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Smart Client-Server Protocol and Architecture for Adaptive Multimedia Streaming

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Abstract

In recent years, multimedia services consumption has increased and it is expected that this trend will continue in the near future, becoming the evaluation of Quality of Experience (QoE) as a very important issue for assessing the quality of providers' services. In this sense, the optimization of the QoE is progressively receiving much attention considering that current solutions are not based on the adaptation, feasibility, cost-effectiveness and reliability.

The present dissertation is focused on the characterization, design, development and evaluation of different multimedia applications aimed to optimize the QoE.

Therefore, this work investigates the influence that the networks infrastructure, the videos' characteristics and the users' terminals present on QoE of the current Internet multimedia services. The work is based on a comprehensive research of subjective and objective assessments in heterogeneous networks. Challenges and research questions related to the state of the art are discussed in this dissertation.

In the first phase of this dissertation, we design a test methodology for assessing QoE of live video streaming and video on demand platforms to be transmitted over Wi-Fi and cellular networks. From this initial step, we will propound the related research issues and questions to solve along this dissertation. Our methodology considers the use of subjective and objective metrics to evaluate the QoE perceived by end-users. A set of laboratory experiments is conducted where our proposed methodology is applied. The obtained results are gathered and analyzed to extract the relations between Quality of Service (QoS) and QoE. From the results, we propose a QoS-QoE mapping which allows predicting QoE.

In the next phase of the research, we develop QoE-optimization algorithms based on network system management for Wi-Fi and cellular networks. The algorithms use the key parameters that were taken into account for QoE assessment. The goal of these algorithms is to provide a flexible management system for the networks in order to achieve the desirable trade-off between QoE maximization and resource usage efficiency.

Lastly, the system testbed is designed in order to evaluate the performance of generic multimedia services applications for the different environments under test. The system testbed is based on virtualization approach; it uses the shared resources of a physical hardware to virtualize all components. The virtualized testbed provides virtualized network functions for the different scenarios such as the Internet (Content Delivery Networks - CDNs) and wireless networks. Therefore, lightweight protocols and agile mechanisms are adopted in the system to provide enhanced service to end-users. The QoE results are reported to the service providers according to the parameters defined in the evaluation process. As a result, we have obtained a cost-effective system, which is considered as a feasible way for test evaluation.

Resumen

En los últimos años, el consumo de servicios multimedia ha aumentado y se prevé que esta tendencia continúe en un futuro próximo, convirtiendo el tema de la evaluación de la Calidad de la Experiencia (QoE) en un tema muy importante para valorar el servicio de los proveedores. En este sentido, la optimización de la QoE recibe cada vez más atención ya que las soluciones actuales no han tenido en cuenta, la adaptación, la viabilidad, la rentabilidad y la fiabilidad.

La presente memoria se centra en la caracterización, diseño, desarrollo y evaluación de diferentes aplicaciones multimedia, con el fin de optimizar la QoE.

Por tanto, este trabajo investiga la influencia que la infraestructura de redes, las características de los videos y los terminales de los usuarios, presentan en la QoE de los servicios multimedia actuales en Internet. Esta tesis se basa en la investigación exhaustiva de la evaluación subjetiva y objetiva de QoE en redes heterogéneas. Los desafíos y cuestiones relacionados con el estado de la técnica y se discuten en esta disertación.

En la primera fase, diseñamos una metodología de prueba para evaluar la QoE en la transmisión de video en directo y a través de plataformas de video bajo demanda en redes Wi-Fi y celulares. A partir de esta fase inicial, propondremos los problemas a investigar y las preguntas para resolver a lo largo de esta disertación. Nuestra metodología hace uso de métricas subjetivas y objetivas para evaluar la QoE percibida por los usuarios finales. Se realiza un conjunto de experimentos en laboratorio donde nuestra metodología de pruebas es aplicada. Los resultados obtenidos se recopilan y analizan para extraer las relaciones entre la Calidad de servicio (QoS) y QoE. A partir de estos resultados, se propone un mapeo de QoS-QoE que permite predecir la QoE.

En la siguiente fase de la investigación, desarrollamos los algoritmos de optimización de QoE basados en la administración del sistema de red para redes Wi-Fi y celulares. Los algoritmos usan los parámetros clave que se tuvieron en cuenta para la evaluación de QoE. El objetivo de estos algoritmos es proporcionar un sistema de gestión flexible para las redes con el ob-

jetivo de lograr un equilibrio controlado entre la maximización de QoE y la eficiencia del uso de los recursos.

Por último, se diseña el banco de pruebas del sistema para evaluar el rendimiento de las aplicaciones de servicios multimedia genéricos en los diferentes entornos de prueba. El banco de pruebas del sistema se basa en el enfoque de virtualización; usa los recursos compartidos de un hardware físico para virtualizar todos los componentes. El banco de pruebas virtualizado proporciona funciones de red virtualizadas para diferentes escenarios, como Internet (las redes de distribución de contenido - CDNs) y redes inalámbricas. Por lo tanto, se adoptan protocolos livianos y mecanismos ágiles en el sistema, para proporcionar un mejor servicio a los usuarios finales. Los resultados de QoE son proporcionados a los proveedores de servicios de acuerdo con los parámetros que se definen en el proceso de la evaluación. Como resultado hemos obtenido un sistema que presenta un servicio rentable como una forma factible para la evaluación de la prueba.

Resum

En els últims anys, el consum de serveis multimèdia ha augmentat i es preveu que aquesta tendència continue en un futur pròxim, convertitnt el tema de l'avaluació de la Qualitat d'Experiència (QoE) una tasca molt important per a valorar el servei dels proveïdors. En aquest sentit, l'optimització de la QoE rep cada vegada més atenció degut a que les solucions actuals no tenen en compte, l'adaptació, la viabilitat, el rendiment i la fiabilitat.

La present memòria se centra en la caracterització, disseny, desenvolupament i avaluació de diferents aplicacions multimèdia, amb la finalitat d'optimitzar la QoE. Per tant, aquest treball investiga la influència que la infraestructura de les xarxes, les característiques dels videos i els terminals dels usuaris tenen sobre la QoE dels serveis multimèdia actuals d'Iinternet. Aquesta tesi es basa en una recerca exhaustiva de l'avaluació subjectiva i objectiva de QoE en xarxes heterogènies. Els desafiaments i preguntes relacionats amb l'estat de la tècnica es discuteixen en aquesta dissertació.

En la primera fase, dissenyem la metodologia de prova per a avaluar la QoE de transmissió de video en directe i de plataformes de video baix demanda en xarxes Wi-Fi i cel·lulars. A partir d'aquest primer pas, proposem els problemes de recerca relacionats i les preguntes a resoldre a través d'aquesta tesi. La nostra metodologia fa ús de mètriques subjectives i objectives per a avaluar la QoE dels usuaris finals. Es realitzen un conjunt d'experiments en laboratori on s'aplica la nostra metodología. Els resultats obtinguts es recopilen i analitzen per a extraure les relacions entre la QoS i la QoE. A partir d'aquests resultats, esproposa un mapatge de QoS-QoE que ens permetrà predir la QoE.

En la següent fase de la recerca, desenvolupem els algoritmes d'optimització de la QoE per a l'administració de xarxes Wi-Fi i cel·lulars. Els algoritmes utilitzen els paràmetres clau que es van tenir en compte per a l'avaluació de QoE. L'objectiu d'aquests algoritmes és proporcionar un sistema de gestió flexible per ales xarxes que permetrá aconseguir un equilibri controlat entre la maximització de la QoE i l'us eficient dels recursos.

Finalment, el banc de proves del sistema està dissenyat per a avaluar el rendiment de les aplicacions de serveis multimèdia genèrics en els diferents

entorns de prova. El banc de proves del sistema es basa en l'enfocament de virtualització; usa els recursos compartits d'un equip físic que virtualitza tots els components. El banc de proves virtualitzat proporciona les funcions de xarxa virtualitzades per a diferents escenaris, com Internet (les xarxes de distribució de continguts - CDNs) i xarxes sense fils. Per tant, s'adopten protocols lleugers i mecanismes àgils en el sistema per a proporcionar un millor servei als usuaris finals. Els resultats de QoE son proporcionats als proveïdors de serveis d'acord amb els paràmetres que es defineixen en el procés de l'avaluació. Com a resultat, hem obtés un sistema que presenta un servei rendible i com a viable per a l'avaluació de la prova.

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Abbreviations

3GPP	3rd Generation Partnership Project
4G	Fourth Generation Wireless Systems
5G	Fifth Generation Wireless Systems
ACR	Absolute Category Rating
AL	Adaptation Logic
ANOVA	ANalysis Of VAriance
AVC	Advanced Video Coding
CBQ	Class-Based Queueing
CBR	Constant Bit Rate
C-ITS	Cooperative Intelligent Transportation Systems
CDN	Content Delivery Network
CSP	Content Service Providers
DHCP	Dynamic Host Configuration Protocol
DiffServ	Differentiated Services
DPI	Deep Packet Inspection
DSCP	DiffServ Code Point
DSSS	Direct Sequence Spread Spectrum
DWFQ	Dynamic Weighted Fair Queueing
DASH	Dynamic Adaptive Streaming over HTTP
DCR	Degradation Category Rating
DSIS	Double Stimulus Impairment Scale
DVB	Digital Video Broadcasting
ETSI	European Telecommunications Standards
FIFO	First-In First-Out
GPRS	General Packet Radio Service
GOP	Group of Pictures
HAS	HTTP Adaptive Streaming
HDS	HTTP Dynamic Streaming
HDTV	High Definition Television
HLS	HTTP Live Streaming
HTTP	Hypertext Transfer Protocol
HVS	Human Visual System
IANA	Internet Assigned Numbers Authority
ICMP	Internet Control Message Protocol
IETF	Internet Engineering Task Force
ICT	Information and Communication Technologies

IEEE	Institute of Electrical and Electronics Engineers
IntServ	Integrated Services
IoT	Internet of Things
IP	Internet Protocol
IPv6	Internet Protocol version 4
IPv6	Internet Protocol version 6
IPTV	Television over Internet Protocol
ISP	Internet Service Provider
KQI	Key Quality Indicators
LAN	Local Area Network
LTE	Long Term Evolution
LTE-A	Long Term Evolution - Advanced
LTE-M	Long Term Evolution - Machine-to-Machine
MOS	Mean Opinion Score
MPEG	The Moving Picture Experts Group
MPD	Media Presentation Description
MPEG2TS	MPEG-2 Transport Stream
MSS	Microsoft's Silverlight Smooth
NAT	Network Address Translation
NFV	Network Functions Virtualization
NGN	Next Generation Networking
NS	Network Simulator
OTT	Over The Top
PSQA	Pseudo-Subjective Quality Assessment
QoE	Quality of Experience
QoS	Quality of Service
QP	Quantization Parameter
RTT	Round Trip Time
RTP	Real-Time Transport Protocol
RTSP	Real-Time Streaming Protocol
SDN	Software Defined Networking
SLA	Service Level Agreement
SI	Spatial Information
SRC	Source video
SS	Single Stimulus
SSCQE	Single Stimulus Continuous Quality Evaluation
SVC	Scalable Video Coding
TCP	Transmission Control Protocol
TOS	Type Of Service
TI	Temporal Information
UDP	User Datagram Protocol
URL	Uniform Resource Locator

VLC	VideoLAN Client
VM	Virtual Machine
VNX	Virtual network over Linux
VQEG	Video Quality Expert Group
VoD	Video-on-Demand
WiMAX	Wireless Interoperability for Microwave Access
Wi-Fi	Wireless Fidelity
WLAN	Wireless Local Area Network

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Chapter1. Introduction

1.1 Preamble

Nowadays, most of the traditional websites have changed their way of viewing to multimedia service, due to the increased use of smart devices where users make video consumption in academic training (YouTube), social networks (Facebook, Instagram) and the world of entertainment such as online movies (Netflix, Hulu) and video games (league of legends). Therefore, the number of devices grows more and more and many people often consume the same video simultaneously, just as the growth in the number of smart devices exceeds the munificent population.

