUC-VF only stores three time values per channel, and it only makes two comparisons to decide whether a new burst can be scheduled in a channel or not. Figure 3.31 (a) illustrates the principle of operation of LAUC-FV scheduling algorithm.

If a reservation request can be served by more than one channel, then the channel that minimizes the remaining idle period is chosen. When a new reservation request arrives, the scheduling unit performs the following three steps:

- It scans all channels for a free interval to schedule the forthcoming data burst, either into the void or after *sched_end* time value.

- It finds the "best fit" if multiple channels were found (where "best fit" is considered the option that minimizes the remaining idle period) or reports failure if none was found.

- It updates the void_start, void_end and sched_end values of the selected channel, if one was found, and reports success

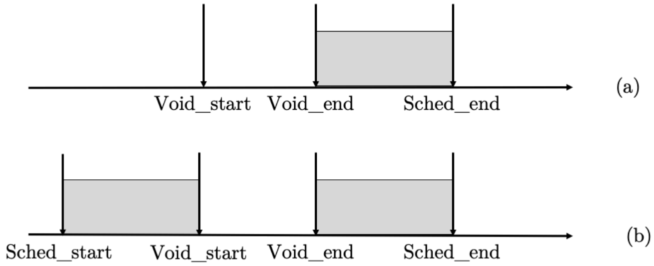


Figure 3.31 (a) LAUC-FV and (b) PLAUC-VF-ext operation principle

In PLAUC-VF-ext the state information stored per channel includes additional variables plus the variables used in the classic LAUC-VF: burst length, class of service, unique burst identifier for scheduled bursts, as well as, three time intervals, *Sched_start*, the semi-unbounded interval *(sched_end, ∞)* and the void period *(void_start,void_end)*, between the two most recent burst reservations (refer to Figure 3.31 (b)). Making the use of *Sched_start*, gives the scheduler the ability to keep track of and preempt the most recently scheduled bursts. For each burst reservation the class of service and the burst identifier has to be stored, while burst sizes are derived from the time values. The channel selection is made, using the time values *void_start*, *void_end* and *sched_end*.

When a new reservation request arrives and the channel selection fail for overlapping reservations the preemtion phase starts and the preemtion decisions in the proposed implementation PLAUC-VF-ext are based on the probabilistic preemptive scheme detailed in subseccion 3.5.1 above.

The preemption-capable scheduling unit follows these steps:

- It scans all channels for an idle period to schedule the burst. If voids are found in more than one channel, the one that minimizes the remaining idle period is chosen, as in LAUC-VF.

- If no voids are found, there is at least one overlapping burst in each channel. Then, the scheduling algorithm iterates over overlapping bursts, and decides whether one of them has to be preempted, in order to free resources for the newly arrived burst. The decision is based on the *Preemption Drop Policy* Equation 3.3.

- If a burst was preempted in step 2, as an optional step, the scheduler can attempt to reschedule the preempted burst at another unused channel.

- If more than one channel is available for rescheduling, the one that minimizes the remaining idle period is chosen. If the rescheduling step fails, the preempted burst is dropped.

Figure 3.32 supports the next example of operational process of PLAUC-VF-ext proposed above. If all bursts in Figure 3.32 belong to the same class of service, burst 0 would be preempted, since it is the smallest among all overlapping bursts and the reservation request, and the new burst would be scheduled in its place. Then the preempted burst 0 can be rescheduled to data channel 1.

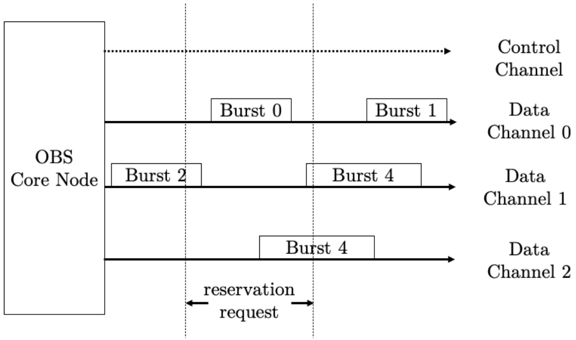


Figure 3.32 PLAUC-VF operation example

Next, it is provided a description of the PLAUC-VF-ext. The function *search_channel(i)* searches for an idle period in channel *i* to schedule the newly arrived burst. The void found or the overlapping burst(s) are stored in the arrays *void[]* and *ovburst[]* respectively. Then, the function *find_best_fit* scans the voids (if more than one was found) and returns the one, which minimizes the remaining idle period or an empty set if none was found. If no void was found, *preempt()* function scans all overlapping bursts in *ovburst[]* array and based on the preemption policy may select one of them to be preempted, so that the newly arrived burst can be scheduled. If an overlapping burst is preempted, the *find_best_fit* function rescans the *void[]* array to reschedule the preempted burst (pburst), or it has to be dropped. Figure 3.33 summaries the pseudo code for the proposed PLAUC-VF-ext scheduling algorithm.

```
for i=0 to num_channels
    [void[i], ovburst[i]] =
search_channel(i)
end i
result = find_best_fit(void, newburst);
if result != empty
    return result;
else
    [result, pburst] = preempt(ovburst, P)
    if result != empty
            find_best_fit(void, pburst)
            return result;
    end if
end if
return failure
```

Figure 3.33 PLAUC-VF-ext pseudo code

3.5.3. Experimental analysis

The preemptive scheduling algorithm has been evaluated through experimentation, modeling an OBS network over the 14-node NSF network topology. The modeled OBS network supports two classes of service, one best effort class for data transmissions and one real time class for video transmissions. The video traffic is generated through trace-driven traffic sources, therefore the input video trace file is produced by the video traffic generator module after parsing the reference MPEG4 video file. Low priority internet traffic is modeled with supFRP [68] traffic sources, which generate self-similar traffic, using superposition of fractal renewal processes. In every OBS edge node, it is employed one supFRP traffic generator with a *Hurst* parameter of 0.75, and multiple video streams with random destinations. The burst assembly process is performed in the OBS network's ingress nodes, using a timer-based aggregation algorithm. Packets from different classes arriving at the ingress node are allocated in distinct burst assembly queues and are never mixed in the same burst. In the simulations, it is considered QoS and QoE performance metrics, which are specific to each priority class. For low priority traffic, the performance metric considered is packet loss ratio. For video traffic (class-1), it is considered perceived video quality metric MOS, which accurately describes the impact of burst losses on video streams.

Figure 3.34 shows a comparation of LAUC-VF, PLAUC-VF and PLAUC-VF-ext. The proposed PLAUC-VF-ext in addition to preemption supports preempted bursts re-scheduling onto an available channel (if one is found). Since LAUC-VF does not support multiple priority classes, P_0 is set to 0, and therefore preemptions are only based on burst size criterion. From Figure 3.34, it can be seen that the preemption of small data bursts by larger ones yields a smaller blocking probability, see PLAUC-VF curve. The extra complexity of re-scheduling preempted bursts results in a further improvement of the scheduling efficiency, see PLAUC-VF-ext curve.

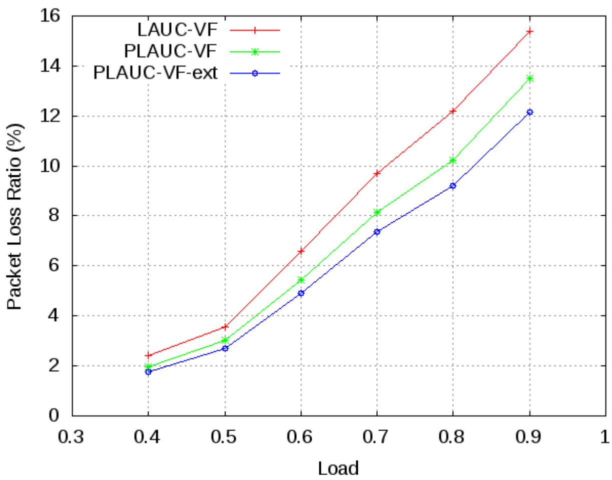


Figure 3.34 Packet loss ratio of LAUV-VF, PLAUC-VF, PLAUC-VF-ext, P_0 = 0

In what follows, it is studied, how P_0 selection affects throughput (class-0) and video quality (class-1). The results are displayed in Figure 3.35, Figure 3.36 and Figure 3.37. It can be clearly seen, when P_0 approaches 1 video quality is favoured against loss rate (and equivalently throughput). The opposite occurs, when P_0 tends to 0. To this end, it could be argued that the proposed scheduling scheme can actually support QoS differentiation over different classes of services, in a very flexible and efficient way. This is done with a simple parameter, P_0, which expresses the guaranteed preemption probability and effectively controls class separation.

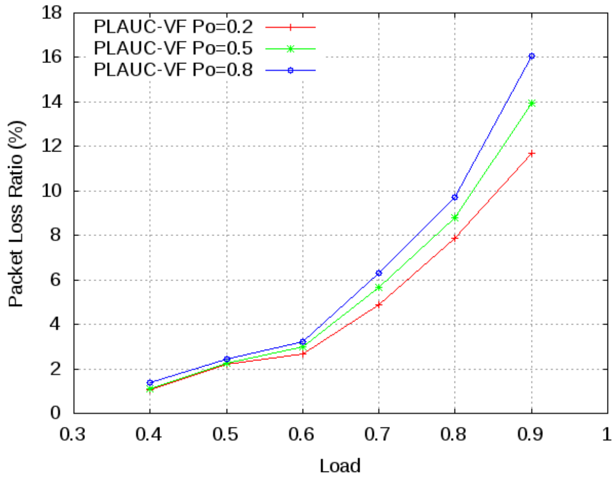


Figure 3.35 Low priority class bursts packet loss ratio, PLAUC-VF-ext

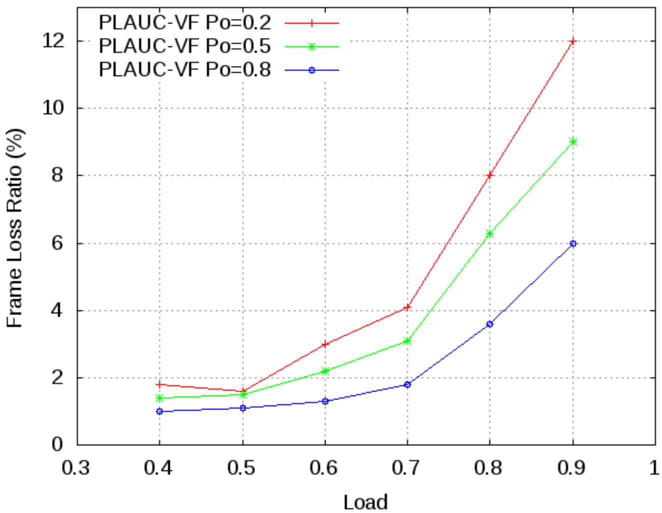


Figure 3.36 Video frame loss ratio for PLAUC-VF-ext.

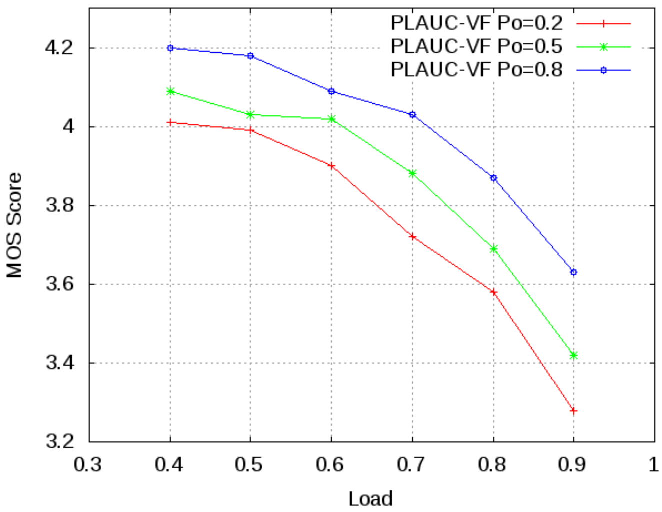


Figure 3.37 Video quality (MOS) for PLAUC-VF-ext.

In this subsection, a multi-class preemptive scheduling scheme was presented. It was demonstrated that proposed scheduling scheme can actually provide QoS differentiation in different service classes, in a very flexible and efficient way, parameterized by a simple parameter, which expresses guaranteed preemption probability. Proposed scheduling scheme clearly performs better than LAUC-VF with a slightly increased complexity.

3.6. Chapter summary

This chapter has presented the main theory about optical burst switching network paradigm reviewing aspects like architecture, protocols, algorithms and QoS. Then the QoE & QoS video over OBS evaluation setup has been explained as a proposal to assess video traffic and to make analysis and proposals for QoS in OBS, the evaluation setup included the video traffic traces, simulation process and results as a framework and methodology. Then a study about video traffic over OBS regard burst assembly algorithms has been introduced as a manner to apply the video over OBS evaluation proposal to finally present two specific contributions for video QoS in OBS at burst assembly and burst scheduling process.

Chapter 4

Video streaming traffic patterns to Energy Efficient Ethernet analysis

Activities that have been developed at the first stages of this work involves experiments regard video streaming and simulation and testbeds, capturing data and obtaining results for analysis and contribution, first applying to studies about video transmissions over OBS networks, specifically evaluating burst assembly algorithms and scheduling algorithms, assessing video traffic quality and contributing on QoS alternatives for video over OBS networks. Since the experiments have been done implementing testbeds using real video streaming scenarios and a methodology proposed and presented in subsection 3.2.3, this work continue with a characterization of Internet Protocol Television (IPTV) video streaming traffic and Dynamic Adaptive Streaming Over HTTP (DASH) traffic, both analysing at application level and ethernet level obtaining its corresponding traffic patterns. Therefore, video streaming over managed and unmanaged IP networks has been considered using RTP/UDP (IPTV) and HTTP/TCP (DASH) transport protocols respectively. The resulting traffic patterns have been used to the study and analysis of energy efficiency and energy savings at Network Interface Cards (NICs) involved in the video transmissions when the IEEE Energy Efficient Ethernet (EEE) standard is implemented at NICs of video servers and clients.

In this chapter, in first place, the video traffic pattern analysis in IPTV networks is presented as well as the explanation of the testbed implemented to this end, followed by the EEE standard and the energy efficiency concept definition. Then, this first part continues presenting the corresponding EEE energy efficiency analysis and energy savings calculation on NICs in video streaming servers. This study about EEE and IPTV video streaming traffic is presented in sections 4.1 and 4.2. In second place, in this chapter, the DASH traffic pattern is characterized and analysed by means of experiments on a proposed testbed for video streaming over unmanaged networks, leading to the evaluation of the corresponding EEE energy efficiency and energy savings analyses to this case of DASH streaming, sections 4.3 and 4.4 presents this final results.

4.1. Video Traffic profile in IPTV networks

In this section the IPTV video traffic profile characterization carried out in this work is presented. The video streaming over IPTV networks and codification standards basic concepts have been explained in chapter 2 subsection 2.1.7. Next the video streaming testbed implemented and used to evaluate the IPTV video traffic profile is explained. The IPTV video traffic profile obtained considers the network interface card output of a video streaming server, that it is RTP/UDP transport segments over IP/Ethernet Frames carry on the H.264/AVC video content information [69].

4.1.1. IPTV testbed

In order to run the experiments to obtain the IPTV traffic profile an IPTV testbed has been implemented. Figure 4.1 shows the building blocks of the IPTV streaming service testbed that makes possible to generate IPTV video traces and characterize the traffic pattern at the NIC card of a video streaming server. Here the video traffic patterns that describe the video traffic at application layer and Ethernet layer have been obtained.

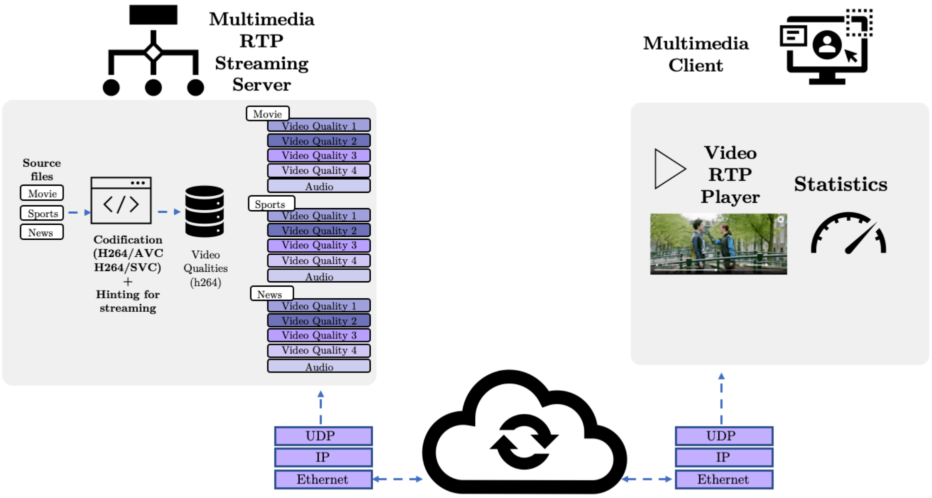


Figure 4.1 IPTV streaming service testbed

As could be seen in the Figure 4.1 the multimedia server host implemented has the following components: (1) the video content source files, (2) a video coding and hinting process for streaming over the network, and (3) a video server entity that encapsulate the video information messages in UPD transport segments to be send over IP end-to-end and over the Ethernet frames at physical link.

As mentioned before, the video traffic profile in IPTV network deployment is evaluated at the Ethernet network interface of a video streaming RTP server, then video traces show its relationship to the Ethernet frames resulting of the video transmission that are needed to the EEE Energy Efficiency study.

In the next subsection the traffic patterns obtained are presented, for the experiments it is used different content types of videos which are news, sports and movie content types. The IPTV traffic patterns are obtained for the H.264/AVC and H.264/SVC codification standards by implementing the codec process and running a series of video streaming sessions. Next, the characteristics of the encoded video in IPTV is presented followed by the resulting IPTV traffic patterns at application and ethernet levels.

4.1.2. H.264/AVC traffic pattern

In IPTV network as video distribution alternative, the service provider often uses MPEG-4 or H.264/AVC codecs [69] at application layer, to either generate fixed or variable quality (*fixed-Q* or *variable-Q*) for standard or high definition video content from a *raw video* source, that is a video file containing uncompressed data from the digital camera sensor. The *fixed-Q* case aims at maintaining the quantization level at the expense of producing more information at codec output when the video sequence contains several moving objects. On the contrary, the *variable-Q* case produces an almost constant bit rate at codec output, however at the expense of having quality degradation at scenes with a large number of moving objects. The codified video is then packetized and encapsulated through UDP/IP/Ethernet to be delivered from the multimedia RTP server to the end user through the network.

Figure 4.2 shows the video encoding and frame packetization process followed in this work and used in the testbed described in Figure 4.1. Essentially, as explained in subsection 2.1.1, the codec generates 12 video frames *Group of Pictures* (*GoP*) every 480 ms, where video frames can be of type *I* (*Intra*), *P* (*Predictive*) or *B* (*Bidirectional*) [70]. Typical examples of *GoPs* are *IPPPPPPPPPPP* and *IBBPBBPBBPBB* as described in 2.1.1.

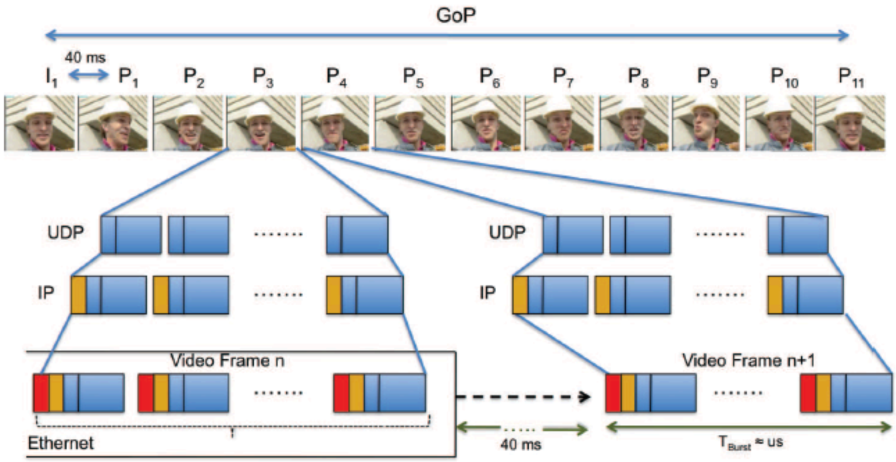


Figure 4.2 Top-down video frame packetization

As the *I* video frames are usually larger than the *P* and *B* video frames, therefore they produce larger Bursts of Ethernet frames, E(B) to transmit, that could be evidenced comparing $E(B_I)$, $E(B_P)$ and $E(B_B)$ in Table 4.1 and Table 4.2 . Additionally, because there are 11 times more *P/B* video frames than *I* video frames, the average burst size $E(B)$ is usually closer to $E(B_P)$ and $E(B_B)$ than to $E(B_I)$.

Table 4.1 Video traffic pattern for fixed bitrate and variable-Q (up) and variable bitrate and fixed-Q (down).

Trace	Resolution (pixels)	Avg. Bitrate (Mbps)	$E(B_I)$ (Frames)	$E(B_P)$ (Frames)	E(B) (Frames)
Variable-Q – GoP structure IPPPPPPPPPPP (without B video frames)					
Movie	640x480	2.41	27.69	7.38	9.28
Movie	720x480	3.91	48.87	11.44	14.74
Movie	1280x720	10.18	93.48	32.30	37.63
Movie	1920x1080	14.36	110.07	47.46	52.90
Sport	640x480	1.69	14.78	5.82	6.65
Sport	720x480	3.91	33.40	12.89	14.75
Sport	1280x720	10.22	68.77	34.66	37.76
Sport	1920x1080	13.45	74.41	47.07	49.56
News	640x480	2.07	13.09	7.00	8.01
News	720x480	3.67	28.77	11.49	13.86
News	1280x720	9.36	48.25	32.28	34.62
News	1920x1080	12.20	54.74	43.28	44.97
Trace	**Codec Quantizer**	**Avg. Bitrate (Mbps)**	**$E(B_I)$ (Frames)**	**$E(B_P)$ (Frames)**	**E(B) (Frames)**
Fixed-Q – GoP structure IPPPPPPPPPPP (without B video frames)					
Movie	31	2.88	24.78	9.53	10.98
Movie	16	3.99	36.75	12.93	15.02
Movie	8	5.52	49.37	17.63	20.62
Movie	2	14.39	101.25	46.99	52.96
Sport	31	1.68	13.10	5.97	6.62
Sport	16	2.54	21.12	8.71	9.75
Sport	8	4.00	29.47	13.61	15.07
Sport	2	13.45	74.41	47.07	49.56
News	31	2.14	12.26	7.50	8.28
News	16	3.87	29.47	13.19	14.61
News	8	4.35	22.28	15.16	16.33
News	2	4.96	21.55	17.98	18.59

Table 4.1 summarizes the main features of three different video traces (Movie, Sport and News) of about two minutes of duration encoded with the H.264 standard and *GoP* structure *IPPPPPPPPPPP*. The upper part of the table considers *variable-Q* and fixed resolution with nearly constant bitrate: 640x480(2 Mbps), 720x480(4 Mbps), 1280x720(10 Mbps) and 1920x1080(14 Mbps) [71]. The lower part of the table considers the same videos codified with *fixed-Q*, level values of the codec quantizer used are 31, 16, 8 and 2 for each video fixed quality generated. In this case, the bitrates experience more variability than before with respect to the video-clip type. Basically, keeping a certain

resolution for a sport event requires a higher bitrate value than for a movie, and much more than for the news TV show. Table 4.2 shows trace statistics, the traffic pattern, for the same video clips and encoding settings, but with *GoP* structure *IBBPBBPBBPBB.*

Table 4.2 Video traffic pattern for fixed bitrate and variable-Q (Up) and variable bitrate and fixed-Q (Down).

Trace	Resolutions (pixels)	Avg. Bitrate (Mbps)	$E(B_I)$ (Frames)	$E(B_P)$ (Frames)	$E(B_B)$ (Frames)	E(B) (Frames)
Variable Q – GoP structure IBBPBBPBBPBB (with B video frames)						
Movie	640x480	2.24	27.84	9.46	5.44	8.65
Movie	720x480	3.92	46.48	17.89	8.44	14.80
Movie	1280x720	10.24	94.95	49.30	24.85	37.82
Movie	1920x1080	13.46	110.09	72.53	31.70	49.59
Sport	640x480	1.78	14.42	8.37	5.39	6.99
Sport	720x480	3.92	33.50	19.25	10.51	14.77
Sport	1280x720	10.03	69.74	55.85	25.60	37.09
Sport	1920x1080	10.68	72.06	61.21	26.85	39.46
News	640x480	1.84	13.54	6.37	5.69	7.20
News	720x480	3.66	27.59	16.23	9.55	13.86
News	1280x720	9.30	55.44	46.44	24.77	34.44
News	1920x1080	10.66	58.78	54.23	28.86	39.38
Fixed Q – GoP structure IBBPBBPBBPBB (with B video frames)						
Movie	31	2.40	25.06	11.87	5.83	9.23
Movie	16	3.44	36.26	16.33	8.49	13.02
Movie	8	5.28	50.97	23.29	13.91	19.74
Movie	2	9.01	73.31	38.72	25.35	33.36
Sport	31	1.78	12.96	8.44	5.60	6.99
Sport	16	2.63	20.65	12.27	7.83	10.06
Sport	8	4.21	29.46	19.89	12.42	15.83
Sport	2	7.26	46.49	34.28	21.58	26.96
News	31	1.88	12.72	7.41	5.69	7.34
News	16	3.00	24.33	13.19	8.16	11.42
News	8	4.28	22.62	18.62	13.27	16.11
News	2	7.19	34.81	31.75	22.65	26.74

For high-resolution videos, the *I* video frames comprise about 50-100 Ethernet Frames whereas the *P/B* video frames contain 20-40, depending on the encoding settings. Thus, every *I/P/B* video frame produces a burst of Ethernet Frames which are then sent back-to-back to the NIC.

In addition, it is worth noticing that the average burst size *E(B)* (in terms of Ethernet Frames) for *GoP IPPPPPPPPPPP* is slightly larger than for *GoP IBBPBBPBBPBB*. Hence, the use of *B* frames contributes to an increase in video compression, but has the disadvantage of a greater computational complexity than for *GoPs* without *B* frames. Concerning transmission, *B* video frames need to be temporally stored until the *I* or *P* video frame has been generated. For example, *GoP* $I_1B_2B_3P_4B_5B_6P_7B_8B_9P_{10}B_{11}B_{12}$ would be transmitted in the following order: $I_1P_4B_2B_3P_7B_5B_6P_{10}B_8B_9I_{13}B_{11}B_{12}$ since B_2 and B_3

depend on P_4, B_5 and B_6 depend on P_7, etc. Finally, since the video server must wait for the generation and codification of P_4 before encoding frames B_2 and B_3, it can be seen from the traces that the video server actually transmits the group $P_4B_2B_3$ back-to-back, thus generating a burst of about $E(B_P) + 2E(B_B)$ Ethernet frames. There is a relationship of bursty traffic patterns which have been found to favours EEE energy efficiency and savings that in next 4.2 section is explained.

4.1.3. H.264/SVC traffic pattern

Another case of video encoding settings has been considered. Such case uses H.264 *Scalable Video Coding* (aka H.264/SVC) that takes into account a different *GoP* structure built by *hierarchical B* video frames. Such hierarchical feature of H.264/SVC refers to the fact that *B* video frames can be used to predict other *B* frames, in contrast with the classical H.264/AVC B video frame prediction process where a *B* video frame depends on the previous and next *I/P* video frames. Essentially, the layer scalable H.264/SVC codec offers temporal, spatial and quality scalability to perform the decoding process using a single base layer or the base layer plus several enhancement layers [6], [7]. This allows to remove some *B* video frames without losing the ability to decode the video.

Table 4.3 Movie video traffic pattern encoded with H.264/SVC

Number of layers	Birate (Mbps)	$E(B_I)$ (Pkts)	$E(B_B)$ (Frames)	E(B) (Frames)	GoP Structure
Five	2.51	46.70	10.79	13.05	$I_0B_1B_2B_3B_4B_5B_6B_7B_8B_9B_{10}B_{11}B_{12}B_{13}B_{14}B_{15}$
Four	1.94	46.70	13.94	18.05	$I_0B_2B_4B_6B_8B_{10}B_{12}B_{14}$
Three	1.43	46.70	17.43	24.78	$I_0B_4B_8B_{12}$
Two	1.02	46.70	20.55	33.66	I_0B_8
Base	0.74	46.70	-	46.70	I_0

More specifically, the sample video Movie has been encoded using H.264 /SVC with temporal scalability and 16 *GoP* size with hierarchical *B* video frames. The traffic pattern (video trace) resulting, and its characteristics are showed in Table 4.3. The video has been encoded using the same software tool as in [6] implemented in the testbed described in Figure 4.1, the H.264 SVC Joint Scalable Video Model (JSVM) reference software. Several differences must be noted with respect to the H.264/AVC cases: Firstly, the *GoP* structure and size varies depending on the number of layers transmitted, going from $G_{16}B_{15}$ for the highest temporal scalability with all layers (5 in this case) down to G_1B_0 for the temporal base layer only, and so does the average *GoP* size.

Concerning transmission, given the hierarchical nature of the prediction of B frames, the encoded video is transmitted on a *per-GoP* basis, that is, all the frames of the same *GoP* are transmitted back-to-back every 680 ms. For instance, the five-layered Movie trace would be transmitted in the following order: $I_0I_{16}B_8B_4B_2B_1B_3B_6B_5B_7B_{12}B_{10}B_9B_{11}B_{14}B_{13}B_{15}I_{32}B_{24}B_{20}\ldots$ So, the whole *GoP* $I_{16}B_8B_4B_2B_1B_3B_6B_5B_7B_{12}B_{10}B_9B_{11}B_{14}B_{13}B_{15}$ would be transmitted at once, that would favour EEE energy efficiency and energy savings, something that in the next 4.2 section is studied.

4.2. Evaluation of Energy Efficient Ethernet in video streaming servers

The H.264/AVC and H.264/SVC traffic analysis carried out and its corresponding traffic patterns, presented in the previous section, have been useful to analyse the performance of Energy Efficient Ethernet on video streaming servers. That study is presented in this section after a brief explanation of the EEE standard, the EEE energy efficiency concept and previous findings that motivate this study about video traffic patterns and its relationship to motive the implementation of the EEE standard at NICs cards on streaming servers.