On the Internet, multimedia services and applications have become an important source of income for the network operators and service providers. The service providers are in a highly competitive market with each other to give strongly user's satisfaction where user expectations are high. Specifically, paying customers to expect their viewing experience to the same across all viewing devices and independently of their used Internet access. Nevertheless, the customers' devices accessed to multimedia services on the Internet through the heterogeneous network environments. When there are insufficient network resources to maintain video streaming session at the desired quality level for each session, which may result in degradation of the quality for one or more users. Expressly in wireless scenarios, the mobility of users and insufficient signal strength can cause in a very poor transport service performance in terms of Quality of Service (QoS) delays, jitters, low and varying bandwidth, etc. For this, the customers' demand needs evaluation by the Internet services. As a result, Quality of Experience (QoE) is introduced as important terminology to describe the user's satisfaction with Internet service. In return, the customers are paying for the expectation of the service.

Nokia in 2005 [1] introduced this concept as a perception of the end users about a service quality and it stated as *"QoE is how a user perceives the usability of a service when in use how satisfied he or she is with a service."* The Broadband forum in 2006 in its technical report, TR-126 [2] defined QoE as a measure and an indicator of a system in fulfilling the requirements of the customers. According to the ITU-T focus Group on IPTV [3], QoE refers to *"the overall acceptability of an application or service, as perceived* *subjectively by the end- user."* Within the context of the COST Action Qualinet [4] defines QoE as *"The degree of delight or annoyance of the user of an application or service. It results from the fulfillment of his or her expectations with respect to the utility and/or enjoyment of the application or service in the light of the user's personality and current state."* QoE: *"Degree of light of the user of a service. In the context of communication service, it is influenced by content, network, device, application, user expectations and goals, and context of user."*

Generally, two approaches are using to assess the service quality of multimedia providers: the subjective and objective methods. The subjective method is proposed by the International Telecommunication Union Telecommunication (ITU-T) [5], which is used to find out the users' perception of the quality of video streaming. The Mean Opinion Score (MOS) is a metric example of the subjective measurement method in which users rate the video quality by giving five different level scores from 5 to 1, where 5 is the best and 1 is the worst quality. However, the objective method uses different models of human expectations and tries to estimate the performance of a video service in an automated way, without involving the humans. The subjective measure can vary according to the user expectation and context.

Moreover, end-to-end system effects on the accuracy of the measure of QoE according to client's behavior, terminal characteristic, network, services infrastructure, media encoding, etc. QoE based on subjective measure requires tests with human participation in controlled or uncontrolled environments in order to properly evaluate the QoE, which is costly and time-consuming. The objective measure calculates perceived quality by the mathematical model. Using objective measure introduces infidelity of results since the complexity of mathematical approaches presents highly degrade result, which is considered in the high-definition of multimedia streaming services [6.] Multimedia service providers and Internet service providers require efficient tools in order to monitor, assess and estimate the QoE of the high and ultra-definition quality such as HD, 2K, 4K, and 8K with reasonable accuracy. This is because the QoE of the high-resolution videos are very sensitive to degrade in heterogeneous networks than standard definitions. Therefore, based assessing, multimedia providers approach to manage the QoE of their subscribers to attract expectation of using services.

1.2 Motivation

The motivation of this dissertation comes from answering the more consequential research study questions.

Multimedia streaming services over Internet grows steadily because of its consumption and its adoption. According to recent reports of [7, 8] as depicted in Figure 1.1 and Figure 1.2. The Internet traffic increases very fast and the traffic of multimedia streaming services is massive traffic, which generates high percentage traffic over the telecommunications network devices. The Internet video to Television (TV) grew 50 percent in 2016 and the amount of Video-on-Demand (VoD) traffic in 2021 will be equivalent to 7.2 billion DVDs per month. Therefore, Live Internet video will account for 13 percent of Internet video traffic by 2021. This growth has created a threat to network resources as well as it opens opportunities to network operators and service providers for revenue generation. Therefore, Content Delivery Network (CDN) [7], as the content distributor, its traffic will carry 71 percent of all Internet traffic by 2021. As a result, in the recent years, multimedia traffic has diverted the attention in both research areas and industrial communities.

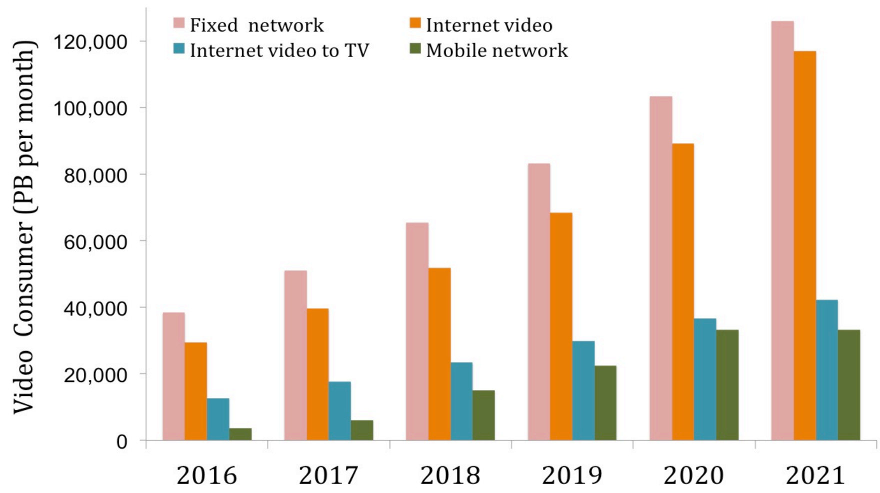


Figure 1.1. Internet video streaming traffic (data from [7]).

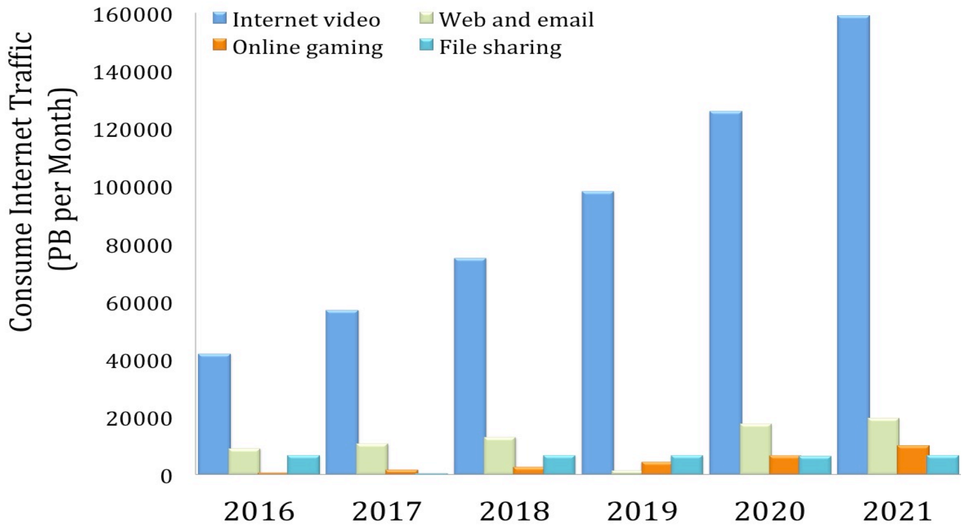


Figure 1.2. Comparison of different services (data from [8]).

In multimedia streaming, QoE monitoring and evaluation are a multiple field research topic based on social psychology, cognitive science, engineering science, economics, etc. The QoE depends on different elements (e.g., media characteristic, network service, clients' application, business model, ubiquitous users access, users' behaviors, etc.) that directly or indirectly affect the user's perception towards the multimedia service [9]. Moreover, the diversity of these elements drives the QoE assessment rather complex and unpredictable. This motivation of this dissertation addresses some prominent challenges related to QoE monitoring and assessment and management as described next:

- **QoE based on subjective method**: QoE is a subjective method as it reflects the perception of the users to a service. User attitude and expectation towards multimedia services play a vital role in determining the QoE. Moreover, the QoE can depend on different user profiles like age, sex, interest, skills, frame of mind, experience, etc. Different environmental conditions impact how users perceive the multimedia service content. Therefore, Multimedia QoE can vary according to novelty assessment and service.
- **Subjective vs. objective evaluation**: The Multimedia QoE can be evaluated based on the methodologies of subjective and objective. The subjective evaluation requires human participation to perform the tests. This method cannot always be applied in real-time and

requires more time and cost (due to human participation). On the other hand, the objective evaluation based on objectively measuring network, video and device capacity is a complex task to the video quality, which considered the original video.

- **QoS impact on multimedia QoE:** The ability to identify the scale of perceived quality due to network impairment is a key point for multimedia streaming service. The impact of QoS impairment and ubiquitous network user's interface affect on QoE of multimedia streaming service. Therefore, extract specific factors in QoS, which involved reducing the degree of QoE, is a complex task.
- **Quantity users and heterogeneity of multimedia technique impacts on multimedia QoE:** QoE linked to end user’s perception, high compete of users to request same video points issue of QoE assessment. i.e. in adaptive video streaming technology [10], quality assessment among competitive users becomes unpredictably degraded, when the users connected to same shared service network point. The result of evaluating this environment becomes imprecise [11].

1.3 Objectives

The contribution and methodology of dissertation focus on QoE assessment and management for different technologies of multimedia streaming. The main objectives cover the research approaches have been considered to identifying different research areas of multimedia streaming, such as live video streaming and on-demand adaptive streaming, in heterogeneous networks such as Wi-Fi and cellular networks. As a general objective, we aim to develop new approaches and systems that should meet the expectations of assessing and managing the QoE of multimedia services.