4.2.1. Overview of Energy Efficient Ethernet

As Internet grows its associate power consumption also grows, that situation has raised the attention to find ways to reduce the energy consumption improving the networks energy efficiency [72]. Applying energy efficiency policies to the Internet could reduce the overall consumption that represent tens of TWh. In this sense, to improve the energy savings specifically at Network Interface Cards (NICs), the IEEE 802.3az Energy Efficient Ethernet (EEE) standard has been defined. The objective of EEE is to save energy by setting the NIC in *low power mode* when it has not any ethernet frame to transmit or receive [73]. EEE could enable large energy savings because its widespread deployment being a dominant technology for local area networks with billions of devices deployed [74].

Regarding the effectiveness of activating the *low power mode* (*sleep*) in the NIC, the energy efficiency has been found to depend on the ethernet traffic pattern, both in studies in simulation [75] and physical power measurements [76]. Therefore, ethernet traffic patterns like those that shows transmissions of ethernet frames in bursts transmissions have been found to impact positively on the EEE energy efficiency because the time the NIC stays in *high power mode* (*active*) is large enough to compensate the energy consumption of putting into *sleep* and *waking up* the ethernet card [76] [77]. When several Ethernet frames are transmitted back-to-back, that burst of data shares the same active and sleep mode transition overhead resulting in larger energy savings and better energy efficiency [77].

As aforementioned, to reduce energy consumption at Ethernet NICs the IEEE 802.3az EEE standard has been proposed. The main function of the proposed standard is to make the network interface card commutes to *low power consumption mode*, aka *low power idle*, when there are not ethernet frames to transmit or receive. In this mode, the NIC reduces energy consumption to 10% of the *high power transmission mode*. Figure 4.3 depicts the operation and energy consumption of the EEE standard, with the abbreviations defined in Table 4.4.

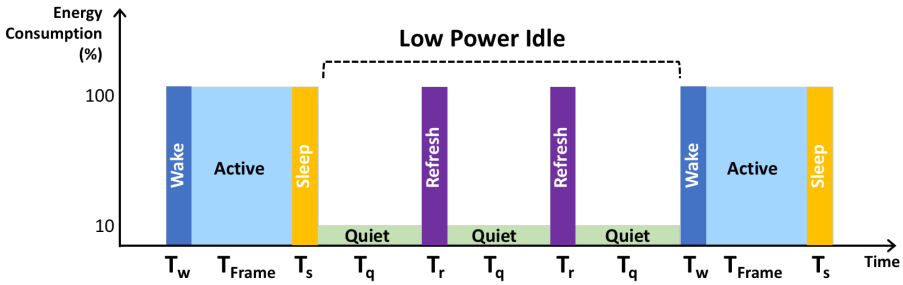


Figure 4.3 Energy consumption and time periods in high and low power modes in 802.3az

The standard defines the energy efficiency in terms of the ethernet interface card transmission rate and the ethernet frame size [73]. The ethernet frame efficiency η_{Frame} is defined by the relationship between the ethernet frame transmission time and the time needed to change from *low power mode* to *high power mode* and back, every time the card needs to transmit a frame. Equation 4.1 describes the ethernet frame efficiency, which relates the T_W *wake-up time*, T_{Frame} *frame transmission time* (active) and T_s *sleeping time.*

Table 4.4 Energy Efficient Ethernet time parameter definitions

Symbol	Name	Definition
T_w	Wake-up time	Time needed by the card to go to active mode from the low power idle mode, so the card is ready for transmission.
T_s	Sleep time	Time needed by the card to go from the active mode to the low power idle mode, so the card can start saving energy.
T_r	Refresh signal time	Time needed by the card to exchange signaling for synchronization between receivers, signal needed during idle times (low-power mode).
T_q	Maximum quiet perieod	Time where the card is in the low-power idle mode, no transmissions. A T_r is required when the low-power idle mode exceeds the T_q.
T_{Frame}	Frame transmission time	Time needed to transmit an Ethernet Frame.

$$\eta_{Frame} = \frac{T_{Frame}}{T_w + T_{Frame} + T_s} \qquad \textbf{Equation 4.1}$$

As a reference, in a 1000BASE-T ethernet link, the frame efficiency η_{Frame} is 0.6% for a frame transmission time of 1.2 µs with a frame size of 1500 Bytes, whereas a frame transmission time of 12 µs with a frame size of 150 Bytes results in a frame efficiency of 5.7%. Table 2 describes the frame transmission efficiency of frames with size 1500 Bytes and 150 Bytes for different link transmission speeds (100BASE-TX, 1000BASE-T and 10GBASE-T). Considering that EEE is not as efficient as it was expected because those wake-up and sleep-down times are too large with respect to the frame transmission time, the EEE standard propose the minimum Tw and Ts presented in Table 4.5.

Table 4.5 Minmum T_w, T_s and T_{Frame} for different link speeds in IEEE 802.3az

Protocol	Min T_w (μs)	Min T_s (μs)	T_{Frame} (1500B) (μs)	η_{Frame}	T_{Frame} (1500B) (μs)	η_{Frame}
100BASE-TX	30	200	120	34.3%	12	4.9%
1000BASE-T	16	182	12	5.7%	1.2	0.6%
10GBASE-T	4.48	2.88	1.2	14.0%	0.12	1.6%

Equation 4.1 entails that the fact of changing to low power mode is not efficient because the time needed to transmit a frame is short compared with sleeping and waking times, making the EEE standard impractical, as could be seen in Table 4.5. Therefore, other studies have concluded that transmitting a burst of ethernet frames in the high power mode rather than one frame results in better efficiency [76] [77]. Thus, the burst efficiency η_{Burst} is defined as in Equation 4.2, where T_{Burst} is the ethernet burst transmission time that depends on the aggregated number of frames.

$$\eta_{Burst} = \frac{T_{Burst}}{T_w + T_{Burst} + T_s} \qquad \textbf{Equation 4.2}$$

Here, $T_{Burst} = N * T_{Frame}$, typically $N \geq 10$

4.2.2. Energy efficiency transmitting a single H.264/AVC encoded video

Taking into account the bursty nature of the video traffic generated by a video streaming server presented above in subsection 4.1.2, it has been used a simulation model of Energy Efficient Ethernet [75] to evaluate the performance of EEE for each video trace (Movie, Sport and News with either variable or fixed Q and both GoP types) of As the *I* video frames are usually larger than the *P* and *B* video frames, therefore they produce larger Bursts of Ethernet frames, E(B) to transmit, that could be evidenced comparing $E(B_I)$, $E(B_P)$ and $E(B_B)$ in Table 4.1 and Table 4.2 . Additionally, because there are 11 times more *P/B* video frames than *I* video frames, the average burst size $E(B)$ is usually closer to $E(B_P)$ and $E(B_B)$ than to $E(B_I)$.

Table 4.1 and Table 4.2. Essentially, the video traces also provide both the frame arrival time and size values, necessary to estimate the amount of time that the Ethernet link spends in the sleep and active modes, as well as the transitions between them, for different link speeds: 100 Mbps, 1 Gbps and 10 Gbps. The simulation tool estimates the average efficiency value (η) in each scenario as:

$$\bar{\eta} = \frac{Total\ data\ transmission\ time}{Total\ time\ in\ acive\ mode\ and\ transitions} \qquad \textbf{Equation 4.3}$$

which, as in Equation 4.2 is near 100% when the number of T_w and T_s is small compared with the portion of time under actual data transmission. The efficiency value η are then obtained for the following configurations:

- **Poisson**: This is the efficiency figure obtained after simulation as if Ethernet Frames were transmitted following a Poisson basis, rather than back-to-back per video frame. This simulation has been performed assuming the same traffic load and frame sizes as in the original trace, but with exponentially distributed packet inter-arrival times with mean 1/λ, where:

  $$\lambda = \frac{\rho}{8E(P)/C}$$

  for a given load ρ obtained from the traces as:

  $$\rho = \frac{Bitrate}{C}$$

  It is important to remark that the average Ethernet frame size in the video traces is E(P) = 1370 bytes, and C refers to the link capacity (100 Mbps, 1 or 10 Gbps).

- **Real**: This is the efficiency value for the transmission of the video trace over EEE as specified in the IEEE 802.3az standard. As shown later, the results are far better than in the Poisson case. Essentially, EEE benefits from the back-to-back nature of frames arrivals at the NIC, which allows to share the energy cost of T_w and T_s per I, P or B video frame.

- **GoP**: This case considers hypothetically that a whole GoP is aggregated before its transmission. It is assumed that the NIC aggregates the I video frame and its 11 B or P subsequent video frames and transmits them all at once. Such a per-GoP burstification is proposed since the video source is able to change its codification parameters every GoP period. When following this strategy, data bursts of about 12 × E(B) ≈ 135 Kbytes are transmitted every 12 × 40 = 480 ms.

- **Ideal:** Ideally, efficiency should be 100% if the T_s and T_w values were negligible with respect to the frame transmission times.

The simulated efficiency results are displayed in Figure 4.4 and Figure 4.5. For instance, taking Figure 4.4 case Movie, 100Base-Tx, resolution 640x480, I/P. This case considers the experiment of encoding the Movie clip using resolution 640x480 and no B frames. Table 4.2 first row states that the resulting Ethernet frame trace has the following features: bitrate 2.41Mbit/s, 27.69 Ethernet Frames per I video frame on average and

7.38 Ethernet Frames per P frame on average, 40ms inter-frame arrival time at the NIC. The Ethernet trace is then transmitted by a 100Mbit/s Ethernet NIC employing EEE. The theoretical efficiency in this case would be (Equation 4.2):

$$\eta_{real} = \frac{E(B)T_{Frame}}{T_w + \; E(B)T_{Frame} + T_s} = \frac{9.28 * 120\mu s}{30\mu s + 9{,}28 * 120\mu s + 200\mu s} = 0.82$$

which is the result shown in the Figure 4.4 (grey bar). A Poisson traffic profile with the same bitrate results in a much lower efficiency, 30% in the figure, since the link is in most cases awaken per frame. Theoretically this would be:

$$\eta_{Poisson} \approx \frac{120\mu s}{30\mu s + 120\mu s + 200\mu s} = 0.32$$

Finally, the *GoP burstification* achieves nearly 100% efficiency since the whole *GoP* is transmitted at once every 480ms:

$$\eta_{GoP} = \frac{9.28 * 12 * 120\mu s}{30\mu s + 9.28 * 12 * 120\mu s + 200\mu s} = 0.98$$

which is consistent with the result depicted in the Figure 4.4. Energy efficiency simulations have been performed for every single experimental case: video type and encoding settings. The following conclusions per link speed can be observed from the Figure 4.4 and Figure 4.5:

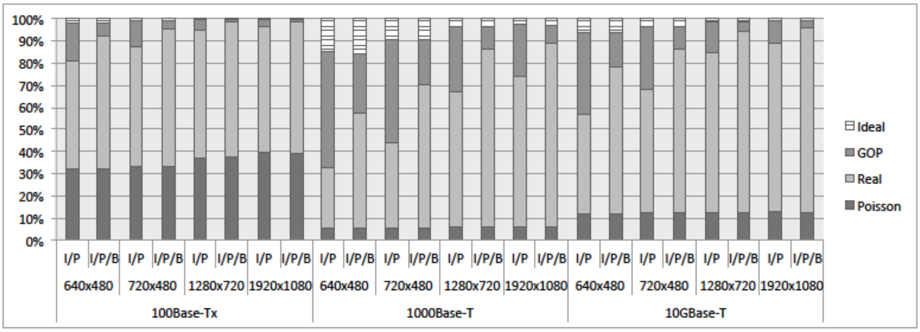
In 100BASE-TX, the efficiency η is in the order of 10-40% for Poisson traffic. This value rises to nearly 70-90% for video traffic due to the back-to-back nature of video traffic, and to nearly 100% when GoP burstification is applied.

In 1000BASE-T, the efficiency η is nearly 5-10% for Poisson traffic and rises to 40-75% for video traffic (depending on the video type and encoding settings), and to 90% or above when GoP-burstification is applied.

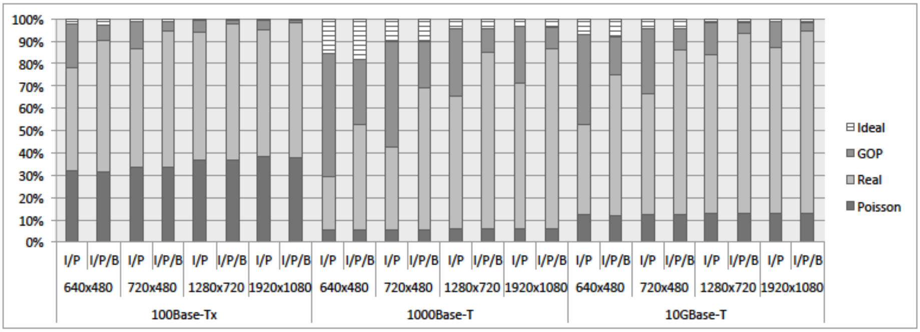
In 10GBASE-T, the efficiency η is around 10-15% for Poisson traffic, and rises to 60-80% for video traffic, and then to 95% when GoP-burstification is applied.

Hence, the back-to-back nature of video-streaming traffic allows a substantial energy efficiency increase in all cases, and particularly in the case of dynamic scenes (movie and sports) and/or high bitrate. Therefore, EEE allows a substantial power consumption decrease in video-streaming sources since the cost of *waking up* and *sleeping down* a link is shared per I, P or B frame, rather than per packet.

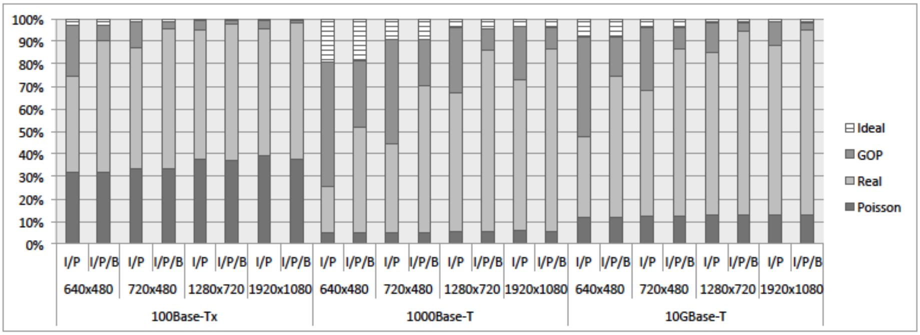
Concerning the GoP type, the use of B frames achieves an extra efficiency of 5-10% for 100BASE-TX, 20-25% for 1000BASE-T and about 20% for 10GBASE-T with respect to GoP type IPPPPPPPPPPP. Such a performance increase for GoP type IBBPBBPBBPBB is the consequence of larger data bursts than for GoP type IPPPPPPPPPPP, since the traces reveal that frames are transmitted together as a group of two B and one I/P every 120ms (instead of one video frame every 40ms). Therefore, the use of B frames is highly recommended for energy efficiency purposes.