The specific objectives of this dissertation are focused on the next objectives:

Objective 1: Create an intensive comprehensive survey of the state-of-the-art concerning QoE of multimedia services such as live streaming and on-demand streaming. Indicate the potential ways to discuss the exiting challenges in these regards.

Objective 2: Investigation and identification of the key factor parameters that are influencing the perceptual quality of real-time video streaming and on-demand adaptive videos streaming for the current service applications.

Objective 3: Improve the subjective test methodology for the QoE evaluation. Take a laboratory-controlled approach to collect the subjective data set with respect different parameters.

Objective 4: Design a smart system to monitor and evaluate the QoE, and correlation between QoS and QoE, we study machine learning and investigate on which approach should be selected to include it.

Objective 5: Select the appropriate protocols that can be used to distribute multimedia content across Internet to provide the best possible QoE.

Objective 6: Develop algorithms to provide QoE optimization based on resource allocation and selection in heterogeneous networks**.**

Objective 7: Develop cost-effective virtualized system testbed for evaluating QoE in real time. Subjective and objective metrics will be included to evaluate automatically QoE in the system.

1.4 Preceding projects

Currently, there are many works toward QoE assessment. Some of them have been designed for researching investigations and other designed by private companies. Generally, the QoE system is aimed to take the correct decisions to improve the video quality of end-users. Therefore, research projects are approached to estimate correctly the QoE of end-users. HTTP adaptive video streaming is a client-aware adaptation to adapt the quality of video according to the clients' network availability. There is bare research on assessment of end-users QoE according to assessment methodology and correlation between subjective and objective to predict the QoE. Moreover, the systems are designed to evaluate the performance of multimedia services still they have restrictions on new technologies and costly to experiments.

However, there are also dissertations that studied assessment and management of quality of user experience, which applied for multimedia services. Some finished PhD thesis related with the dissertation is listed below:

- PhD thesis: Alejandro Canovas Solbes, “Diseño y Desarrollo de un Sistema de Gestión Inteligente de QoE para Redes HD y Estereoscópicas IPTV.”, Universidad Politecnica de Valencia. [12]
- PhD thesis: Juan Ramón Diaz Santos, “Design and Implementation of a Communication Protocol to Improve Multimedia QoS and QoE in Wireless Ad Hoc Networks.”, Universidad Politecnica de Valencia. [13]

1.5 Dissertation structure

After introducing the main issues that have motivated to write this thesis, as well as a description of the main objectives pursued, the rest of the structure of the thesis is organized as follows.

Chapter 2 reviews the general literature and related works in relation to this dissertation. It gives a deep explanation to understand the matters researched in the dissertation. The chapter divides into nine sections. An introduction presents briefly on the chapter's content. Classification of the multimedia stream on the Internet describes according to technologies and its types. Then, the aspect of identified factors influencing the QoE of multimedia services is explained in details. Afterward, assessment test methods and models are explained linking to research studies of QoE. Moreover, QoE optimization methods are explained in heterogeneous networks by managing the network resources in the areas of network communication services. The implementation of test environment describes according to the approaches and requirements taking to measure the QoE of multimedia services. Thus, the existing challenges in the current QoE assessment discussed in the last section.

Chapter 3 presents a case study of the significant impact of network behavior on QoE of live video streaming. Assessment method based on subjective and objective is used to evaluate QoE of the end-users.

The purpose of the method defines the type of video artifacts have occurred in video transmission in the network infrastructure. Following this, the algorithm provides aware detection of the video artifacts, which is related to the QoE.

Chapter 4 presents the description of the subjective and objective test methodology for video multicasting service in the wireless networks. The evaluation of QoE tests considers impacting of factors of QoS, video characteristic, and device capacity. Subjective and objective metrics use to evaluate the service. The evaluation results store into a data set. In order to provide mapping between QoE and QoS, different machine learning models use to train the data set. The accurate and fast approach selects to predict the QoE. From this, the algorithm is based on smart QoE evaluation for multicast system proposes in order to provide better QoE.

Chapter 5 shows the description of the subjective user studies for HTTP Adaptive Streaming (HAS). The subjective test methodology develops to evaluate QoE. Different metrics use in the test methodology in order to provide the requirement of the methodology. Subjective result is collected into a data set for assessing the user's QoE. Objective metrics also present to evaluate QoE. Following this, the statistical correlation presents between the restriction parameter value of the QoS, media characteristic and subjective assessment in order to estimate QoE in the HAS clients.

Chapter 6 includes QoE optimization for live video streaming and on-demand adaptive streaming in heterogeneous networks. It is based on SDN. The advantage of Software Network Define (SDN) takes into consideration to observe the activity of the concurrent users when they consume the multimedia services. The management algorithm is developed to allow service providers to ensure fairness and stable service to the users. Therefore, in cellular networks, the developed algorithm is enabled to select the best network resource to provide better QoE in the handover process.

Chapter 7 presents a virtualized testbed design for the experimental study of generic multimedia streaming services. The system provides different network infrastructure to evaluate the QoE. Algorithms are developed to give different functions and mechanisms in the system, which are provided better service to users. Therefore, the QoE evaluations are found on clients' side and they report to the service provider in order to find out the expectation of users according to the consumption of the video service.

Finally, Chapter 8 draws our conclusion and contribution, and the future research will be outlined. It also includes the list of publications derived from the PhD thesis.

Chapter 2. Background and state of the art of Multimedia QoE

2.1 Introduction

Nowadays there is a facility to watch online video streaming on the Internet. The remarkable growth of video-enabled electronic devices such as personal computer (PC), smart devices, tablets, and Internet-enabled television and the high-speed Internet like different ubiquitous network access (Wi-Fi, 3G, 4G and 5G) are key factors to the growing multimedia resolution and quality (2k, 4k, 8k) than standard-definition.

The technology of video streaming is highly developed and it is necessary to guarantee to deliver high quality video streaming over heterogeneous networks to the users. The service providers are interested in analyzing and assessment video streaming services thoroughly in order to out the degree on influence of technical and non-technical parameters on user satisfaction. This chapter presents a background on multimedia streaming technology and its developments on the Internet. The general literature and related works in relation to this dissertation will review to give a deep explanation to understand the matters. Finally, the challenges will specify according to limitation still exist according to QoE.

2.2 Streaming in Internet

The rapid advances of network topologies brought some of the broadcast television services to the proprietary Internet Protocol (IP) delivery networks, and the IP Television (IPTV) [14]. In fact, IPTV has been considered as an evolution from traditional television broadcasting rather than a revolution [15], increasing the user experience with means of interaction. Users are allowed to choose which content to watch among a pool of content service given by the Internet Service Providers (ISP). In addition, IPTV offers the ability of stream the multimedia content directly from the source so that the client media player can begin playing the multimedia file before the entire file has been transmitted. This approach allows for the transmission and rendering of live content. Over The Top (OTT) technology is another development in the media service industry, which uses the Internet for multimedia service delivering. In spite of that, there are also several differences between both technologies. Mainly IPTV is run over managed networks. The ISPs have their own network infrastructure because of using multicast transport, and the technology can control and offer high priority of QoS

[16]. Whereas in OTT, the video content is streamed to the end-users over a unicast connection from a server or one of a server of Content Delivery Networks (CDN), through the using unmanaged networks, for this reason, the QoS cannot be controlled or guaranteed. Service providers of OTT usually use either their own proprietary streaming protocols running on top of an existing transport protocol and being completely dependent on the underlying best effort network, therefore the benchmark comparison between OTT and IPTV technologies according to [17][18][19][20][21] are explained in Table 2.1.

Table 2.1. Comparison between OTT and IPTV.

Comparison Category	OTT (Over the Top)	IPTV (Internet Protocol TV)
Content Delivery	Uses open internet, un-managed network. Open ecosystem	Uses dedicated, managed network. Walled garden ecosystem
Network Type	Delivered from content provider / aggregator to the viewer using open network. Usage of CDN	Closed, proprietary network, accessed via a specific internet service provider
Network Relationship	Without the need for intervening carriage negotiations, or infrastructure investments	Services are delivered on optimized and custom high bandwidth network
Quality of Service (QOS)	Not guaranteed, works under best effort conditions	High quality, reliable network with control over quality of services
Service Examples	Popular Video on Demand services like YouTube, Netflix, Amazon LoveFilm, Hulu, Sky Go, BBC iPlayer etc.	IPTV services like U-Verse (AT&T), Prism TV (CenturyLink)
Delivery Protocol	Delivered over HTTP / TCP, a connected transport protocol. Movement towards adaptive streaming technologies HLS (Apple), Smooth Streaming (MS) and HDS (Adobe)	IPTV uses Transport Stream (TS) transmission technology. Uses RTP (Real time protocol) over UDP, a connectionless protocol

Comparison Category	OTT (Over the Top)	IPTV (Internet Protocol TV)
Major Platform Players	OVP (Online Video Platforms) like Kaltura, Brightcove, CDN Players like Akamai, L3, Limelight, Cloud Service Providers like Amazon	TSP (Telecom Service Providers) and IPTV platform vendors - Microsoft Mediaroom (now Ericsson) , ALU, Cisco
Key Challenges	Low quality of service, absence of live broadcast, non premium content, unicast delivery model	Expensive, Heavy investment in Bandwidth and infrastructure
Key Benefits	Low cost, flexible model, Easy to manage and operate	High quality of service and quality of experience. Monitoring and control, interactive services
Content Type	Typically not premium in nature due to security, absence of DRM	Premium content
Routing Topology	Unicast (HTTP), Simulated Multicast (UDP/TCP)	Multicast. Initial unicast burst during channel change leading to Multicast

2.2.1 Multimedia streaming techniques

Streaming multimedia is the transmission of data from a server to one or several clients. The client's playback is started by a few seconds after it begins receiving the data from the server. There are many providers of streaming services, which are typically run the best effort over the Internet. To cope with varying network conditions and at the same time being able to provide a good service, several steaming approaches have been provided. There are three main methods used today for streaming the multimedia: traditional streaming, progressive download and adaptive streaming, these approaches are described as follows.

2.2.1.1 Traditional streaming

In traditional streaming approach [22], the stateful protocols are employed. Being stateful means that the server keeps tracking the client's state from the time they get connected to each other until the connection is terminated. The client communicates its state to the server by issuing its commands such as Play, Pause, Fast-Forward or Teardown. These protocols are also called push-based protocols as they push the data toward the client. Real-Time Transport Protocol (RTP) and Real-Time Streaming Protocol (RTSP) are typically used to implement such services. RTP operates over UDP and is suitable for multicast distribution. These protocols are widely used in IPTV technology. Using the scaling advantage of multicast distribution, ISPs can control the amount of traffic that they allow in their network. Nevertheless, RTP neither provides a mechanism to ensure timely delivery nor guarantees the provided QoE. Additionally, there is no flow control or congestion avoidance provided by the protocol itself, rather these are up to the application to implement. RTSP is useful for establishing and controlling the media sessions between the end-points, but not being responsible for the transmission of media data. Instead, it relies on RTP-based delivery mechanisms. The packets in RTSP can be transmitted over either UDP or TCP transports. When firewalls or proxies block UDP packets, there is an increase of latency. Some other important points about traditional streaming are as follows.

- The streaming server sends the data packets to the client at a real-time rate only.
- The server only sends ahead enough data packets to fill the client buffer. The client buffer is typically between 1 and 10 sec. This means that if the user paused the streamed video and waits for a couple of minutes, still only less than 10 sec of video will have downloaded to the client in that time.
- As examples of traditional streaming, Adobe Flash Player and Apple QuickTime

2.2.1.2 Progressive download

Over the past several years, the streaming media industry has had a steady shift away from traditional streaming protocols back to plain HTTP progressive download [22]. This has been mainly due to the following reasons. HTTP runs over TCP port 80 and will not have any firewall blocking problems on intermediate network nodes, because the Internet is basically built and optimized for HTTP delivery, therefore, no special proxies and caches are required for that. A media file is just like any other file to a Web cache. TCP delivers most part of the Internet traffic and is able to guarantee the stability of the network by means of a congestion control algorithm [23]. It is much cheaper to move HTTP data to the edge network as most network nodes support HTTP and no need for specialized servers. Apart from these, HTTP is a stateless pull-based protocol so that the streaming logic is more on the client rather the server. This will lead to a more scalable system compared to the case when using traditional, stateful streaming protocols. In addition, thanks to TCP reliable delivery, there will be no image distortions due to the insufficient network. Other differences between the HTTP/TCP based streaming and the traditional methods are presented in Table 2.2.

Table 2.2: Comparison between HTTP and RTP.

Category	**RTP/UDP**	**HTTP/TCP**
Supported technology	Unicast, multicast	Unicast
Content Source	Live, pre-encode	Live, pre-encode
Service	IPTV	OTT
Results of insufficient bandwidth	Packet loss, Stalling	Stalling
Delay packets	Low	Medium to high
Session management	Server, Client	Client
Firewall/NAT friendly	No	Yes
Congestion control	No	Yes

Progressive download is a simple file download from a HTTP Web server. The term "progressive" arises from the fact that clients allow the media file to be played back while the download is still in progress. Unlike traditional streaming servers that merely send more than several seconds of media data to the client at a time, HTTP Web server keeps the data flowing until the download file is complete. Therefore, if the user pauses the downloaded video at the beginning of playback and then waits, the entire video will be eventually downloaded to user's browser cache allowing to smoothly playback the entire video without any interruption. Therefore, protocols built on top of TCP, sending the content using HTTP can only be over a unicast connection [24]. Progressive download has some disadvantages as well; 1) there is no bitrate adaption since an ordinary HTTP server is unaware of the content and treats the media bit stream equal to other files. Accordingly, the media content is delivered using best effort with respect to available resources, 2) trick modes such as Fast-forward, Seek/Play or Rewind, are often limited or unavailable. There is a waste of the bandwidth if the user decides to stop watching the video content, since the video's data, which is not going to be played, has been already transferred and buffered [25].

2.2.1.3 Adaptive streaming

There are a number of real-world scenarios in which the properties of a communication link are fluctuating when serving a certain multimedia service. Such changes can typically appear when communicating through a best effort network where the networking infrastructure is not under the management of an operator from end-to-end, and thus its performance cannot be guaranteed. Another example is the reception of multimedia content through mobile high-speed Internet connections like WLAN/3G/4G/5G, where the channel conditions are changing over the time, due to fading, interferences, noise, massively connected users to the connection point or due to the user mobility (handover). These network issues decrease the throughput and introduce high delays at the application layer. As a consequence, the playout buffer fills more slowly or even depletes. If the buffer is empty, the playback of the video has to be interrupted until receiving enough data for the playback continuation. These interruptions are denoted as stalling or rebuffing, which have a significant effect on QoE of the end-users [26, 27]. Although using basic progressive HTTP download avoids packet loss because of the TCP reliable delivery attribute, it cannot avoid stalling or rebuffering degradations at insufficient bandwidth conditions. To cope with this problem, adaptive streaming techniques have been proposed to provide the best possible quality to the user by adjusting the presented quality to the current conditions including network's conditions, available bandwidth and buffer status, user's device capability and CPU capacity, etc. The techniques are used to adapt the video source bitrate to the current condition can be classified into three categories [28]; transcoding-based, scalable encoding-based and stream-switching technique as outline next.

Transcoding- based adaptation

This approach consists in adapting the video content to match a specific bitrate by means of on-the-fly transcoding of the raw content It can achieve a very fine granularity by throttling frame rate, compression, and video resolution, but also has a very high cost due to transcoding the raw video content several times for each quality request. As a result, scalability decreases since transcoding needs to be performed for every different client's available bandwidth. In addition, due to the computational requirements of a real-time transcoding system, the encoding process is required to be performed in appropriate servers [29].

Scalable encoding- based adaptation

Another important class of adaptation algorithm is employing Scalable Video Coding (SVC) as an alternation of H264/MPEG-4 AVC [30, 31]. SVC provides three scalability options: Spatial scalability, which allows for switching to different resolutions, temporal scalability, which enables the adaptation of the frame rate, and encoding scalability, which allows adaptation of quality of the content. In SVC encoding, the base layer provides the lowest level of quality in one or more of the aforementioned scalable quality parameters while each enhancement layer on top of it provides a quality improvement for those parameters. All enhancement layers depend on the base layer on the previous enhancement layer of the same scalability dimension. In order to switch to a higher layer, only the missing difference data have to be transmitted and added. This is the major difference to adaptation with single-layer codecs like AVC, that quality can be increased incrementally only using enhancement layers, rather than downloading a higher quality bit stream and discarding the already downloaded lower quality stream. In fact, this is the key advantage of SVC to distribute information among various layers with minimal added redundancy. In other words, while a stream that is traditionally encoded at different quality levels has significant redundancy between the different encoding layers, each layer in an SVC-encoded stream has minimal common information between the layers. This makes SVC efficient for media storage at various quality levels. Further, SVC, allows more download flexibility since already downloaded parts of the video clip can be enhanced at a later time. Nevertheless, there is also a trade-off in regard to overhead introduced by multi-layer codecs. This means, that, overall, SVC files of a video content of a certain bitrate, including a base layer and enhancement layer(s), are larger compared to an AVC file of the same video and same bitrate. In addition, SVC streams are typically more complex to generate and impose codec restrictions. Thus, the adaption rate for SVC could be slower.

Stream-switching technique

The stream switching approach encodes the raw video content at several different increasing bitrates using single-layer codecs like AVC, and generates different versions of the same content. An algorithm must dynamically choose the video level, which matches the user's available bandwidth. When changes in the available bandwidth occur, the algorithm simply switches to different levels to ensure a continuous playback. The main purpose of this method is to minimize processing costs since no further processing is needed once all quality levels are generated. In addition, this approach is completely codec agnostic, this means it does not require a specific codec format to be implemented. In contrast, storage and transmission requirements must be considered as well (because of encoding the same video content different times at different bitrates). The disadvantage of this approach is the coarse granularity since there is only a discrete set of quality levels. Furthermore, if there are no clients for a given bitrate, there is no need to generate this level; however, this only costs storage space at the server side and not all servers need to store all levels of a stream. The following section presents a detailed description of the stream switching technique over the HTTP.

2.3 HTTP adaptive streaming

HTTP Adaptive Streaming (HAS) typically stands for a delivery technique based on the stream switching approaches using HTTP connection between a client and a standard HTTP web server as shown in Figure 2.1 and Figure 2.2. It can be also considered as a classical progressive download with the possibility of switching the video quality streams during the playback. In HAS, the video available in multiple encoded and the multiple encoded videos is split into small segments each containing a few seconds of playtime.

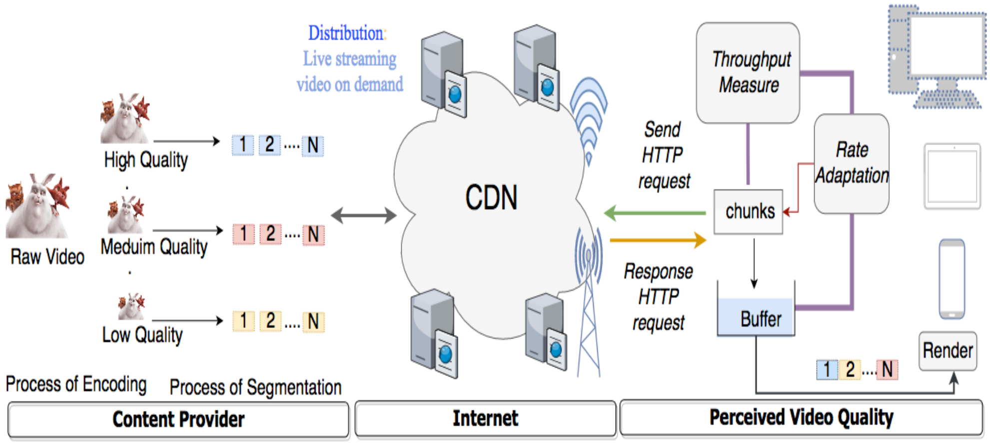


Figure 2.1. Architecture of HAS in Internet.