(a) Movie

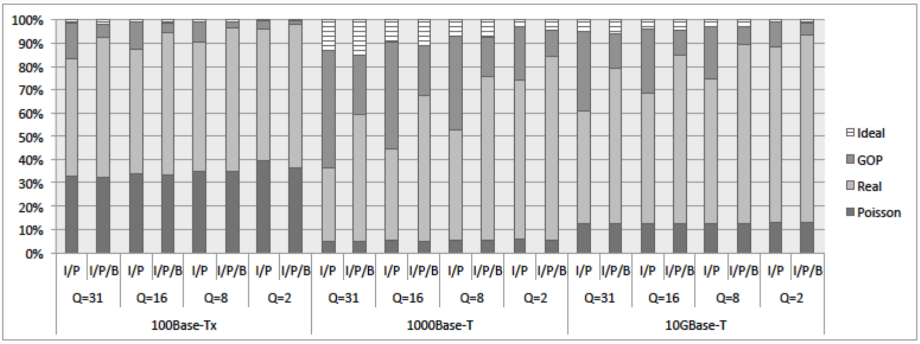


(b) News

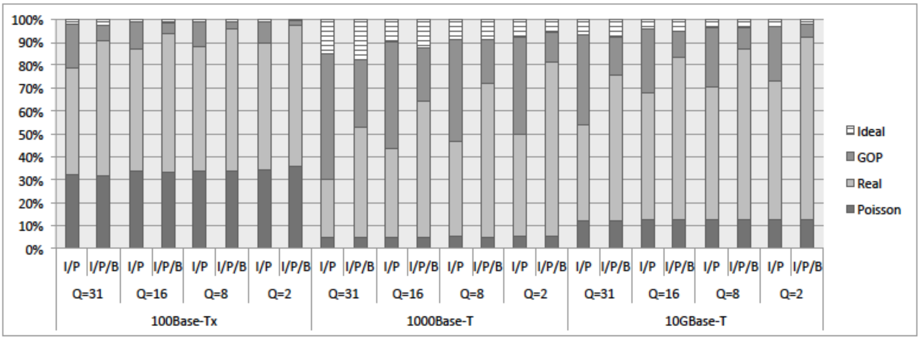


(c) Sport

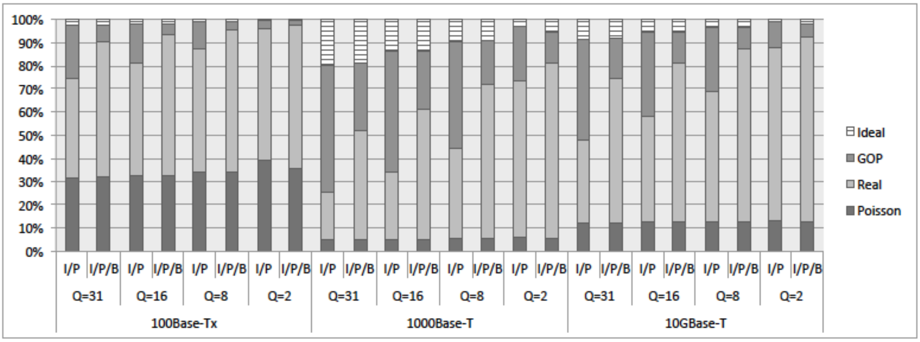
Figure 4.4 Efficiency for variable-Q



(a) Movie



(b) News



(c) Sport

Figure 4.5 Efficiency for fixed-Q

4.2.3. Energy efficiency transmitting a single H.264/SVC encoded video

Figure 4.6 shows the energy efficiency values for the H.264/SVC encoded video traces over 100BASE-TX, 1000BASE-T and 10GBASE-T, also taking into account each temporal scalability layer where *EL* and *BL* stand for *Enhancement Layer* and *Basic Layer* respectively. Essentially, given the hierarchical nature of B video frames, the whole GoP is transmitted in a single burst, thus producing very-high efficiency values, similar to those obtained in the hypothetical GoP burstification case of H.264/AVC. In addition, it could be observed that removing different enhancement layers reduce the average burst size, and its energy efficiency consequently (smaller bursts). The results obtained show that the H.264/SVC encoding settings provide better energy efficiency values than H.264/AVC while increasing the compression ratio.

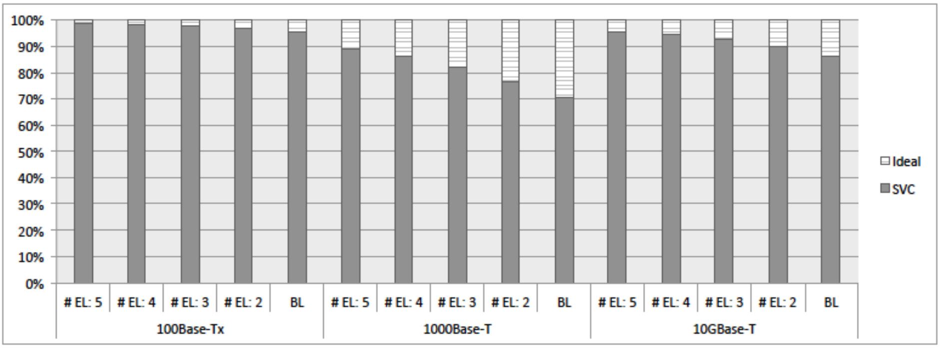


Figure 4.6 Efficiency for H.264/SVC encoded videos

The burty traffic pattern has been observed in video streaming traces in IPTV. The adoption of EEE in video streaming RTP servers may achieve large power savings. Taking into account the traffic patterns at output of the Ethernet NIC at access network of a Video Streaming server in IPTV network explained before. The traces presented in As the *I* video frames are usually larger than the *P* and *B* video frames, therefore they produce larger Bursts of Ethernet frames, E(B) to transmit, that could be evidenced comparing $E(B_I)$, $E(B_P)$ and $E(B_B)$ in Table 4.1 and Table 4.2 . Additionally, because there are 11 times more *P/B* video frames than *I* video frames, the average burst size $E(B)$ is usually closer to $E(B_P)$ and $E(B_B)$ than to $E(B_I)$.

Table 4.1 and Table 4.2, such traffic generation patterns are very suitable for EEE. This is because since the link only goes to sleep after the whole video frame is transmitted, thus sharing the cost of waking up and sleeping down the link, very much like in burst transmission EEE. Indeed, it is the video traffic source which generates the data bursts, and no burst-assembler is needed at the NIC, which brings the energy efficiency benefits of burst transmission EEE without the delay penalty.

This work has studied the behaviour and performance of Energy Efficiency Ethernet in video-streaming servers, showing good results in terms of energy savings. Essentially, the back-to-back transmission nature of the video streaming sources results in the transmission of a large portion of data per wake-up and sleep-down cycle. Indeed, the number of power-mode transitions is reduced with respect to other traffic patterns, and the energy overhead caused by switching between the active and low-power sleep mode is minimized. The experiments have further revealed that the use of B video frames produce larger transmission bursts than those without B frames, hence achieving larger efficiency values. Furthermore, it has been shown that the H.264/SVC codec not only increases the compression ratio with respect to H.264/AVC but is also more energy efficient since it generates GoP size data bursts.

Such behaviour is quite different from that observed in previous studies for other applications, where EEE showed worse performance. Hence, video streaming servers do not need any further technique to improve Energy Efficient Ethernet such as the aggregation of packets before transmission [77], since the video-streaming codecs already implement this functionality as part of its encoding process.

4.3. DASH traffic pattern

One of the main goals of this part of the study is to characterize the DASH traffic pattern and to analyse its relationship with the ethernet traffic in order to obtain conclusions about the efficiency of the IEEE 802.3az – EEE standard taking into account the results and methodology in the previous work about video traffic in IPTV and the energy efficiency presented in sections 4.1 and 4.2 respectively. The basic concepts about DASH have been presented in sub section 2.1.9, in this section, the testbed implemented, the experiments carried out and the resulting traffic patterns are presented.

4.3.1. DASH testbed

In order to characterize the ethernet traffic pattern in a DASH-based streaming platform, a DASH testbed has been deployed as shown in Figure 4.7. The testbed consists of a web server acting as a multimedia server and a web video player that will be the multimedia client. The multimedia content is hosted at the server. This content is a movie encoded in three different representations or qualities using the *ffmpeg tools*, so the multimedia content involves several MP4 containers and the MPD file. Additionally, in order to be able to perform an extensive experiment, the videos are available in different segment sizes, which are 4, 6 and 10 seconds. The DASH player is a *Javascript* web application, so although the *Shaka Player* [90] is also hosted at the server, the player application will run at the client side. *Apache* has been used as the web server entity. At the multimedia end-point, the *Chrome* browser is used as the HTTP client entity, where the *Shaka Player* will be loaded. In order to control the bandwidth available for the target streams, the *bandwidth throttling* from *Chrome developer tools* has been used. The testbed uses the TCP/IP protocol stack over Ethernet 802.3 1000BASE-T.

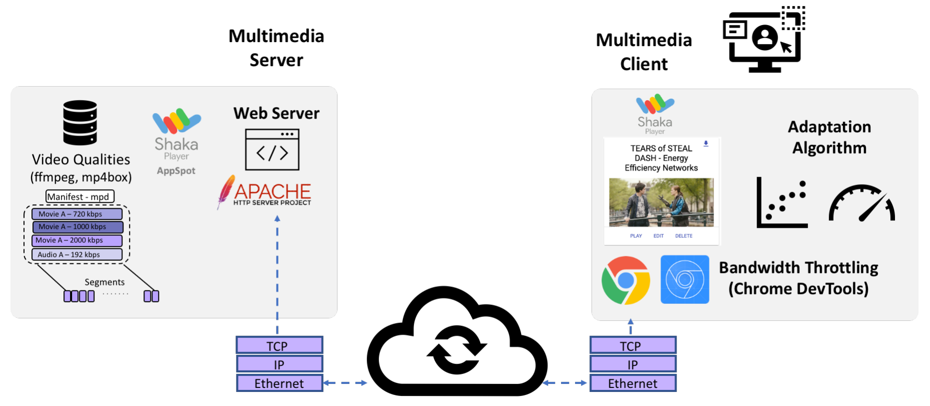


Figure 4.7 DASH testbed

Regarding the DASH content, an open movie called *Tears of Steel* from the *Blender Mongo Project* [91] has been used, encoded and prepared for streaming using DASH. Specifically, a portion of 180 seconds of the movie has been used, encoding three different video qualities in H.264, using the MP4 container. The same video and qualities are also available in 4, 6 and 10 seconds for the DASH video segment duration. Target encoding bitrate, average bitrate and resolution of the video representations are shown in Table 4.6. Additionally, one audio track is encoded and used in the streaming service. The difference between the target encoding bitrate and the average bitrate is due to the variability of the video and audio content and the nature of the codec, which is able to compress contiguous similar frames and samples more than different ones.

Table 4.6 Video available representations

Movie	Content Type	Target Coding Bitrate (Kbps)	Avg. Bitrate (Kbps)	Resolution (px)	Segment Size (s)
	High Quality Video	2000	1672	2592x1080	4, 6, 10
Tear of Steel	Medium Quality Video	1000	830	1728x720	4, 6, 10
	Low Quality Video	720	429	864x369	4, 6, 10
	Single Quality Audio	192	131	-	4, 6, 10

4.3.2. DASH traffic pattern

To obtain the traffic pattern for the DASH application and describe it at application and Ethernet levels, experiments were performed using the DASH testbed described in Figure 4.7. Four transmission scenarios were used that differ from each other in bandwidth limitation at the DASH client. Bandwidth throttling is carried out using the *Google Chrome developer tools throttling* feature. Bandwidth limitation for each scenario is adjusted to 1000 kbps, 2000 kbps and 3500 kbps, that are values close to the encoding objective of each one of the available video qualities, resulting in three scenarios with fixed bandwidth, and a fourth scenario with variable bandwidth every 30 seconds is also added to analyze the performance on variable scenarios. That four transmission scenarios are analyzed also for three different values of DASH video segment sizes, 4, 6 and 10 seconds. Conducting the experiments allows to obtain traces in the client and the server and to carry out the required processing and analysis to obtain the DASH traffic profile and the corresponding generated Ethernet traffic pattern.

The DASH application-level message exchange regarding the conducted experiments could be seen in Figure 4.8, which shows the messages from the loading, buffering and the steady states of the streaming session. This message exchange shown in the figure corresponds to an experiment of a *highest throttling* and 4 seconds segment case scenario. As it could be seen, at buffering state the DASH player requests the MPD file to the server, downloads it and reads it to start the retrieving process of the available segments of video content in a selected quality in subsequent HTTP GET messages. Also, it could be observed that the client starts different HTTP/TCP stream sessions to download the video segments for each one of the available qualities, thus requesting the first segment for each available representation. As Figure 4.8 depicts, the first group of HTTP request messages asking the first video segments of each different quality are those that have the corresponding HTTP 206 replay messages carrying a Partial Content (video/mp4). This HTTP response messages convey the first part of the MP4 containers named VID_1080p, VID_720p, VID_360p. Then, the player based on the available throttling (bandwidth) continues requesting the next segment of the selected quality.

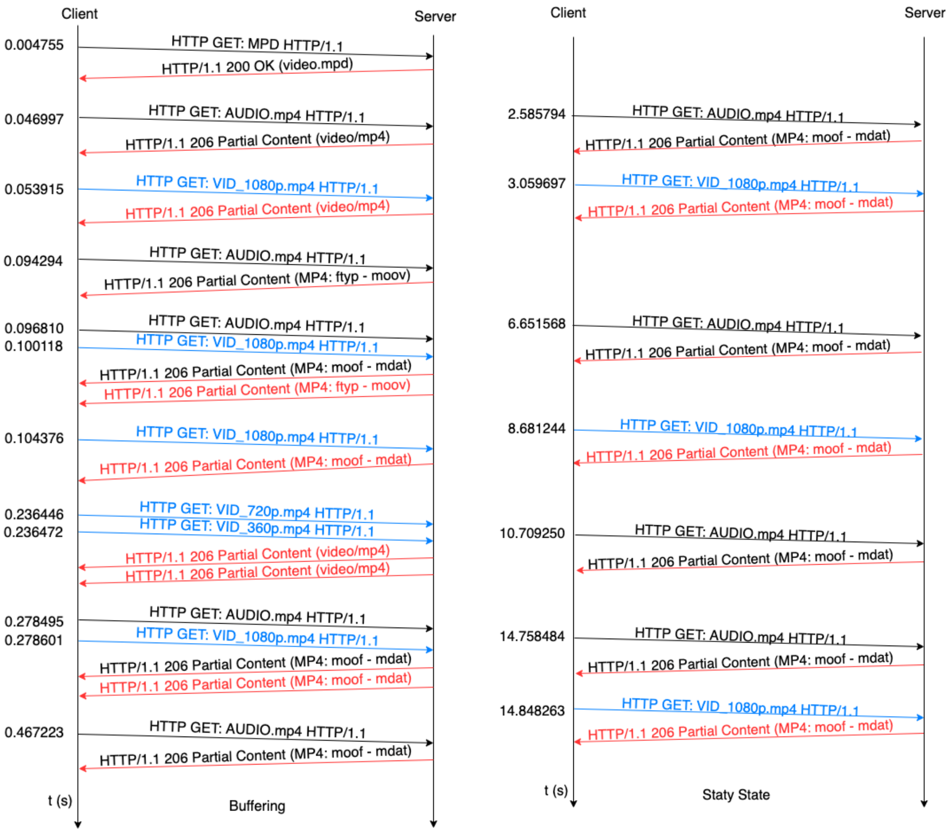


Figure 4.8 Buffering and steady state in a DASH session

On the other hand, at the steady state the DASH client requests segments of a selected quality each time it needs a new segment, and the requested quality depends on the available bandwidth. As it could be seen in Figure 4.8, the segment retrieving process is done with seconds of difference depending on the segment duration. If the available bandwidth changes during the streaming of the video, the DASH player will use the other HTTP/TCP stream sessions initialized beforehand to download the next low or high quality video segment.

Running the experiments for the scenarios described above, the DASH traffic pattern is obtained. Description of the DASH traffic pattern at application level is presented in Table 4.7, where there are parameters from the DASH segment perspective that includes: *average segment download time (seconds)*, *average segment size (Bytes)* and the *segment transmission average throughput (bps)* attending to the different bandwidth throttling scenarios. These results are measured at the client where the bandwidth throttling is applied. These parameters are shown to characterize and understand the behaviour of the DASH traffic and are measured at application level.

Table 4.7 DASH traffic pattern at client

Available Bandwidth (kbps)	Segment Size (s)	Avg. Segment Download Time (s)	Avg. Segment Size (Bytes)	Avg. Segment Throughput (bps)
3500	4 s	2.99 s	1.2 MB	3.3 Mbps
2000		3.09 s	643 kB	1.7 Mbps
1000		3.22 s	332 kB	820 kbps
3500-1000		3.05 s	791 kB	2.1 Mbps
3500	6 s	4.56 s	1.9 MB	3.5 Mbps
2000		4.70 s	977 kB	1.7 Mbps
1000		4.89 s	505 kB	827 kbps
3500-1000		4.56 s	1.2 MB	2.2 Mbps
3500	10 s	7.23 s	3.1 MB	3.4 Mbps
2000		7.48 s	1.5 MB	1.6 Mbps
1000		7.79 s	805 kB	826 kbps
3500-1000		7.48 s	2.3 MB	2.1 Mbps

Table 4.8 DASH traffic pattern at server

Available Bandwidth (kbps)	Segment Size (s)	Avg. Segment Upload Time (s)	Avg. Segment Size (Bytes)	Avg. Segment Throughput (Mbps)
3500	4	1.73 s	1.38 MB	16 Mbps
2000		1.24 s	661 kB	22 Mbps
1000		0.99 s	344 kB	36 Mbps
3500 – 1000		0.98 s	891 kB	67 Mbps
3500	6	3.25 s	1.99 MB	7.6 Mbps
2000		3.15 s	997 kB	6.4 Mbps
1000		2.34 s	516 kB	10 Mbps
3500 – 1000		1.76 s	1.3 MB	22 Mbps
3500	10	6.02 s	3.33 MB	4.1 Mbps
2000		5.45 s	1.66 MB	2.1 Mbps
1000		4.43 s	859 kB	1.4 Mbps
3500 – 1000		2.93 s	2.22 MB	2.9 Mbps

After illustrating this traffic characterization, the DASH traffic profile at the Ethernet connection level is presented above in

Table 4.8. These measurements correspond to the DASH traffic pattern at server side for the same streaming scenarios. The results presented are the *average segment upload time* (s), *average segment size* (Bytes) and *segment throughput* (Mbps).

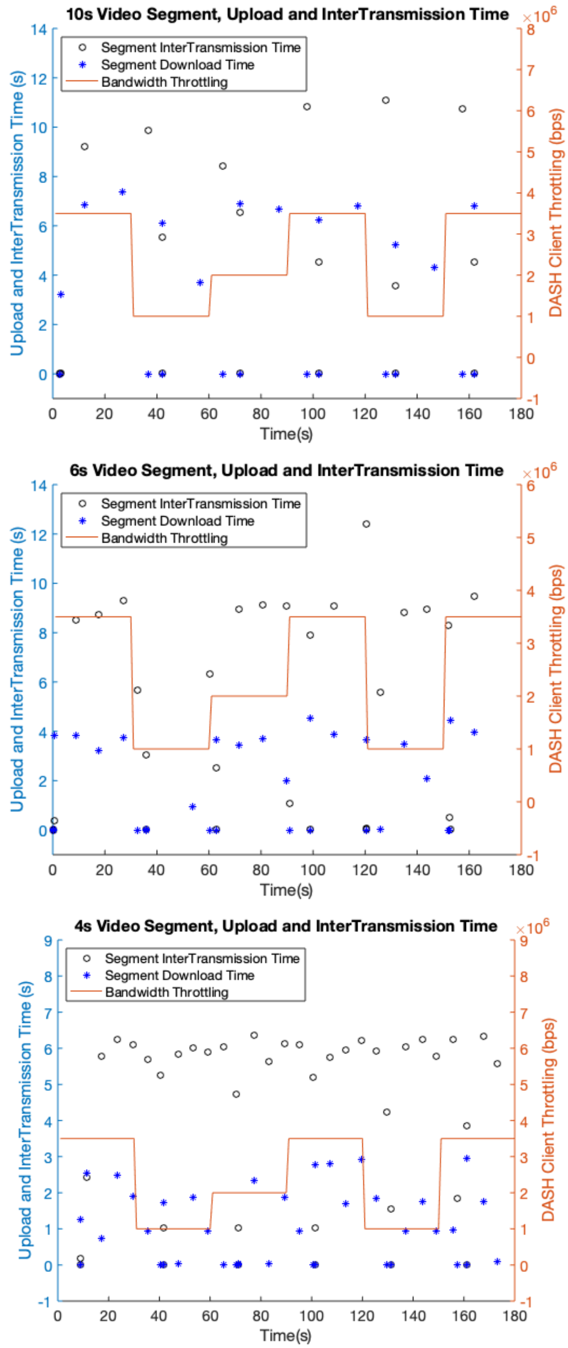


Figure 4.9 Segments inter-transmission and transmission time at server. Variable bandwidth throttling

Visually, the same DASH traffic characteristics are presented in Figure 4.9, showing for each video segment in the transmission the upload time and the *segment request interarrival time*. Understanding the start times of new segment transmission, the time between segment transmissions, the *segment download time*, the *segment size* in Bytes is relevant to determine the traffic nature of the corresponding Ethernet traffic generated. In the next section, the traffic pattern is analyzed in terms of ethernet frames

4.3.3. Ethernet DASH traffic pattern

Considering the outcomes obtained and presented in the previous section about the trace analysis in DASH at segment level, this section focuses on the DASH traffic pattern at Ethernet level.

First, the encapsulation of DASH content over Ethernet can be graphically observed depicted in Figure 4.10. At application layer, the MP4 video messages are shown, which represent the DASH movie segments, i.e. 4 seconds of movie in this case. Also Figure 4.10 presents the *Intra Segment Transmission Time* (ISTT), i.e. 5.5 seconds in average. Those video segments are transmitted in HTTP messages on TCP streams via IP datagrams. The group of IP packets that convey all MP4 messages (video segments) generates a burst of ethernet frames (E(B)) to be transmitted by the Network Interface Card. Each E(B) has a duration time, T_{Burst}, depending on the number of ethernet frames. For instance, a $T_{Burst} = 20$ ms is presented for an ethernet burst of 400 frames. Also, the *Intra Burst Transmission Time* (IBTT) is represented, which is approximately equal to the ISTT.

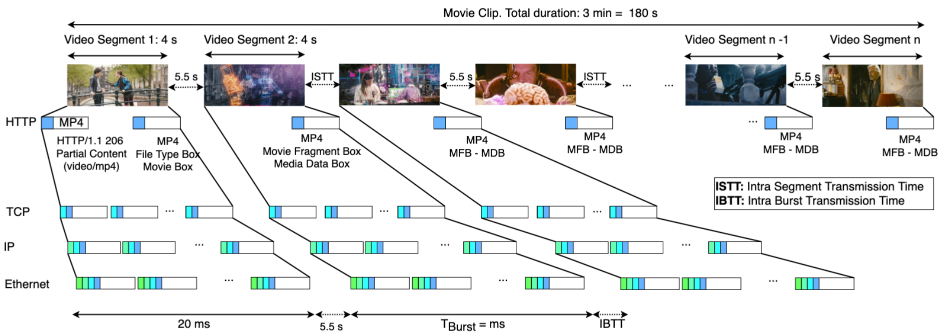


Figure 4.10 DASH protocol stack encapsulation

For the 4 bandwidth scenarios and the 3 different segment size use cases of the experiments carried out, the average time between ethernet burst in seconds and the average burst size E(B) in number of ethernet frames is obtained, as shown in Table 4.9. These values characterize the ethernet traffic pattern using DASH. This characterization of the ethernet traffic handled by DASH segments presents valuable values that allow the Ethernet Burst Size in number of frames and the transmission time to be calculated and, thus, the calculation and computation of the energy efficiency and energy savings, which will be presented in the following section.

Table 4.9 Ethernet DASH traffic at server

Bandwidth (kbps)	Segment Size (s)	Avg. Time Between Ethernet Burst (s)	Avg. Ethernet Burst Size E(B) (Frames)
3500	4	3.97	1104.87
2000		4.48	568.70
1000		4.73	326.97
3500 – 1000		4.21	753.63
3500	6	5.16	1706.10
2000		5.31	939.90
1000		5.95	502.85
3500 – 1000		5.85	1163.47
3500	10	8.00	2926.17
2000		8.05	1561.59
1000		9.46	820.67
3500 – 1000		5.97	1976.8

4.4. Energy Efficient Ethernet on DASH streams

The DASH ethernet traffic pattern described in the previous section is profiled using parameters like video *segment inter-arrival time, video segment download and upload time, ethernet burst size* E(B) in Ethernet frames and *time between ethernet bursts*. These parameters make possible to calculate the time the NIC is in *active* and *low power idle* modes. Therefore, the DASH segment efficiency can be proposed and calculated using the efficiency relationship of the total data transmission time over the total active and transitions time of the NIC, as in Equation 4.4. This DASH segment efficiency can be achieved because the observed DASH traffic pattern implies transmissions of MP4 content in segments, so the DASH content streams conform ethernet bursts with E(B) size.

4.4.1. Segment efficiency

$$\eta_{segment} = \frac{Total\ segment\ transmission\ time}{Total\ time\ in\ acive\ mode\ and\ transitions}$$ **Equation 4.4**

The DASH segment efficiency will be around 100% *when wake-up time*, Tw, and *sleep time*, Ts, are negligible from the ethernet burst transmission time perspective. The *ethernet burst transmission time* is equal to the *ethernet burst size in frames*, E(B), multiplied by the *ethernet frame transmission time*, T_{Frame}. Based on the GoP efficiency calculation method used in the previous IPTV and EEE efficiency study section 4.2 [88], the video segment efficiency for DASH streaming is presented in Equation 4.5.

$$\eta_{segment} = \frac{E(B) * T_{Frame}}{T_w + E(B) * T_{Frame} + T_S} \qquad \textbf{Equation 4.5}$$

As an example, Equation 4.6 presents the DASH video segment efficiency calculated for the case of the 3500 kbps bandwidth throttling and 4 s segment size. In Table 4.10, the segment efficiency is presented for the other DASH streaming experiments carried out in this study, for different available throttling, and different video segment sizes. E(B) values used are those presented in Table 4.9 and the T_{Frame}, T_w and T_s are constant values defined by the EEE 802.3az standard for the 1000Base-T transmission speed.

$$\eta_{segment} = \frac{1104.87 * 12\mu s}{30\mu s + 1104.87 * 12\mu s + 200\mu s}$$

$$\eta_{segment} = 0.9829$$

As shown in Table 4.10, it could be concluded that the segment efficiency takes advantage of the transmission of burst of ethernet frames, being higher when high bandwidth is available to send bursts of the corresponding DASH segments, thus implying that the NIC card could stay longer in low-power idle mode than high-power active mode. The calculated DASH segment efficiency is related to the energy consumption ratio of the wake-up and activation processes of the network card, going from low-power idle mode to active transmission mode. Then, DASH traffic pattern driven by video segments allows better efficiency ratio for IEEE 802.3az.

Table 4.10 Segment efficiency

Bandwidth (kbps)	Segment Size (s)	Segment Efficiency $\eta_{Segment}$
3500	4	0.9829
2000		0.9674
1000		0.9446
3500 – 1000		0.9752
3500	6	0.9889
2000		0.9800
1000		0.9632
3500 – 1000		0.9838
3500	10	0.9934
2000		0.9879
1000		0.9771
3500 – 1000		0.9904

Another important aspect of the DASH traffic pattern is the time between video segment transmissions. As presented in the previous section, the average time between video segments or Ethernet burst is approximately 3.97 s in the 3500 kbps case. Bearing this in mind, it is possible to calculate the total time of the transmission of the movie when the card is in *low power or low energy consumption mode* and in *high power or high energy consumption mode*, which enables the possibility to verify the energy savings achieved by activating the standard 802.3az on the network interface card.

As an example, the calculation is made for the scenario with bandwidth variations where different video segment file sizes are transmitted from different qualities of the video because of the variable bandwidth. The average time between ethernet bursts is approximately 4.21 seconds and 42 Ethernet Bursts are transmitted, which takes the ethernet card to be in low power energy consumption mode approximately 117.9 seconds. For a duration of the movie of 180 seconds, it is equivalent to an energy saving of 65.51% when implementing the EEE standard. Table 4.11 shows the average energy saving calculated for each streaming scenario.

Table 4.11 Segment efficiency and energy savings

Bandwidth (kbps)	Segment Size (s)	Segment Efficiency $\eta_{Segment}$	Avg. Energy Saving (%)
3500	4	0.9829	61.81
2000		0.9674	69.71
1000		0.9446	73.64
3500 – 1000		0.9752	65.51
3500	6	0.9889	51.68
2000		0.9800	53.11
1000		0.9632	62.88
3500 – 1000		0.9838	61.84
3500	10	0.9934	48.90
2000		0.9879	49.20
1000		0.9771	57.80
3500 – 1000		0.9904	53.08

Figure 4.11 evidences the fact that the average segment efficiency improves when the available bandwidth and the segment size are higher. For a given bandwidth, higher segment sizes will improve the segment efficiency. In the same way, for a given segment size, segment efficiency will be higher when the available bandwidth is also higher. Therefore, since the total energy saving, Figure 4.12, for the entire video stream depends on the segment size in time and bytes, using 4 s segments, for instance, in low quality video of about 1000 kbps for the target encoding bitrate causes the NICs to stay longer in low-power idle mode than when using 6 or 10 second segments.

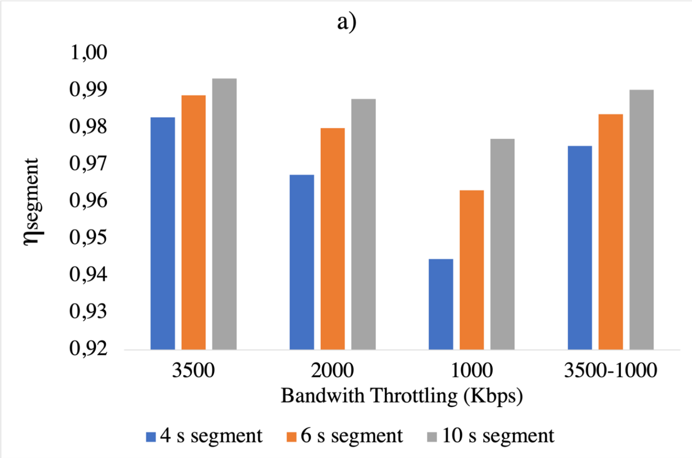


Figure 4.11 Segment efficiency

Moreover, observing the video segment duration and its relationship with energy savings, Figure 4.12, it is found that short video segments, e.g. 4 seconds, favours energy savings for all the streaming scenarios, especially in the case of low bandwidth and low quality video segments, because the video segment size and its transmission time present influence on longer low-power transmission times, and consequently, this entails transmitting faster short video segments in short high-power transmission times.