The client measures the current bandwidth and/or buffer status and requests the next segment of the video in an appropriate bitrate such that stalling is reduced and the available bandwidth is utilized in a best possible way. The increasing number of video applications such as YouTube, Hulu and Netfix employ the HAS technique, as it has several benefits compared to classical streaming approaches. First, offering multiple bitrates of video enables video service providers to adapt the delivered video to the users' demands [32]. For instance, a high bitrate video, which is desired by home users typically having access to high speed Internet and big display screens, is not suitable for mobile users with a small display device and slower data access. Second, different service levels and/or pricing schemes can be offered to customers. For example, the customers could select themselves which bitrate level (quality level) they want to consume. Third, adaptive streaming allows for flexible service models, such that a user can increase or decrease the video quality during playback if desired, and can be charged in the end of a viewing session taking into account the consumed service levels. Finally, and of course the most important advantage is dynamically adapting the current video bitrate, and hence the demanded delivery bandwidth, to changing network and server/CDN conditions. If the video is available in only one bitrate and the conditions change, either the bitrate is smaller than the available bandwidth, which leads to a smooth playback but spares resources, which could be utilized for a better video quality, or the video bitrate is higher than the available bandwidth which leads to delays and eventually stalling, which degrades the QoE [33]. Thus, adaptive streaming might improve the QoE of video streaming. There are several different proprietary solutions based on the same principles for HAS as presented as follows

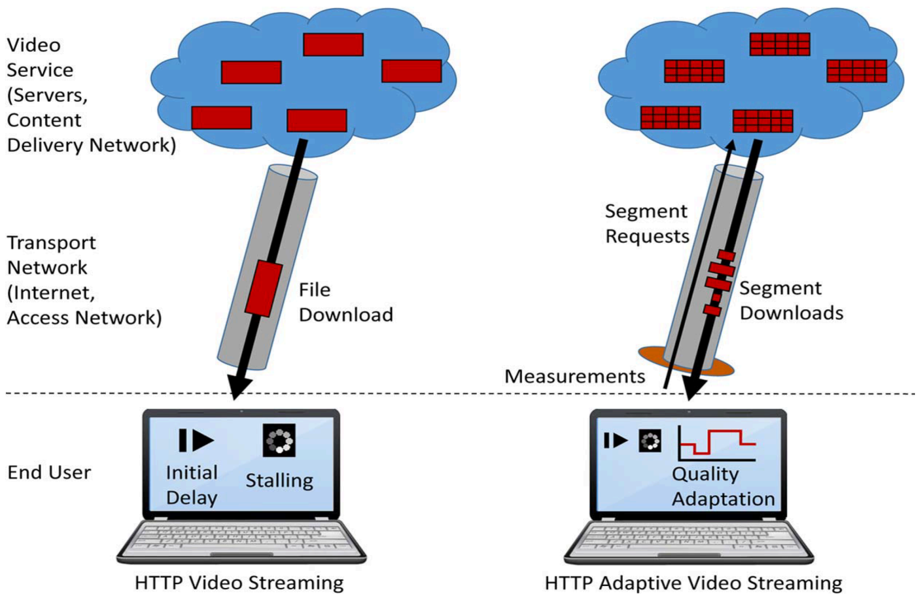


Figure 2.2. Traditional streaming vs. adaptive streaming over HTTP.

Microsoft's Silverlight Smooth Streaming (MSS)

MSS also known as "Silverlight", is a standard for streaming media over HTTP. As in the adaptive streaming over HTTP solutions, in Smooth Streaming the video is segmented into fragments, in this case two seconds segment duration, the segments are available in several bitrates. At the beginning, the lower bitrate segments are sent and, depending on the bandwidth of the client and the processor cycles, the bitrate is increased until the optimum bitrate is reached. Smooth Streaming employs MP4 containers and the H.264 video codec, due to its popularity. Two different formats are used: Disk File Format, an encoded file, and Wired Format, which defines the structure of the fragments. Therefore, several Wired Formats correspond to each Disk File Format. The mp4 files that contain the video and audio have the extension .ismv if they contain only video or audio and video, and .isma if they contain audio only. The server manifest with the bitrates information has the extension .ism. Finally, the customer manifest where the available flows are specified, presents the extension .ismc [34].

Apple's HTTP Live Streaming (HLS)

Apple HTTP Live Streaming is a standard for streaming video over HTTP for playback on iOS devices, including iPhones, iPads, iPods touch and Apple TV, in addition to MAC OS X computers. It supports both live video as content on demand. It also supports the transmission of multiple flows to different bitrates, allowing the client to change the flow depending on the changes in available bandwidth. HLS allows content protection through encryption and authentication over HTTPS of users. In HLS, a file called "Manifest" with extension ".m3u8" is used with the information of the flows and bitrates available. The content must be divided into segments, whose extension is ".ts". The software that serves as the client is part of the iOS 3.0+ operating systems for Apple devices and Safari 4.0+ for web browsers. In live transmissions, the server encodes the video it receives in H.264 and the audio in ACC using the Media Encoder. The output is an MPEG-2 Transport Stream. A stream segmenter fragments the video and the generated manifest is placed on a web server. This manifest is updated periodically each time a segment is generated [35].

Adobe's HTTP Dynamic Streaming (HDS)

HDS is an adaptive transmition standard that allows the distribution of live and on demand videos at adaptive bit rates. It is based on the MP4 Part 14 and Part 12 standards, and transmits over HTTP connections. It transmits high quality video in the H.264 or VP6 formats and can be played in the Adobe Flash Player 10.1 and Adobe AIR 2 players. HDS offers a QoS monitoring service in addition to other features common to similar standards. Mainly the TCP protocol is used for the transmission, possible with formats that allow the interactions between the multimedia server and the player. HDS allows transmission to different bitrates by monitoring the availability of end-to-end bandwidth and the CPU cycles that the player has available. It also allows the transmission of large video files without the need to download the content completely. The HDS technology is based on fragmentation, in which the content is divided into small packages and then proceeded to its transmission. It allows the fragmentation in segments of two or three seconds, but the recommendation is ten seconds. HDS can be implemented on Apache servers [36].

MPEG DASH (Dynamic Adaptive Streaming over HTTP)

DASH is a standard of adaptive streaming of multimedia content over HTTP developed by MPEG (Moving Picture Expert Group). It is related to other technologies of adaptive streaming over HTTP developed by private companies such as Adobe HDS, Apple HLS or Microsoft Smooth Streaming. The client requests the content from the web server through the HTTP protocol. The server provides the client with information about the video segments of different qualities available in an XML document. The ISO / IEC 23009 standard [38] primarily defines two formats: 1) The Media Presentation Description (MPD): Document in which the formats of the segments are specified. 2) Segments: Video content divided in time for proper transport. The advantages of this standard are described: MPEG-DASH is a standard that works independently of the codec, so can be used multimedia content in H.264, WebM and MPEG2TS. This means that it can be used for content encoded according to the specifications of Adobe HDS, Apple HLS or Microsoft Smooth Streaming. MPEG-DASH supports DRM or Digital Rights Management. The content can be encrypted and sent to the client using different DRM methods. The supported DRM methods can be specified in the MPD. The MPD allows to have audio in multiple languages, to select between different videos taken from different angles, to choose to visualize the subtitles in any of the languages that are provided or to change the quality of the video. MPEG-DASH allows the use of segments of variable duration, which is very useful in live retransmissions, where the duration of the next segment can be specified when sending the current segment. In MPEG-DASH, the same content can be available in different URLs, so they can be stored in several servers or come from different sources. The advantage is that the end user can perform the streaming of any of them to obtain a higher performance [37].

Segments and Media Presentation Description (MPD)

HTTP adaptive video streaming works by breaking the content into a sequence of small files called (segments or chunks), each segment contains a short interval of playback time of video content that is potentially many minutes in duration.

Selection of the chunk size is related to placement of I-Frame in the sequence. Due to the temporal prediction commonly applied between the video frames, the frames are not necessarily independently decodable. Therefore, partitioning for segment construction of the video is performed at the group of picture (GOP) boundary. The GOP structure is often referred to two numbers, e.g., M=*X*, N=*Y*. The first number tells the distance between two anchor frames. The second one tells the distance between two I-frames, or the GOP size. For this example, the GOP structure is IBBPBBPBBPBBP. Because exactly one I-frame exists per GOP, longer GOP sizes generally provide greater compression, because encoded B- and P-frames are smaller than I-frames. GOPs are either open or closed. Open GOPs start with one or more B-frames that reference the last P-frame of the previous GOP in addition to the first I-frame of its own GOP. In contrast, closed GOPs cannot contain any frame that refers to a frame in the previous or next GOP. Open GOPs generally provide slightly better compression than closed GOPs of the same structure and size as they allow an extra B-frame in their GOP pattern. However, if the chunk length is large compared to a typical GOP size, more than one GOP is packed into a chunk.

As mentioned before, different information about the content resource is described in MPD. Figure 2.3 illustrates the structure of a MPD and its three major components: periods, representations and segments. Periods are typically larger pieces if media that are played out subsequently, each containing one or more different representations. During a period, sets of adaptation exist which do not change. This means, for instance, period 1 could contain several adaptation options, while period 2 is only available with a reduced set of options. Typically, there are three adaptation sets including different

representations for each period of full-length movie, one for the video, one for the audio and one for the subtitle.
These alternatives representations for the video can have different bitrate, frame rate, frame resolutions or combination of thereof. Finally, at the end of hierarchy, the media chunks are placed. Each chunk is assigned a start time in the media presentation to enable downloading the appropriate chunk in regular playout mode or after seeking. It also contains the location (URL) of the described content [32][37].

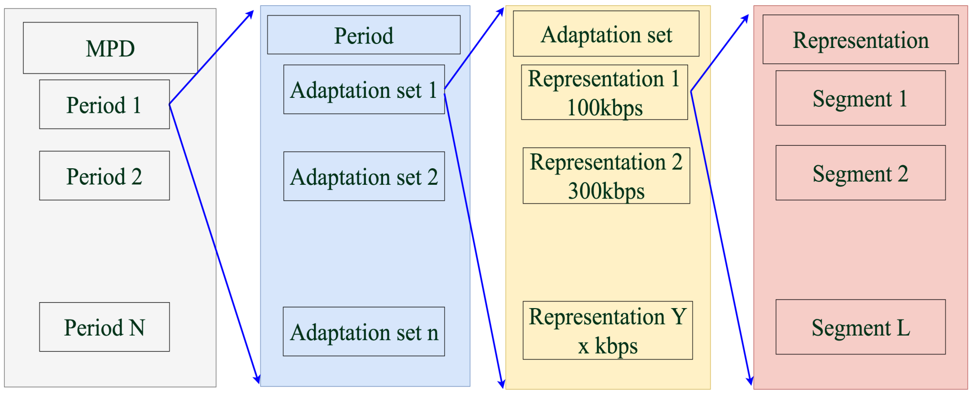


Figure 2.3. Architecture of MPD.