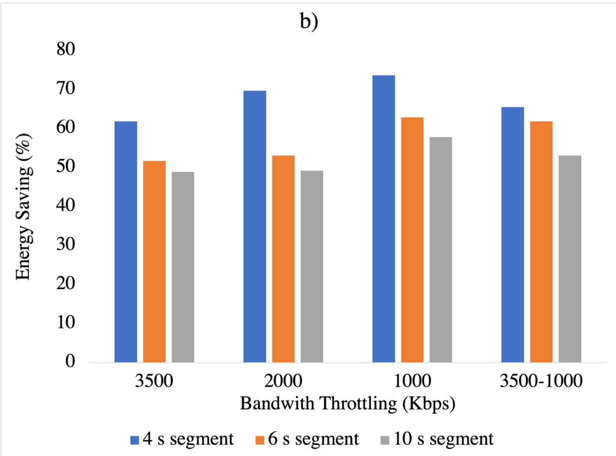


Figure 4.12 Energy saving (%).

In contrast, a higher segment efficiency is obtained with longer video segments with a little reduction in energy savings with respect to shorter video segments. However, the energy saving is still around 50% so implementing EEE is favourable on servers and clients involved with DASH traffic, regardless the video segment duration selected, although the impact of these variations at the application level do not affect energy efficiency much.

The EEE standard allows switching to low power mode when the network card is not transmitting, which saves energy efficiently when sending bursty ethernet traffic. Consequently, the analysis and experiments carried out in this study have shown that the ethernet traffic pattern generated by DASH streams implies burst of ethernet frames to be transmitted back-to-back, during short periods of time, in a periodic manner based on the DASH video segments size. That characteristic of the ethernet traffic pattern impacts positively on the efficiency and energy savings when implementing the IEEE 802.3az standard.

In order to carry out these analyses, the traffic pattern of a DASH streaming service has been characterized at application level and at ethernet level, using different segment sizes. This DASH service has been used to demonstrate under several bandwidth use case scenarios the improvements in segment efficiency, energy efficiency and energy savings that can be achieved using the EEE standard.

Given the large number of existing DASH video clients worldwide, this study aims to encourage the possibility of implementing the EEE standard on as many network interface cards as possible. This fact could lead to a high decrease in energy consumption globally. Energy efficiency research today is not only relevant due to the energy high cost and global warming, but it is also important to meet the QoS requirements of DASH streaming, the user QoE as well as the total energy needed to perform all functions of the video distribution network, as is analysed in other studies [92][93] and still there are challenges to meet and asses requirements of QoE to address in balance with energy efficiency [94]. As a future work the Live Low-Latency DASH could be characterized to study the influence in Energy Efficient Ethernet to improve energy savings under streaming live conditions using other encoding techniques.

4.5. Chapter Summary

In this chapter has been presented two video traffic patterns characterization, for IPTV and DASH streaming, by setting up the video streaming testbeds to trace, analysis and contribute. These two video traffic patterns have been used to evaluate the efficiency of the EEE standard and energy savings. The studies conclude that the bursty nature of the Ethernet traffic generated by this two video streaming applications favours the energy efficiency of the EEE standard leading to energy savings. Higher energy savings could be obtained if the EEE standard is implemented at NICs of video streaming servers and video streaming endpoints as a norm world-wide.

Chapter 5
Discussion and future work

This chapter presents general conclusion about this thesis. The next general conclusions complement results discussion presented in each chapter. Also, this chapter presents future research work that could be done in future postdoctoral research. The chapter is organized in two sections, in the first one the general conclusions are presented and on the other one the future research work is addressed.

5.1. General conclusions

1. In order to carry out studies and proposals to benefit QoS and QoE in video transmissions over broadband networks, it is necessary to know in detail how the video generation, encoding, adaptation, transmission, reception, decoding and reproduction process is carried out, as well as the techniques to measure and evaluate the quality of video transmissions. Due to the mentioned above, for the development of this project thesis, it has been necessary to carry out a complete study of the existing technologies and to configure and deploy real environments of video transmission and evaluation. This has demanded a significant dedication of resources for the development of activities for this purpose. It is highlighted that the development of this thesis has enabled the author to specialize and gain experience and knowledge in the Multimedia Communications area. As a contribution to the academic and scientific community, chapter 2 of this document is presented as a knowledge base for interested parties on the video streaming subject and to be an enabler for the fulfilment of the objectives of the thesis. In this thesis, it has been necessary to investigate and work with free software tools and to scripts design and program to automate video transmissions, measurements and results analysis that have been reached and presented in this document.

2. OBS networks have also been studied in detail during the development of this thesis. It has been found that the OBS network proposal is interesting to undertake Telematics Engineering studies for algorithms and protocols design. For this research work, a complete OBS network study has been made, regard its

architecture, protocols, algorithms and how QoS is proposed to be provided. The OBS network knowledge and its challenges has been the fundamental basis for the thesis contributions regard video transmissions evaluation and algorithm proposals to provide QoS to video transmissions. During the development of this study, opportunities to hold update conferences for the academic community have had arisen, so conferences, seminars has been held at the Universitat Politècnica de València, Universidad Santo Tomás de Bucaramanga and the Universidad Sergio Arboleda Bogotá to teach and update undergraduate and postgraduate students.

3. Undertake research on transmission of multimedia content over telecommunications networks, in addition to requiring the researcher to be an expert in the theory and techniques of transmission and evaluation of video streaming, also requires full knowledge of telematics engineering and communications networks. For the development of this doctoral thesis, it has been necessary to fully study optical communication networks, optical switching technologies, specifically burst optical switching. Therefore, the researcher of this thesis learnt in detail aspects of design and operation of OBS networks and knew in detail the aspects of video transmission over broadband networks. Due to the above, the researcher has integrated these two areas of knowledge proposing a methodology and proposals to carry out experiments using testbeds. This has been an important milestone, to be able to simulate video transmissions over OBS networks based on real video traffic traces and thus be able to carry out their evaluation to highlight problems and be able to make the contributions as a result of this doctoral thesis.

4. The methodology and setup for video traffic over OBS networks evaluation provided in this study has been put into practice for the burst assembly algorithms analysis. The methodology has been tested and it has found to been useful to draw conclusions about the OBS network challenges in terms of improvements for the assembly and scheduling algorithms to benefit video traffic without affecting data traffic. The results of this phase allowed to demonstrate the advantages of the evaluation environment and to project contributions for the algorithms assembly of bursts and scheduling in OBS networks. Something interesting is that this work has attracted the attention of researchers in the OBS area and has been presented at a national telematics engineering thematic and to the Building the Optical Network in Europe (BONE) network of excellence. At national level, we were invited by researchers from the Universidad Publica de Navarra to present the video over OBS evaluation methodology and tesdbed. At the European level, these results allowed to propose joint research activities with researchers from University of Patras and the University of Essex. This experience has been interesting in the sense of showing how the research thesis has been generating collaboration and interest that leads to results presented in this thesis.

5. Adaptive burst assembly algorithm (ABAA) has been proposed to infer the quality of video when it is transmitted over an OBS network, guaranteeing a bounded delay while maintaining a bounded dynamic burst size threshold in the burst assembly

process. In ABAA the burst size threshold is dynamically computed taking into account RTP/UDP packet loss measurements at the ingress node. The ABAA proposal has been based on the TCP-Friendly strategy. ABAA avoids the problems generated by congestion that can be generated in the entry node of the OBS network. The simulation results allowed to demonstrate that ABAA reduces video frames losses at the entry node, thus maintaining the transmitted video quality when there is a dynamic variation of traffic and congestion.

6. A Preemptive LAUC-VF scheduling algorithm has been also proposed as a multi-class scheduling algorithm to provide QoS by differentiating classes of service. The algorithm has been proposed to make their parameterization flexible with a parameter that expresses the guaranteed preemption probability. The proposal has been developed through collaboration in research stays at University of Patras and the University of Essex through a joint research activity within the BONE European network of excellence.

7. This thesis has allowed to gain experience in Multimedia Communications, which has made it possible to setup testbeds to characterize video traffic patterns for the study regard energy efficiency standards implemented at the transmission and reception equipment that implement the Ethernet standard, specifically the IEEE 802.3az Energy Efficient Ethernet standard (EEE). Two testbeds have been carried out, one to characterize the video traffic in IPTV networks and the other one to characterize the video traffic on the Internet using the DASH standard. To carry out energy efficiency calculations in EEE for video traffic, programs have been developed for the analysis of real video transmission traces, evidencing that the EEE standard is efficient under video transmissions scenarios, considering the bursty traffic nature of video. Implementing the EEE standard allows energy savings at servers and clients that implements the EEE standard. The EEE research of this thesis has an initial collaboration with researchers from Universidad Carlos III Madrid y la Universidad de Nebrija.

8. The development of this research project has allowed making contributions that have been published in conferences and journals allowing the results dissemination. In addition to the above, there has been the opportunity to present the results within the framework of networks of excellence, thematic networks, European summer, and master schools and to make contributions to national research agendas by participating in technological platforms in photonics and the Internet, such Fotonica21 in Spain.

9. The experience developed during the development of this doctoral thesis has allowed the participation in technological development and research projects where there has been the opportunity to develop software and adapt technologies and devices for multimedia communications area, specifically in the field of digital signage and telemedicine, carrying out knowledge transfer. These additional results related to the development of the thesis are presented in the appendices of this document.

5.2. Future work

During the development of this thesis, experience has been gained in the development of studies on the transmission of multimedia flows, considering the process chain from the adaptation and encoding of video sources. Also, the evaluation of the quality of the video transmissions from the point of view of the network and from the perception of the end user. Through the generation of testbeds for the study of algorithms, protocols and standards linked to digital video distribution, contributions have been made both in protocols and algorithms and in recommendations to offer QoS, QoE and contributing to the reduction of energy consumption.

In the final part of the work, it has been found that there is an opportunity to continue researching in the line of energy efficiency aspects regarding video transmissions, taking into account, on the one hand, the new coding standards, novelties in the distribution of video on the Internet such as *Low Latency DASH* and on the other hand the evolution of the EEE standard to undertake studies in high speed switching equipment at core network, where multiple NICs are necessary in a single equipment. The goal is to reduce energy consumption and take advantage of the new features of video transmission standards.

References

[1] Begen, A.C.; Akgul, T.; Baugher, M., "Watching Video over the Web: Part 1: Streaming Protocols," in Internet Computing, IEEE, vol.15, no.2, pp.54-63, March-April 2011.

[2] Wes Simpson. Video Over Ip, Second Edition: Iptv, Internet Video, H.264, P2p, Web Tv, and Streaming a Complete Guide to Understanding the Technology (2 ed.). Focal Press, 2008.

[3] ISO/IEC/JTC1/SC29/WG11, Overview of the MPEG-4 standard (July 2000).

[4] A. Lo, G. Heijenk, I. Niemegeers, "Performance Evaluation of MPEG-4 Video Streaming over UMTS Networks using an Integrated Tool Environment", Proceedings SPECTS 2005, 2005 International Symposium on Performance Evaluation of Computer and Telecommunication Systems, Philadelphia, PA, USA, July 24--28, 2005 3. MPEG 4.

[5] J. Gross, J. Klaue, H. Karl, A. Wolisz, Cross-layer optimization of OFDM transmission systems for MPEG-4 video streaming, Computer CommunicationsVolume 27, Issue 11, Applications and Services in Wireless Networks, 1 July 2004, Pages 1044-1055.

[6] P. Seeling, M. Reisslein, Video transport evaluation with H.264 video traces, IEEE Communications Surveys Tutorials PP (99) (2011) 1 –24. doi:10.1109/SURV.2011.082911.00067.

[7] H. Schwarz, D. Marpe, T. Weigand, Overview of the scalable video coding extension of the H.264/AVC standard, IEEE Trans. Circuits Syst. Video Technol. 17 (9) (2007) 1103–1120).

[8] “A Transport Protocol for Real-Time Applications”, RFC 3550, Network Working Group.

[9] Thomson, Gavin; Shah, Athar (2017). "Introducing HEIF and HEVC" (PDF). Apple Inc. Retrieved 5 August 2019.

[10] B. Bross et al., "Overview of the Versatile Video Coding (VVC) Standard and its Applications," in IEEE Transactions on Circuits and Systems for Video Technology, vol. 31, no. 10, pp. 3736-3764, Oct. 2021, doi: 10.1109/TCSVT.2021.3101953.

[11] Diyanov Nikolov, A. (2021). Evaluación del nuevo codificador VVC/H266 para sistemas de streaming. Universitat Politècnica de València. http://hdl.handle.net/10251/174471

[12] L. Rubio, “A Dynamic Adaptive HTTP Streaming Vídeo Service for Google Android,” M.S. thesis, dissertation. Dept. School of Information and Communication Technology (ICT), Eng. Royal Institute of Technology (KTH)., Stockholm. Sweden, 2011

[13] Ernst Biersack, Christian Callegari, Maja Matijasevi. Data Traffic Monitoring and Analysis: From Measurement, Classification, and Anomaly Detection to Quality of Experience, Springer Berlin Heidelberg, Mar 21, 2013.

[14] ITU, “Methodology for the Subjective Assessment of the Quality of Television Pictures”, ITU-R Recommendation BT.500-10, 2000.

[15] J. Klaue, B. Rathke, and A. Wolisz, "EvalVid - A Framework for Video Transmission and Quality Evaluation", In Proc. of the 13th International Conference on Modelling Techniques and Tools for Computer Performance Evaluation, Urbana, Illinois, USA, September 2003.

[16] Agata Jung, "Comparison of Vídeo Quality Assessment Methods". Master thesis, Blekinge Institute of Technology, Karlskrona, Sweden, 2017.

[17] Li, Z.; Aaron, A.; Katsavounidis, I.; Moorthy, A.; Manohara, M. Toward a practical perceptual video quality metric. The Netflix Tech Blog. 2016. Available online: https://netflixtechblog.com/toward-a-practical-perceptual-video-quality-metric-653f208b9652 (accessed on 28 Oct 2021).

[18] C. Qiao and M. Yoo, “Optical burst switching (OBS) – a new paradigm for an optical Internet”, Journal of High Speed Networks, No. 9, 1999, pp. 69-84.

[19] Jason P. Jue and Vinod M. Vokkarane. 2011. Optical Burst Switched Networks (1st ed.). Springer Publishing Company, Incorporated

[20] K.C. Chua, M. Gurusamy, Y. Liu and M.H. Phung Quality of Service in Optical Burst Switched Networks Springer, 2007.

[21] S. Oh y Kang. M., "A Burst Assembly Algorithm in Optical Burst Switching Networks", en Proceedings of Optical Fiber Communication Conference and Exhibit (OFC), Anaheim, pp. 771–773, marzo 2002.

[22] X. Yu, Y. Chen, and C. Qiao, “A Study of Traffic Statistics of Assembled Burst Traffic in Optical Burst Switched Networks”, Proceedings, SPIE Optical Networking and Communication Conference (OptiComm) 2002, Boston, MA, pp. 149-159, July-Aug 2002.

[23] X., Li, J. Cao y Y., Qiao, C. Chen, "Assembling TCP/IP Packets in Optical Burst Switched Networks", en Proceedings of IEEE Global Communications Conference (GLOBECOM), vol. 3, Taipei, pp. 2808 - 2812, November 2002.

[24] D. Kang, W.H. Yang, J.Y. Lee y Y.C. Kim, "Dynamic Time based Burst Assemble scheme to reduce energy consumption in OBS network with Low-Power Idle", en Proceedings of the 16th Asia-Pacific Conference on Communications (APOC), Auckland, pp. 183-187, November 2010.

[25] R. Nakkeeran, C. Balaji, N. Gujju, Balakumaran K. y M.K. Sasidharan, "Enhanced Burst Assembly Mechanism with Hybrid Signalling Scheme for Optical Burst Switched Networks",

en Proceedings of the 15th International Conference on Advanced Computing and Communication (ADCOM), Guwahati, pp. 727-732, December 2007.

[26] M. Düser, I. de Miguel y P. Bayvel, "Timescale analysis for wavelength-routed optical burst-switched (WR-OBS) networks", en Proceedings of Optical Fiber Communication Conference (OFC), Anaheim, paper WG7, Marz 2002.

[27] K. Ramantas, K. Vlachos, O. González De Dios y C. Raffaelli, "Window-based burst assembly scheme for TCP traffic over OBS", Journal of Optical Networking, vol. 7, no. 5, pp. 487-495, 2008

[28] T. Hashiguchi, X. Wang, H. Morikawa y T. Aoyama, "Burst Assembly Mechanism with Delay Reduction for OBS Networks ", en Proceedings of Conference on the Optical Internet (COIN), Melbourne, pp. 664-666, July 2003.

[29] M. Yoo, C. Qiao y S. Dixit, "QoS Performance of Optical Burst Switching in IPOver-WDM Networks", IEEE Journal on Selected Areas in Communications, vol. 18, no. 10, pp. 2062-2071, October 2000.

[30] I. Baldine, G. N. Rouskas, H. G. Perros y D. Stevenson, "JumpStart: A Just-In-Time Signaling Architecture for WDM Burst-Switched Networks", IEEE Communications Magazine, vol. 40, no. 2, pp. 82–89, February 2002.

[31] M. Klinkowski, D. Careglio, J. Solé-Pareta y M. Marciniak, "Performance Overview of the Offset Time Emulated OBS Network Architecture", Journal of Lightwave Technology, vol. 27, no. 4, pp. 2751 - 2764, July 2009.

[32] E. Mannie, "Generalized Multi-Protocol Label Switching (GMPLS) Architecture", RFC 3945, octubre 2004

[33] A. Farrel y J. P. Vasseur, "A Path Computation Element (PCE)-Based Architecture", RFC 4655, agosto 2006.

[34] F. Zhang, Y. Lee, J., Bernstein, G. Han y Xu Y., "OSPF Extensions in Support of Routing and Wavelength Assignment (RWA) in Wavelength Switched Optical Networks (WSONs)", draft-zhang-ccamp-rwa-wson-routing-ospf-03, March 2010.

[35] M. Düser y P. Bayvel, "Modelling of Optical Burst-Switched Packet Networks", en Proceedings of European Conference on Optical Communications (ECOC), Munich, paper 1.3, September 2000.

[36] M. Yoo and C. Qiao. A new optical burst switching protocol for supporting quality of service. In SPIE Proceedings, All Optical Networking: Architecture, Control and Management Issue, 3531:396-405, Nov. 1998.

[37] Kian Jui Tan, Li-Jin Ian Thng, and Hailong Li, "Absolute priority just-enough-time scheme for absolute quality of service in optical-burst-switching networks," J. Opt. Netw. 3, 573-588 (2004) .

[38] A Kaheel, H Alnuweiri, “A strict priority scheme for quality-of-service provisioning in optical burst switching networks”, in proceed. of ISCC 2003.

[39] Gjessing. Stain. Maus. Arne. “Discrete Event Simulation of a large OBS Network”, Systems, Man and Cybernetics, 2005 IEEE International Conference on Volume 3, Issue, 10-12 Oct. 2005 Page(s): 2512 - 2517 Vol. 3.

[40] O'Mahony, M.J . Results from the COST 239 project. Ultra-High Capacity Optical Transmission Networks. Optical Communication, 1996. ECOC '96. 22nd European Conference on, Vol.2, Iss., 15-19 Sept. 1996 Pages: 11- 18 vol.2

[41] Moonsoo Kang and Jeonghoon Mo. “On the Pacing Technique for High Speed TCP Over Optical Burst Switching Networks”. Advanced Communication Technology, 2006. ICACT 2006. The 8th International Conference Volume 2, Issue , 20-22 Feb. 2006 Page(s): 1421 - 1424

[42] Sukyoung Lee. “Packet-based burst queue modelling at an edge in optical-burst switched networks”. Computer Communications 29(5): 634-641 (2006)

[43] Kim. LaeYoung, Lee. SuKyoung, Song. JooSeok. “Dropping Policy with Burst Retransmission to Improve the Throughput of TCP Over Optical Burst-switched Network”s. Photonic Network Communications, Vol. 12, No. 1. (July 2006), pp. 29-40.

[44] JoungYul CHOI, JinSeek CHOI, Minho KANG. “Performance Analysis of Optical-Level Buffered Optical Burst Switching Node with Retransmission Techinque”. IEICE Transactions on Information and Systems 2006 E89-D(2):452-45.

[45] Keping Long, R. S. Tucker and Chonggang Wang, "A new framework and burst assembly for IP DiffServ over optical burst switching networks," GLOBECOM '03. IEEE Global Telecommunications Conference (IEEE Cat. No.03CH37489), 2003, pp. 3159-3164 vol.6, doi: 10.1109/GLOCOM.2003.1258818

[46] Soares, V.N.G.J.; Veiga, I.D.C.; Rodrigues, J.J.P.C., "OBS Simulation Tools: A Comparative Study," in Communications Workshops, 2008. ICC Workshops '08. IEEE International Conference on , vol., no., pp.256-260, 19-23 May 2008.

[47] Xiph.org Foundation, Video Test Media [derf's collection], Akiyo Video, On-line: https://media.xiph.org/video/derf/ (Accessed 04/09/2021).

[48] Teerawat Issariyakul, Ekram Hossain. Introduction to Network Simulator NS2. Springer US, 2009.

[49] Ge, A.; Callegati, F.; Tamil, L.S., "On optical burst switching and self-similar traffic," Communications Letters, IEEE , vol.4, no.3, pp.98-100, Mar 2000.

[50] V.M. Vokkarane, K. Haridoss, J.P. Jue, Threshold-based burst assembly policies for QoS support in optical burst-switched networks, in: Proc. SPIE OptiComm'02, July 2002, pp. 125-136.

[51] Gowda, S.; Shenai, R.K.; Sivalingam, K.M.; Cankaya, H.C., "Performance evaluation of TCP over optical burst-switched (OBS) WDM networks," Communications, 2003. ICC '03. IEEE International Conference on , vol.2, no., pp. 1433-1437 vol.2, 11-15 May 2003.

[52] Xiang Yu.; Chunming Qiao., “TCP Performance over OBS Networks with Multiple Flows Input” Broadband Communications, Networks and Systems, 2006. BROADNETS 2006. 3rd International Conference on, vol., no., pp.1-10, 1-5 Oct. 2006.

[53] Kostas Ramantas, Kyriakos Vlachos, Óscar González de Dios, Javier Aracil and Carla Raffaelli, “Timer-based burst assembly algorithms and their effect in TCP Congestion window”, OSA J. Optical Networks, vol. 7, Iss. 5, pp. 487-495, May 2008.

[54] Optical Internet Research Center, "OIRC OBS-ns Simulator", http://wine.icu.ac.kr/~obsns/index.php, accessed at April, 2008.

[55] Guoqiang Hu; Kohn, M., "Evaluation of Packet Delay in OBS Edge Nodes," Transparent Optical Networks, 2006 International Conference on , vol.3, no., pp.66-69, 18-22 June 2006.

[56] M.V. Rodrigo and J. Gotz, "An analytical study of optical burst switching aggregation strategies," Workshop on OBS'04, 2004.

[57] K. Christodoulopoulos, E. Varvarigos and K. Vlachos, “A new Burst Assembly Scheme based on the Average Packet Delay and its Performance for TCP Traffic”. Optical Switching and Networking, Elsevier 2007.

[58] Pastor A. Burst formation strategies in OBS. U. Navarra. https://www.tlm.unavarra.es/investigacion/seminarios/.

[59] Yang Chen, et al., “Optical burst switching: a new area in optical networking research,” Network, IEEE , vol.18, no.3, pp. 16-23, June 2004.

[60] Basem Shihada, Pin-Han Ho, A novel TCP with dynamic Burst-Contention Loss notification over OBS networks, Computer Networks, Volume 52, Issue 2, pp. 461-471, Feb. 2008.

[61] Xiang Yu, et al., “Traffic statistics and performance evaluation in optical burst switched networks,” Lightwave Technology, Journal of, vol.22, no.12, pp. 2722-2738, Dec. 2004.

[62] Xiang Yu.; Chunming Qiao., “TCP Performance over OBS Networks with Multiple Flows Input,” Broadband Communications, Networks and Systems, 3rd International Conference on, Oct. 2006.

[63] S. Floyd; M. Handley; J. Padhye; J. Widmer, “TCP Friendly Rate Control (TFRC): Protocol Specification”, RFC 5348, Sep. 2008.

[64] J. Klaue; B. Rathke; A. Wolisz, “EvalVid - A new Framework for Video Transmission and Quality Evaluation,” Computer Performance, Springer, vol. 2794, pp. 255-272, Sept. 2003.

[65] Y. Xoing, M. Vandenhoute, and C. Cankaya, “Control architecture in optical burst-switched WDM networks”, IEEE J. on selected areas in communications, 18:1838–1851, Oct. 2000.

[66] K. Ramantas, T.R. Vargas, J.C. Guerri and K. Vlachos. “A preemtive scheduling scheme for flexible QoS provisioning in OBS networks”. Broadnets 2009, Sep. 14 – 17, 2009. Madrid, Spain.

[67] H. Øverby, N. Stol, Providing quality-of-service in optical packet/ burst switched networks with the preemptive drop policy, in proceedings of ICN, vol. 1, pp. 312–319, 2004.

[68] M. Yuksel, B. Sikdar, K.S. Vastola, and B. Szymanski, 'Workload generation for ns Simulations of Wide Area Networks and the Internet', in proceedings of CNDS Conference, 2000.

[69] H. Kalva, The H.264 video coding standard, IEEE Multimedia 13 (4) (2006) 86–90.

[70] P. Seeling, M. Reisslein, B. Kulapala, Network performance evaluation using frame size and quality traces of single-layer and two-layer video: A tutorial, IEEE Communications Surveys & Tutorials 6 (3) (2004) 58–78.

[71] S. Han, S. Lisle, G. Nehib, IPTV transport architecture alternatives and economic considerations, IEEE Communications Magazine 46 (2) (2008) 70–77.

[72] M. Gupta, S. Singh, Greening of the Internet, in: Proc. ACM SIGCOMM, 2003, pp. 19–26.

[73] IEEE Std 802.3az Energy Efficient Ethernet-2010 (2010).

[74] K. Christensen, P. Reviriego, B. Nordman, M. Bennett, M. Mostowfi, J. Maestro, IEEE 802.3 az: the road to energy efficient ethernet, IEEE Communications Magazine 48 (11) (2010) 50–56.

[75] P. Reviriego, J. A. Hernández, D. Larrabeiti, J. A.Maestro, Performance evaluation of Energy Efficient Ethernet, IEEE Communications Letters 13 (9) (2009) 697–699.

[76] P. Reviriego, K. Christensen, J. Rabanillo, J. Maestro, An Initial Evaluation of Energy Efficient Ethernet, IEEE Communications Letters 15 (5) (2011) 578–580.

[77] P. Reviriego, J. A. Maestro, J. A. Hern´andez, D. Larrabeiti, Burst transmission for Energy Efficient Ethernet, IEEE Internet Computing 14 (4) (2010) 30–37.

[78] C. W. Paper, Cisco Visual Networking Index: Forecast and Methodology, 2011-2016, Tech. rep., Cisco Systems inc. (2012).

[79] A. Odlyzko, Data networks are lightly utilized, and will stay that way, Review of Network Economics , (2003).

[80] Myungsik Yoo , Chunming Qiao , "Just-Enough-Time (JET): A high speed protocol for bursty traffic in optical networks" Vertical-Cavity Lasers, Technologies for a Global Information Infrastructure, WDM Components Technology, Advanced Semiconductor Lasers, Gallium Nitride Materials, Processing, 1997 Digest of the IEEE/LEOS Summer Topical Meetings Volume , Issue , 11-15 Aug 1997 Page(s):26 - 27

[81] El-Bawab, T.S.; O'Mahony, M.J.; Jayasumana, A.P.; A European multiwavelength optical network. Computers and Communications, 1997. Proceedings., Second IEEE Symposium on 1-3 July 1997 Page(s):568 - 573

[82] Gauger, C.M.; Kohn, M.; Scharf, J. Comparison of contention resolution strategies in OBS network scenarios. Transparent Optical Networks, 2004. Proceedings of 2004 6th International Conference on, Vol.1, Iss., 4-8 July 2004 Pages: 18- 21 vol.1.

[83] Rodrigues, J.J.P.C.; Freire, M.M.; Lorenz, P. One-way resource reservation protocols for IP over optical burst switched mesh networks. Systems Communications, 2005. Proceedings, Vol., Iss., 14-17 Aug. 2005 Pages: 229- 23.

[84] Elbiaze, H., "Quality of Service in OBS Networks: Issues, Challenges and Solutions," Transparent Optical Networks, 2006 International Conference on, vol.3, no.pp.23-26, June 2006.

[85] Qiong Zhang; Vokkarane, V.M.; Jue, J.P.; Biao Chen, "Absolute QoS differentiation in optical burst-switched networks," Selected Areas in Communications, IEEE Journal on, vol.22, no.9pp. 1781- 1795, Nov. 2004

[86] Kaheel, A.; Khattab, T.; Mohamed, A.; Alnuweiri, H., "Quality-of-service mechanisms in IP-over-WDM networks", Communications Magazine, IEEE, vol.40, no.12pp. 38- 43, Dec 2002.

[87] Kim, S.; Choi, J.S.; Kang, M., "Providing absolute differentiated services for optical burst switching networks: loss differentiation", Communications, IEE Proceedings-, vol.152, no.4pp. 439- 446, Aug 2005.