2.3.1 Server side actions

The main concerns on the server side include the preparation of multimedia representation, and a decision about the content delivery to end-user requests, by selecting the CDNs for clients' requests. The server side actions are also the decision on the selection of chunk length. In this regard, two contradictory concerns should be taken into account [39]. In one hand, the chunk length needs to be short enough to allow for the fast reaction to dynamically changing the network conditions. This appends the granularity at which the switching decisions can be made. On the other hand, as stated in [40], there is a trade-off between small chunks resulting in many small files, which have to be stored for multiple bitrates of each video. Furthermore, choosing longer chunk length (longer chunk with longer GOP) increases the coding efficiency of the source video encoder due to the more temporal redundancy in longer segments [41] keeping the amount of overhead low. These two requirements provide an optimization problem, which needs to be considered during content preparation. The study presented in [42] targets the optimization of chunk length so that I-frames and representation switches are placed at optimal positions. Such an approach led to approximately 10% decrease of the required bitrate for a given video image quality. In [43], variable chunk lengths across different representations were considered that for higher bitrates and longer duration of chunks can be used in order to improve the coding efficiency. Selection of the chunk size should be according to the requirements of the actual use case as well. For instance, in context of a live broadcast event, where the content is being made available at the server during the viewing time, a low overall delay introduced by the system should be achieved. This implies that chunk length should be short to be able to be streamed faster. On the contrary, for VoD case, a larger receiver buffer can be used together with longer chunks to avoid flickering caused by frequent quality representation changes. Considering the vehicular mobility scenario, the study presented in [39] show that using long chunk size can be decreased the number of switching, thus, not be suitable for adapting to rapid bandwidth fluctuations and leads to more video stalling. However, this effect can be balanced by increasing the buffer threshold, i.e., the amount of data, which is buffered before the video playback starts. The authors explicitly stated that it is important to configure the buffer threshold in accordance with the used video chunk size. In [44], it is indicated that using a longer chunk leads to higher quality levels of the video and fewer quality changes. However, the number of stalling events and total delay also increase. Therefore, in this dissertation, we focus on the generating chunks sizes can be provided on the server side for the end-users, which are connected to wireless networks.

2.3.2 Client side actions

Mainly the adaptation decision engine of HAS system is running on the client side, it is responsible for selecting which segments in which video quality should be streamed, when to start streaming, and how to manage the receiver video buffer size is running. These decisions are made based on different aspects such as measured client's throughput, the actual video buffer status, device or screen properties, or context information (e.g. mobility, home environment). Figure 2.4 and 2.5 illustrate a general structure of HTTP stream switching approach common between different HAS solutions. In the beginning of the session, the client makes a HTTP Get request to the server in order to obtain the MPD. By default, HTTP GET requests use a single TCP connection, but some adaptive streaming implementations support using multiple concurrent TCP connections for requesting multiple chunks at the same time or for pulling audio and video segments in parallel When the streaming starts, the client requests the chunk at an initial bitrate. After an initial startup delay during which the playout buffer is filled to allow for network throughput variations, the client starts displaying the video. The client continues fetching the subsequent chunks of different quality level to maintain an adequate buffer. Typically, the client keeps a playout buffer for some seconds. The client will request the next chunk based on the result of network tests, monitoring its buffer, measurement of the download time of the previous chunk, as well as considering other local resources. In case the downlink throughput decreases, the client buffer depletes. To prevent stalling, the client requests lower bitrate chunks from the server (down-switching). In turn, when the throughput increases, the client requests higher bitrate chunks (up-switching). Deciding about which chunk to be requested next will be according to the adaptation algorithm. For instance, if the client adaption application is programmed for providing the best quality, the algorithm chooses the highest quality, but in this case, if there is no enough available throughput to the client, the quality of the video is degraded to low quality. If the application is planned for a smooth watching experience, then the adaptation algorithm chooses the most suitable chunk at each request. When a quality profile is selected and the client finds the URL associated with it in the MPD.

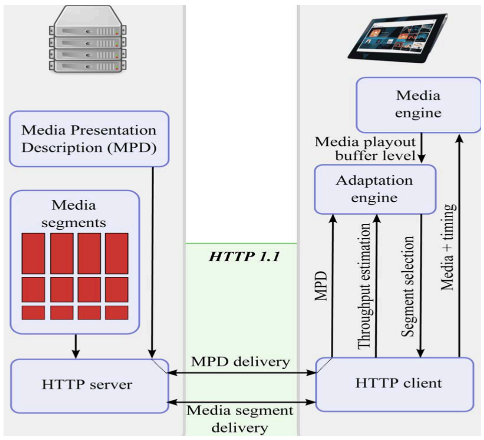


Figure 2.4. Architecture of HAS.

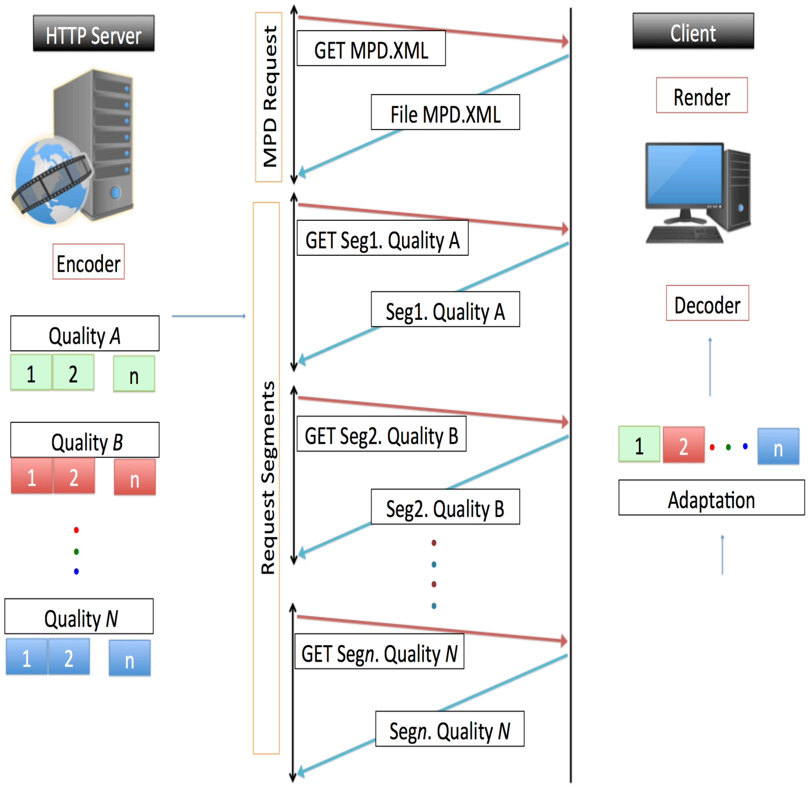


Figure 2.5: Adaptive streaming techniques.

2.4 Influence factors (IFs) in QoE

In order to design reliable QoE metrics, it is necessary to understand the meaning of quality of experiment of the users and the different factors that may influence the QoE in the context of multimedia consumption, networked services, and other electronic communication services and applications. QoE aims at taking into consideration every factor that contributes to a user's perceived quality of a system or service. This includes system, human and contextual factors. QoE can be subject to a range of complex and strongly interrelated factors, falling into three categories: human, system and context. The human factors discuss the variant and stable factors that may potentially bear an influence on QoE. System IFs are classified into four distinct categories, namely; content, media, network and device-related IFs. Finally, the context IFs is decomposed into factors such as physical, temporal, social, economic, task and technical information context [45].

2.4.1 Human influence factor

2.4.1.1 Low-level

Low-level processing properties are related to the physical, emotional and mental constitution of the user might play a major role. These characteristics can be dispositional (e.g., the user's visual and auditory acuity, gender, age) as well as variant and more dynamic (e.g., lower-order emotions, user's mood, motivation, attention). In the human visual system (HVS), visual sensitivity might be the most important factor influencing visual quality. Traditional psychophysical studies assume that visual sensitivity to external stimuli is determined by the spatial and temporal frequencies of the stimuli [46]. Additionally, QoE of visual content can significantly be improved by taking it into account. For example, visual sensitivity models have been widely applied in many advanced video/image compression algorithms and quality assessment methods [47].

Similarly to the HVS, auditory quality and QoE depend on the sensory processing by the periphery of the human auditory system (HAS) [48].

2.4.1.2 High-level

High-level is called cognitive processing, it relates to the understanding of stimuli and the associated interpretative and evaluative processes. It is based on knowledge, i.e. "any information that the perceiver brings to a situation" [49]. As a result, a wide range of additional HIFs is important at this level. Some of them have an invariant or relatively stable nature. Examples in this respect include first of all the socio cultural and educational background, life stage and socio economic position of a human user. Especially in the context of studies investigating the monetary dimension of QoE. Another higher-level characteristic that is often related to the viewing or hearing behaviors when consuming multimedia services, it is guided by the attention mechanism. Attention is a cognitive process of selectively concentrating on certain external objects (e.g., visual or auditory) while paying less or no attention to others [50]. Objects might be salient not only because of their characteristics but also because surrounding objects are not.

2.4.2 Infrastructure influence factors

2.4.2.1 Content and media

The content itself and its type are highly influential to the overall QoE of the system, as different content characteristics might require different system properties [51, 52]. According to [53], in most cases the resources for distributing media are limited. There are both economical as well as hardware-related reasons for limiting the size of media. This is usually accomplished by applying compression. As well as some of the characteristics of multimedia contents and their description are listed in Table 2.3

2.4.2. Network

Network-related SIFs refer to data transmission over a network. The main network characteristics are bandwidth, delay, jitter, loss and error rates and distributions, and throughput [54, 55]. The network-related SIFs may change over time or as a user changes his location, and are tightly related to the network QoS. Network-related SIFs are impacted by errors occurring during the transmission over a network. Streaming video and IPTV are examples of services with more passive consumption, but depending on how they are distributed over the network, Most often the video is deliberately delayed by using strategically placed buffers in order to be more resilient towards network capacity variations and errors.

For UDP and RTP based transmission, the most severe errors are packet losses. Recently, the popularity of over-the-top (OTT) streaming video, e.g. YouTube or Netflix, has increased very rapidly. The distribution method is HTTP and the influence of packet loss and bandwidth limitations is quite different. Network problems will result in freezes without loss of content in the video. Freezing also has a bad influence on the experienced video quality, but can be avoided by using adaptive or scalable codecs in conjunction with OTT video services [56]. Details of the QoS parameters are explained in Table 2.4.

Table 2.3. Description of the characteristic of media content according to codecs and types.

Content characteristic	**Description**
Bit rate	Content bit rate in terms of video transmission refers to the minimum rate at which video bits are transferred from a service source to a destination. The higher bit rate means the better multimedia quality.
Frame rate	Multimedia frame rate refers to a number of media frames are presented per second. The higher frame rate means the video appears smoother and hence, and presented the better video QoE.
Resolution SD/HD/2k/4k	Video resolution refers to the number of pixels in both directions (width and height) of a video frame. A higher frame resolution yields to a better video quality.
2D/3D	Video types i.e., 2D/3D refers to the visual dimensions of a video content. These content types have different service and network requirements.
Media Compression	Lossy compression gives higher compression rates at the cost of quality, Lossless give lower compression rates.
Codec	MPEG2-H264-etc.

2.4.2.3 Device

The visual interface to the user is the display. Capacity of user's device has a tremendous impact on the end-user experience. Therefore, the content quality interacts with the devices. For instance, whether a high quality, high-resolution image is shown on a low-resolution display with few colors, most of the original intent of the image might be lost. However, if a low-resolution image is shown on a large high-resolution display, most likely a very blocky and blurry image will be displayed, but the end result will be highly dependent on the final image scaling procedure [57].

Table 2.4: QoE parameters.

QoE parameters	Description
Bandwidth	Bandwidth is the amount of information that can flow in a network during a specific period of time. To some extent, the higher is the network bandwidth, the higher is the multimedia QoE.
Delay	Delay is the distance between arrival packets. Higher delay results in a lower multimedia QoE.
Packet loss %	Packet loss rate is the ratio of the total number of packets lost in transmission compared to the total number of packets sent. The higher is the packet loss rate; the lower is the multimedia QoE.
Burst loss	If a group of consecutive packets are lost then it is defined as a burst packet loss. A higher burst loss results in a lower multimedia QoE.
Packet error %	Packet error rate is the ratio of the total packets received with errors to the total number of transmitted packets. The higher is the packet error rate; the lower is the multimedia QoE.

2.4.3 Context influence factors

1. **Physical**: The physical context describes the characteristics of location and space, including movements within and transitions between locations; spatial location (e.g. outdoor or indoor, in a personal, professional or social place), functional place and space; sensed environmental attributes); movements and mobility (e.g. sitting, standing, walking or jogging); artifacts. The personal context described in [58] can be partially included here, namely at the user location, user activity and user physiological information level. Hence, physical factors like; heart beat, body temperature, air temperature, noise volume, humidity, lighting conditions, motion and spatial location are used to get similar user clusters. These physical context factors also allow for context-specific processing to increasing QoE, e.g. the ad-

justment of screen brightness on a mobile, depending on lighting conditions. Moreover, the use of spatial context is proposed to provide a better visualization and tracking in multi-camera video surveillance systems in [59; 60].

2. **Temporal:** The temporal context is related with temporal aspects of a given experience, e.g. time of day (morning, afternoon or evening), week, month, season (spring, summer, fall or winter) and year; duration [61], and frequency use of the service/system); before/during/after the experience; actions in relation to time; synchronism.
3. **Social:** The social context is defined by existing of the interpersonal relations during the experience. Hence, it is important to consider if the application/system user is alone or with other persons, and even how different persons are involved in the experience.
4. **Economic:** Costs, subscription type, or brand of the application/system are part of the economic context. Network cost information (e.g. relative distances between the peers) is used in [62], jointly with some physical and social factors, to enable network optimization strategies for media delivery.
5. **Task:** The task context is determined by the nature of the experience. Depending on these situations may arise such as multitasking or interruptions, or task type.
6. **Technical and information context:** The technical and information context describes the relationship between the system of interest and other relevant systems and services including: equipment devices, devices over different wireless connection, applications, Networks availability of other networks than the one currently used, and additional informational artifacts.

2.5 Technical and perceptual effects on QoE-HAS

2.5.1 Impact of waiting-time related impairments

2.5.1.1 Initial delay

In multimedia services, the video playback before starts, there is always a certain amount of data must be transferred before video decoding and playback, this is called an initial startup delay. This initial delay is usually more than technically necessary in order to fill the playback buffer with a bigger amount of video data in the beginning. The playback buffer is an efficient tool used to grab throughput variations in short term. However, a trade off should be also considered between the actual lengths of the corresponding delay (more buffer time = longer initial delay = more frozen playback) and the risk of buffer depletion leads to stall playout of video (more buffer time = higher throughput variations in short term = more frozen playback) [63]. The study presented in [64] showed that the impact of initial delays strongly depends on the type of application. Therefore, related studies for other multimedia Services cannot easily be applied on to other applications like HAS. According to large-scale experiments shown in [65], the initial startup delay does not significantly worsen the perceptual quality and, overall, end-users are willing to tolerate larger startup delays if this results in less video stalling [66]. On the other hand, the initial delay depends on the bitrate of the chunks that are going to be downloaded. If chunks at high (low) bitrate are downloaded, the initial delay will be long (short), but the starting video quality will be high (low). According to [67], indicates a logarithmic relationship between waiting times and mean opinion score (MOS), which is a measure of subjectively perceived quality (QoE). The impact of initial delay on perceived quality is small and depends only on its length but not on video clip duration.

In contrast to expected initial delay, which is waiting before the service and is well known from everyday usage of video applications, stalling invokes a sudden unexpected interruption within the service. Therefore, Few studies have investigated the effect of initial delay of HAS.

2.5.1.2 Stalling

Stalling of video is stopping the video playback due to the playback buffer is underrunning. If the throughput of the video streaming application is lower than the lower video quality bitrate, the playout buffer is depleted. As a result, due to insufficient amount of available data, the playback is interrupted until the buffer receives a certain amount of video data. As a result, amount of rebuffered playback time has to be traded off between the length of the interruption (more buffered play time equal to longer stalling duration) and the risk of a shortly recurring stalling event (more buffered play time equal to longer playback). The exponential relationship between stalling parameters and MOS was presented in [65].

2.5.2 Impact of quality switching related impairments

2.5.2.1 Quality adaptation dimension

Multimedia content provider, in order to provide the video content representations as adaptive bitrate streaming, one or several quality dimensions can be generated. This means, representations can be differed in terms of frame rate (encoding a lower number of frames per second = decreasing video quality), spatial resolution (decreasing the number of pixels in the horizontal and/or vertical dimension of each video frame = decreasing video quality), encoding quantization settings (increasing QP = decreasing video quality), and audio bitrates.

2.5.2.2 Adaptation strategy (switching behavior)

Study reported in [68] has shown that video quality switching is perceived as a degradation of the same video. However, quality switches are often inevitable due to varying bandwidth condition. In this situation, in order to provide an optimal QoE with a given adaptation set, perceptual influence of some key factors must be taken into account, namely, switching frequency, switching magnitude, chunk length, in addition to influence of content characteristics on QoE of aforementioned factors. Apart from that, a fundamental question is whether switching to higher quality gains better QoE at all in comparison to staying in low quality. Type of switches is shown as follows,

Frequency of Switching

Related to the switching frequency, different factors can influence on user's QoE such as: 1) number of quality switches in each adaptation event to reach the target quality level and 2) number of adaptation events occurring during the video playback. The experimental results presented in [69] showed that higher switching frequencies are not penalized in terms of QoE if the duration spent on the high quality is sufficiently long. Therefore, the study was presented in [70] focused on the frequent quality switching should be avoided to allow the users to become familiar with presented video quality.

Switching magnitude

With respect to switching amplitude, two different switching face to end-users, when there is smooth switching (i.e. stepwise change from current to target quality level) and abrupt switching, if the bitrate difference between the current and target quality level impact on QoE. The comparison between smooth and abrupt up- and down-switching in [71] showed that down switching is generally considered annoying. Abrupt up switching, however, might even increase the QoE as users might be happy to notice the visual quality improvement.

2.5.3 HAS Chunk length

Another influence factor on perceptual quality of adaptive streaming could be the chunk length used for switching between different representations. There are different technical concerns on selection of chunk length, which should be taken into account. For instance, employing small chunks improves the client reaction time to network bandwidth variations, but also increases the activity on the client side. This dissertation also observes how much the chunk size affects on the evaluation of the QoE.

2.6 QoE assessment Methods

Generally, video quality assessment refers to the quality evaluation of original version to the processed or impaired version. There are two methods mainly for measuring the quality of service at the level of a user, objective and subjective methods. Therefore, different methodologies of QoE assessment exist. It can be divided into five categories. And the detailed description explains next.

2.6.1 Subjective assessment

Subjective quality assessments are based on psychoacoustic/visual experiments, which represent the fundamental and most reliable way to assess users' QoE, although the most complex and costly method for evaluating users' experience. These methods have been investigated for many years and have enabled researchers to obtain a deeper understanding of the subjective dimensions of QoE. Most commonly, the outcomes of any subjective experiment are quality ratings from users obtained during use of the service (in-service) or after service use (out-of-service), which are then averaged into Mean Opinion Scores (MOSs) [72]. Meanwhile, ITU also sets some corresponding standards for conducting the experiment. The QoE evaluation for video services is the most complicated and some standard are set to conduct the experiment of evaluating subjective video quality. Absolute Category Rating (ACR) method [73], or Single Stimulus (SS) method, where the test sequences are presented one at a time and are rated independently without comparison to an explicit reference. After each presentation, the assessors are asked to evaluate the quality of the sequence presented using an absolute scale, normally with five levels as shown in Table 2.5. Degradation Category Rating (DCR), also known as Double Stimulus Impairment Scale (DSIS) method [74], where each presentation consists of two different video versions, original and the processed or impaired versions of the same content. Both videos are watched consecutively, and the subject is asked to rate the impairment of the second stimulus in relation to the reference. Therefore, Double Stimulus Continuous Quality Scale (DSCQS), Single Stimulus Continuous Quality Evaluation (SSCQE), Simultaneous Double Stimulus for Continuous Evaluation (SDSCE), and Stimulus Comparison Adjectival Categorical Judgment (SCACJ) are the different experiment settings, these settings are similar and the changes mainly reflect in metrics, video reference, video length, number of users, number of observers, etc. [75].

Table 2.5: Subjective measures.

Value	ACR	DCR
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

In addition to standardized subjective QoE assessment methods used for long-term user experience assessment have been used. Studies were involving QoE evaluations of mobile applications focused in [76], they collected users' QoE ratings on their mobile devices via an Experience Sampling Method (ESM) [77] several times per day, while a Day Reconstruction Method (DRM) [78] has been used to interview users on a weekly basis regarding their usage patterns and experiences towards the mobile applications. The method is served to analyze relation between QoE ratings, QoS, and context. With regards to collection of data and running of QoE experiments, evaluations may be conducted in a laboratory setting [79], in a controlled labs environment, or in an actual real world environment [76]. Some performance criteria was modified in a given range in a controlled fashion and subsequently users' opinions regarding the service performance are quantified. As an emerging and very prospective solution focusing on obtaining a large number of ratings in a real world environment, crowdsourcing methodology [80, 81] were studied and utilized.

2.6.2 Objective assessment

The objective evaluation method is defined as using separately the measurement of objective quality to evaluate the subjective quality [82]. In other words, these objective models are expected to provide an indication, which approximates the rating that can be obtained from subjective assessment methods. Different types of objective quality estimation and prediction models have been studied. Each model has its applicable scenarios and corresponding constraints. There are many objective evaluation methods to assess QoE, which can be generally classified into three kinds: full reference, no reference and partial reference. The full reference method is to compare reference video and test video frame by frame while no reference method is to analyze test video only without reference video. Partial reference method is the compromise of the first of two previous approaches, which extracts some characteristics from the reference video and then analyzes the test video according to these characteristics. The advantage of an objective evaluation method lies in its convenience and tractability. Researchers only need concern about QoS parameters, which can be measured and related mathematical problems. It also has its disadvantage of inaccuracy, i.e., the QoE obtained is only an approximation rather than a precise value for each user.