[88] Antonio de la Oliva, Tito R. Vargas Hernández, Juan Carlos Guerri, José Alberto Hernández, Pedro Reviriego, "Performance analysis of Energy Efficient Ethernet on video streaming servers", Computer Networks, Available online 11 October 2012, ISSN 1389-1286, 10.1016/j.comnet.2012.09.019

[89] ISO/IEC 23009-1, Dynamic adaptive streaming over HTTP (DASH) -Part 1: media presentation description and segment formats, 2012. (2017)

[90] Shaka Player Demo. [Online]. Available: https://shaka-player-demo.appspot.com. [Accessed: 01-Jul-2021]

[91] Tears of Steel Movie, Mongo Open Movie project. Blender Foundation. Available online: https://mango.blender.org/ (Accessed 01-Jul-2021)

[92] Bolla, R., Bruschi, R., Davoli, F., Depasquale, E.V. (2018). Energy-Efficient Management and Control in Video Distribution Networks: 'Legacy' Hardware-Based Solutions and Perspectives of Virtualized Networking Environments. In: Popescu, A. (eds) Greening Video Distribution Networks. Computer Communications and Networks. Springer, Cham. https://doi.org/10.1007/978-3-319-71718-0_2.

[93] Popescu, A., Yao, Y., Fiedler, M., Ducloux, X. (2018). Video Distribution Networks: Models and Performance. In: Popescu, A. (eds) Greening Video Distribution Networks. Computer Communications and Networks. Springer, Cham. https://doi.org/10.1007/978-3-319-71718-0_3

[94] K. Bouraqia, E. Sabir, M. Sadik, and L. Ladid, "Quality of Experience for Streaming Services: Measurements, Challenges and Insights," IEEE Access, vol. 8, pp. 13341–13361, 2020.

Appendix A. Publications and Research Activities

These appendix lists publications that have been writing and presented during the development of this thesis. First, publications related with the main subject of research are listed. Then publications associate with the field of multimedia networking are presented.

Also, in the appendices there are listed other research activities carried out during the development of this thesis, such as: a) contributions in strategic research agendas, b) research fellow visits, c) participation in projects, research networks and strategic platforms; d) multimedia detworking developments.

A.1. List of Publications

Following there is a list of publications that provide most of the content of this thesis dissertation. The following notation is used: B refers to book chapters, C to conference papers and J to journal papers.

Chapter 3

[C.1] T.R. Vargas, J.C. Guerri, S. Sales. Effects of size-based burst assembly algorithm over video traffic transmissions in OBS networks (poster paper). VII Workshop in G/MPLS in the 12th International Conference on optical Network Design and Modeling, ONDM 2008. March 12-14, 2008. Vilanova i la Geltru, Spain.

[C.2] T.R. Vargas, J.C. Guerri, S. Sales. Optimal configuration for size-based burst assembly algorithms at the edge node for video traffic transmissions over OBS networks. 10th Anniversary International Conference on Transparent Optical Networks - ICTON 2008. June 22-26, 2008. Athens, Greece. ISBN: 978-1-4244-2625-6.

[C.3] T.R. Vargas, J.C. Guerri, S. Sales. Effect and Optimization of Burst Assembly Algorithms for Video Traffic Transmissions over OBS Networks. Fifth International Conference on Broadband Communications, Networks, and Systems - Broadnets 2008. International Workshop on Optical Burst/Packet Switching – WOBS 2008. September 8 – 11, 2008. London, UK. ISBN: 978-1-4244-2392-7.

[C.4] T.R. Vargas, J.C. Guerri, S. Sales. Efectos del algoritmo de ensamblado basado en tamaño sobre el tráfico de vídeo en redes OBS. XXIII Simposium Nacional de la Unión Científica Internacional de Radio - URSI 2008. Universidad Complutense de Madrid 22 – 24 de Septiembre. Madrid. España. ISBN 978-84-612-6291-5.

[C.5] T.R. Vargas, J.C. Guerri, S. Sales. Effect and Optimization of Burst Assembly Algorithms to Deal with Edge Congestion in OBS Networks to Improve Video Transport. Workshop Red Temática Española: Redes Multinivel: IP sobre redes de transporte. Universidad Politécnica de Cataluña. 23 y 24 de Febrero, 2009. Vilanova í la Geltrú. España.

[C.6] Kostas Ramantas, Tito Raúl Vargas, Juan Carlos Guerri and Kyriakos Vlachos. A preemtive scheduling scheme for flexible QoS provisioning in OBS networks. Sixth International Conference on Broadband Communications, Networks, and Systems - Broadnets 2009, September 14 – 17, 2009. Madrid, Spain. ISBN 978-963-9799-49-3.

[C.7] T.R. Vargas, J.C. Guerri, S. Sales. Performance evaluation and optimization of OBS networks to provide QoE for video transport. 1er Workshop en Redes Multinivel. Red Temática Española: "Redes Multinivel: IP Sobre redes de transporte" Universitat Politècnica de Catalunya. 14 y 15 de diciembre 2009. Vilanova i la Geltrú, España.

[C.8] T.R. Vargas, K. Ramantas, P. Arce, J.C. Guerri, K. Vlachos, S. Sales. A preemptive scheduling scheme for flexible QoS provisioning in OBS networks – A review. 2do Workshop en Redes Multinivel. Red Temática Española: "Redes Multinivel: IP Sobre redes de transporte" Universitat Politècnica de Catalunya. 14, 15 y 16 de Junio 2010. Vilanova i la Geltrú, España.

[C.9] T.R. Vargas, K. Ramantas, J.C. Guerri, K. Vlachos, S. Sales. Providing QoS to multimedia streams in OBS networks. BONE Summer School 2010. September 6 – 10, 2010. Budapest, Hungary.

[C.10] T.R. Vargas, P. Arce, I. de Fez, V. Murcia, F. Fraile, R. Belda, P. Acelas, J.C. Guerri. Solutions to Improvet the Video Streaming Service over Heterogeneous Networks. Workshop On Future Internet: Efficiency in High-Speed Netwoks. W-FIERRO, Universidad Politénica de Cartagena. Julio 2011. Cartagena, Murcia, España. ISBN: 978-84-96997-69-1.

Chapter 4

[J.1] Tito Raúl Vargas, Juan Carlos Guerri, Pau Arce, DASH Streaming traffic influence over energy efficient ethernet to improve energy savings, Ad Hoc Networks, Volume 136, 2022, 102951, ISSN 1570-8705, https://doi.org/10.1016/j.adhoc.2022.102951.

[**J.2**] Antonio de la Oliva, Tito R. Vargas Hernández, Juan Carlos Guerri, José Alberto Hernández, Pedro Reviriego, "Performance analysis of Energy Efficient Ethernet on video streaming servers", Computer Networks, Available online 11 October 2012, ISSN 1389-1286, 10.1016/j.comnet.2012.09.019.

[**J.3**] Rafael Álvarez, Tito R. Vargas, Juan C. Guerri, "Estudio y análisis del streaming de vídeo adaptativo basado en DASH". 3er Congreso Internacional Control, Instrumentación y Telecomunicaciones 2013, Tunja, Colombia ISSN: 2346-3716.

[**C.11**]T.R. Vargas, J.P. Duarte, G.F. Manotas, L.A. Rodriguez, J.D. Velandia. Video Coding and Streaming Capabilities over IP Networks of a SoC-FPGA ARM-based Embedded System. Congreso Internacional de Ciencias Básicas e Ingeniería. Octubre 2016. Villavicencio. Colombia. ISBN; 978-958-8927-23-7

[**C.12**]T.R. Vargas, J.C. Guerri, P. Arce. Energy Efficient Ethernet study for DASH streams. 18th ACM Symposium on Performance Evaluation of Wireless Ad Hoc, Sensor, & Ubiquitous Networks (PE-WASUN'21), November 22–26, 2021, Alicante, Spain. Submitted

Appendix B. List of related publications

Additionally, the author has participated in the following publications:

[J.4] P. Acelas, P. Arce, R. Belda, W. Castellanos, I. de Fez, F. Fraile, V. Murcia, T.R. Vargas, M. Monfort, J. C. Guerri. An Interactive Wireless-based Telemedicine System for Back Care Applications. Waves 2010 (iTEAM Journal) ISSN: 1889-8297 Volumen: 2 Páginas, inicial: 56 final: 65. España, 2010.

[C.13] Víctor Murcia, Meli Delgado, Tito R. Vargas, Juan C. Guerri, Javier Antich. VAIPA: A Video-Aware Internet Protocol Architecture. IEEE International Conference on High Performance Switching and Routing (HPSR). Universidad Politécnica de Cartagena, Julio 2011. Cartagena, España. ISBN 978-1-4244-1982-1

[C.14] F. Fraile, P. Guzmán, T.R. Vargas, Natalia Flórez, J.C. Guerri. Evaluation of an interactive TV service to reinforce dental hygiene education in children. 9th European Conference on Interactive TV & Video. EuroITV 2011. Universidad Lusófona de Humanidades y Tecnologías. 29 Junio a 1 Julio 2011 Lisboa, Portugal. ISBN 978-1-4503-0602-7 Pag 159 -162 EuroITV 2011 Adjunct Proceedings

[C.15] P. Guzmán, P. Acelas, T. Vargas, P. Arce, J. C. Guerri, E. M. Macias, and A. Suarez, "QoE evaluation and adaptive transport for 3D mobile services," Reunión Red Temática W-FIERRO, Cartagena, Spain, Jul. 2012.

[C.16] Santiago González, Tito Raúl Vargas, Pau Arce, Juan Carlos Guerri. Energy Optimization for Video Monitoring System in Agricultural Areas using Single Board Computer Nodes and Wireless Ad hoc Networks. 2016 XXI Symposium on Signal Processing, Images and Artificial Vision STSIVA 2016, septiembre 2016, Bucaramanga, Colombia. Electronic ISSN: 2329-6259. DOI: 10.1109/STSIVA.2016.7743350.

[C.17] Casares-Giner, V., Inés Navas, T., Smith Flórez, D., & Vargas Hernández, T. R. (2019). End to End Delay and Energy Consumption in a Two Tier Cluster Hierarchical Wireless Sensor Networks. Information, 10(4). https://doi.org/10.3390/info10040135

Appendix C. Strategic Research Agendas contributions

[**B.1**] Red de Colaboración para la I+D+I en Internet del Futuro en Colombia – RECIIF – Plataforma Tecnológica en Internet del Futuro. RECIIF: Future Internet Strategic Research Agenda. Bogotá, Junio 2011.

[**B.2**] Plataforma Tecnológica Española de Fotónica (Fotonica21). FOTÓNICA21: Agenda Estratégica de Investigación 2009. Valencia, Junio de 2009.

Appendix D. Projects, Research Networks, Strategic Platforms

Also, during the elaboration of their thesis, the author has participated in the following projects, network of excellence, technology platforms

[1] Red Temática RiegoNets: Aplicaciones para Comunicación y Control de Redes de Riego sobre Redes y Sistemas de Comunicación Inalámbricos: Red Temática RiegoNets para la Apropiación y uso de TIC en el sector agrícola. Entidad Financiadora: CYTED: PROGRAMA IBEROAMERICANO DE CIENCIA Y TECNOLOGIA PARA EL DESARROLLO Tipo: Convocatoria Externa Iberoamericana. Entidad Titular: Universidad Santo Tomás, Seccional Bucaramanga. Periodo: 2014 – 2018.

[2] Evaluación del Servicio de Video Streaming sobre Redes de Conmutación Óptica de Ráfagas. Entidad Financiadora: Universidad Santo Tomás, Seccional Bucaramanga. Tipo: Convocatoria Interna de Investigación. Entidad Titular: Universidad Santo Tomás, Seccional Bucaramanga. Periodo: 14/06/2011 – 14/10/15.

[3] Future Internet: Eficiencia en las Redes de Altas Prestaciones Entidad Financiadora: Ministerio de Ciencia e Innovación Referencia: TEC2010-12250-E/TEC Tipo: Acciones Complementarias a Proyectos de Investigación Fundamental no Orientada Entidad Titular: Universidad Politécnica de Cartagena. Period: 01/04/2011 a 31/07/2012.

[4] Building the future Optical Network in Europe: The E-Photone/ONE Network. Entidad financiadora: Comisión de las Comunidades Europeas. Referencia: 216863. Tipo: I+D Colaborativa competitiva, Red de Excelencia Europea. Entidad Titular: IBBT (Interdisciplinary Institute for Broadband Technology), Ghent University, others. Periodo: 01/01/2008 a 01/03/2011.

[5] Redes Multinivel: IP sobre Redes de Transporte. Entidad financiadora: Ministerio de Educación y Ciencia. Referencia: TEC2008-02552-E/TEC Tipo: Apoyo competitivo a la transferencia y difusión de tecnología. Entidad titular: Universidad Politécnica de Cataluña. Periodo: 11/04/2007 a 30/09/2010.

[6] Optical Networks: Towards Bandwidth Manageability and Cost Efficiency. Entidad Financiadora: Comisión de las Comunidades Europeas. Referencia: 027497 Tipo: Proyectos europeos del programa Marco, Colaborativa Competitiva, Red de Excelencia Europea. Entidad Titular: Politécnico de Turín, others. Periodo: 01/03/2006 a 01/03/2008.

[7] Plataforma Tecnológica Española de Fotónica (Fotonica21). Entidad Financiadora: Ministerio de Ciencia y Tecnología, Ministerio de Industria, Turismo y Comercio, Unión Europea: Fondo de Desarrollo Regional, Generalitat Valenciana: IMPIVA. Tipo: Plataforma Tecnológica Entidad Titular: Instituto Tecnológico de Óptica y otras entidades del sector de la Fotónica en España. Periodo: 2007 a 2012.

[8] Red de Colaboración para la I+D+i en Internet del Futuro en Colombia (RECIIF), Plataforma Tecnológica en Internet del Futuro. Entidad Financiadora: Unión Europea: First Project under the FP7 Program. Referencia: 248753. Tipo: Plataforma Tecnológica. Entidad Titular: Centro de Investigación de las Telecomunicaciones, CINTEL y otras entidades del sector TIC en Colombia. Periodo: 01/01/2010 a 31/12/2012.

Appendix E. Research Mobility

[**1**] iTEAM Research Institute. Multimedia Communications Group. Universidad Politécnica de Valencia. 08/12/2019 – 25/01/2020

[**2**] Networking technologies and applications Research Group, Department of Telecommunications, Faculty of Electrical Engineering and Computer Science. VSB – Technical University of Ostrava. CZ. 15/10/2016 – 30/11/2016

[**3**] iTEAM Research Institute. Multimedia Communications Group. Universidad Politécnica de Valencia. 18/09/2015 – 20/10/2015

[**4**] iTEAM Research Institute. Multimedia Communications Group. Universidad Politécnica de Valencia. 01/10/2010 – 31/07/2012

[**5**] Photonic Networks Laboratory. University of Essex, UK. BONE Mobility Action. November 16 to December 20, 2008.

[**6**] Research Academic Computer Technology Institute. RACTI. University of Patras, Greece. BONE Mobility Action. 05/26 to 06/20, 2008

Appendix F. Multimedia Networking Developments

[**1**] USTAView – Software Registration. Digital Signage Web Application for Stmart TVs. Universidad Santo Tomás Bucaramanga, 2016

[**2**] Streamer TeleVGA DE0 NANO SOC FPGA – HPS ARM. Communication Module to capture, codification, encryption and transmission of a VGA source from a Vital Signals Monitor. Fundación Cardiovascular de Colombia, Universidad Santo Tomás Bucaramanga. 2017.

[**3**] USTAView EndPoint for Non Smart TVs. Digital Signage Project. Universidad Santo Tomás Bucaramanga. 2017.