2.6.3 Hybrid Assessment

Both objective and subjective methods separately be applied present significant drawbacks. Subjective methods are very expensive and inappropriate to be used in automatic processes. Objective methods, to be highly reliable, require the original video and the distortion simultaneously, thus preventing its use in real-time systems. For these reasons arises the need for the use of so-called hybrid methods. These methods basically consist of in the use of artificial intelligence tools in which to learn use both subjective and objective measures. One of these methods is the known as Pseudo-Subjective Quality Assessment (PSQA) [83] based networks neuronal and carried out by Project-Team DIONYSOS. Hybrid methods combine the advantages of automating objective methods with the precision of the subjective measures and can be applied in many contexts.

According to the level at which the input information is extracted, there are five types of objective QoE models: (1) media-layer; (2) packet-layer; (3) bit stream-layer; (4) hybrid; and (5) planning models [84]. In addition to ITU-T standards, ETSI gives a comprehensive guide to generic definitions and test methods for most of the key telecommunication services [85]. There are a number of other standardization bodies that deal with QoE assessment, including VQEG, MPEG, and JPEG. While most of the current literature considers objective measures in relation to technology oriented collections of data, it is important to note that objective measurements may also refer to objective estimations of user's behavior (e.g., task duration, number of mouse clicks) which is commonly considered only as subjective [86]. Therefore, MLQoE a modular algorithm for user-centric QoE prediction was proposed [87].

2.6.4 Mathematic model

As a prerequisite to successful QoE management, there is a need for a deep and comprehensive understanding of the influencing factors and multiple dimensions of human quality perception. QoE modeling aims to model the relationship between different measurable QoE influence factors and quantifiable QoE features. Such models serve the purpose of making QoE estimations, given a set of conditions, corresponding as closely as possible to the QoE as perceived by end users.

Therefore, The ability to identify the perceived degree of multimedia stream impairment due to QoS parameters is a key aspect of the QoE prediction of video traffic [88]. Moreover, as discussed in ITU-G. 1080 [89] and TR-126 [90], not all impairments of QoS parameters necessarily result in visible degradations. Therefore, measuring the impacts of a combination of QoS parameters on the quality of the video traffic is still a challenging task. Mathematical models are one of the traditional ways, which create the mapping between the influencing factors to the users' QoE. The data of related factors and users' QoE are obtained in a laboratory environment in general. Researchers conduct statistical analysis to formulate the specific relationship between QoE and the parameters. The E-model recommended by ITU-T SG12 is a classic linear model, which is used to predict the overall quality in a voice conversation at the network planning stage [91]. Although E-model is a mature one, it is restricted in voice service over telecommunication networks. The basic idea of [92] was that the human sensory system can be traced back to the percipience of so called "just noticeable differences" and the differential perception was directly proportional to the relative change of physical stimulus. By taking network QoS as stimulus and QoE as the perception, can be obtained a logarithmic relationship mapping from QoS to QoE. Although the logarithmic model is very convincing which lies in that it is based on the psychological theory, there are limitations that the input QoS should be viewed as a physical stimulus. However, the processes limited numbers of factors, which indicated could not extended. Another mathematical QoE model was based on the "IQX hypothesis" in [93], which argues that a change of QoE depends on both the identical QoS changes and the actual level of the QoE.

2.6.5 Machine learning Model

In recent years, researchers find out that it is hard to formulate the relationship between influencing factors and users' QoE explicitly by a mathematical model in most cases [94]. Machine learning methods are widely applied to solve the problem of the connotative relationship between QoE model and the influencing factors. The Recurrent Neural Network (RNN) model is a classical one of machine learning models, which is applied in the Pseudo Subjective Quality Assessment (PSQA) assessment [95]. RNN is made up of a group of neurons that can communicate with each other by signals. In the RNN network, the state of every neuron is a nonnegative integer named potential and it can be changed by the signals coming from other neurons. At the end of the training, every neuron has a computed potential. Accordingly, the QoE assessment can be obtained by synthesizing the potentials of the RNN. This is not ideal as neural networks and it was computationally complex, and it required large training datasets and prolonged training time. Moreover, their reasoning processes were not transparent. Support Vector Machine (SVM) method is used to assess QoE [96]. A hyper plane can be calculated according to training set data and validated with test data set, then the trained SVM model can be applied to evaluate user current QoE with input factors. Decision Tree (DT), another machine learning model, can also be applied to build QoE model. DT is a widely used classification model and the relationship between QoE and researching influencing factors can be learned by decision tree building and pruning as in [97]. A training data set with input (continuous parameters including time, spatial, bit rate, frame rate information) and output (binary values indicating whether current quality is acceptable or unacceptable for users) were utilized to train the tree model after some pre-processing measures. However, DT only partially suit small datasets, small variations in the dataset require the regeneration of the tree, and the reasoning process is not completely transparent [98]. Moreover, in [99], DT and SVM were also used to build an objective QoE model. The results were then compared with other machine learning methods including ANN, k-Nearest Neighbors (k-NN) and Random Forest (RF). RF was found to perform slightly well than the other examined methods. In addition, a key limitation of current researches focused on compressed or distorted videos [100].

The majority of learning-based on supervised learning techniques, this process obviously slows down the assessment procedure, it scales poorly as the introduction of new video types in the system and distortion conditions in the network requires manual full subjective reference sample labeling. Therefore, the real time cognitive video quality assessment method is proposed in [100], it enables accurate real-time analysis of delivered video quality on client side and offline deep unsupervised learning processes on server side. Thus, different machine learning models have been used for prediction QoE in the field of the Internet multimedia streaming. In Table 2.6 the comparison of some applicable models is shown according to types of applications, influences factors of QoS, assessment metrics, accuracy, and complexity. SVM, DT, and RNN are most useful models to predict QoE. As shown in the table, RNN has high complexity than SVM and DT however precisely is unaccounted. Therefore, the rate of error of IF is significantly lower than other predictor models and the low complexity is shown to predict QoE in this predictor. Although, the entrance parameters of most of the models are focused on the parameters of bandwidth and packet loss, however, the other parameters of the QoS such as jitter and delay are not focused in order find out error rate when the prediction of the QoE is found.

Table 2.6. Different machine learning algorithms.

Technique	Type of service	Factor	Assessment Metric	Precision	Complexity
Logarithmic function [IF]	File downloading	Bandwidth, file size	MOS	RMSE = 0.063	Low
Exponential function [EXF]	VoIP	Packet loss	MOS	R = 0.998	Low
		Re-order	MOS	R = 0.993	
	Web browsing	Waiting time	MOS	R = 0.951	
		Bandwidth	MOS		
Support vector machine [SVM]	Video streaming	Time, bit rate, frame rate	Acceptability (Binary)	Precision 88.592.85% 89.382.77% 91.452.66%	Medium
Decision tree [DT]	Video streaming	Time, frame Rate, bit rate	Acceptability (Binary)	Precision 93.551.76% 90.292.61% 95.462.09%	Medium
Random neural network [RNN]	Video streaming	Packet loss, Time	MOS	Unaccounted	High

2.7 QoE management and optimization approaches

Based on the QoE models and real-time QoE monitoring, intelligent QoE management can be further conducted by network operators based on actual network conditions. Appropriate measures are taken according to the network problem and optimization methods. On the one hand, quality improvement of a current flow, or maximization of system average QoE can be achieved by reasonable network controls, e.g. admission control, priority decision, congestion control and packet scheduling [101]. On the other hand, based on a given QoE model specifying a weighted combination of QoE dimensions and a further mapping to influence factors. A QoE management approach aims to derive Key Quality Indicators (KQIs) [102] and their relation with measureable parameters, along with quality thresholds, for the purpose of fulfilling a set optimization goal (e.g., maximizing QoE to maximize profit, maximizing the number of "satisfied" customers). An important issue to note is that different actors involved in the service provisioning chain use a QoE model in different ways, focusing on those parameters over which a given actor has control (e.g., a network provider is considered how QoS-related performance parameters impact on QoE, while a content or service provider is interested in how the service design or usability impacts the QoE). In wireless networks, a common approach used to optimize QoE is to perform cross layer optimization. In [96], a cross-layer model was established for multimedia traffic in mobile communication systems. QoE is obtained by mapping from the network layer parameter symbol rate. The maximized QoE can be achieved by adjusting the symbol rate for each user with a greedy algorithm. A wise resource allocation strategy with QoE awareness can be realized to efficiently save network resources without user experience degradation.

2.7.1 QoE optimization in Wi-Fi

Delivery adaptive video streaming over wireless networks leads to a trivial performance of the end-users due to video chunk size, video encoding rate, the performance of HTTP/TCP and lack of rate adaptation, which leads to a high oscillation of video quality. These factors provide unsatisfactory Quality of Experience (QoE) to end-users. Several studies have documented the effects of unfairness when streaming adaptive video. Rate adaptation algorithm in adaptive video streaming is employed to identify and select a future stream video segment. The process of the rate selection of the segments is based on estimating the available bandwidth of the end-users. The adaptation algorithm should properly choose the appropriate video quality in order to provider better QoE to end-users. The authors in [63] explained several approaches, which have been used to decide on the selection of the chunk's download, such as buffer-approach and bandwidth-approach. In order to develop a better understanding of effects on QoE, analyzed the video pattern of the adaptive streaming video is explained in [104]. They performed different measures both in competing and non-competing scenarios and studied their effects on the quality of experience of the users. They concluded that using bandwidth allocation techniques it is possible to improve the quality of experience. Moreover, [105] studied the effects produced by ABR video flows when a bottleneck is shared. They concluded that the incorporation of traffic shaping improves the overall quality of the video, the network utilization, and fairness between several video clients. Furthermore, many researchers are working on improving the decrease of the unfairness problem. In [106], a traffic shaping technique has been proposed. It employs the OFF-periods generated by the video traffic pattern to improve the throughput of the different flows. They increment the limit of the throughput at the OFF-periods to improve the bandwidth and decrease the energy consumption of mobile devices. To implement their proposal, an open source SDN controller is employed in their project. , A multi-agent algorithm proposed by authors in [107]. They explained that there is no need to establish communication between agents, furthermore, the algorithm is able to adapt to changing network conditions and it is possible to implement it without changing the architecture of the HAS application. They are able to improve fairness utilizing an intermediate node that collects information of the system. A fairness aware adaptation algorithm has been presented in [108]. It provides a stable bitrate for DASH. Their algorithm utilizes a probe-based bandwidth estimation technique and employs a maximum and minimum threshold that helps to avoid application based-buffer overflow and underflow. Authors of [109] explained the effects of fairness in LAN networks and propose an adaptation scheme that performs an adaptation technique on MAC-layer back-off parameters depending on the parameters of quality of service required by the application layer and the conditions of the physical layer channel. In addition, they employed neural networks to obtain the cross-layer correlations. In [110], authors proposed an OpenFlow assisted system that provides fairness when streaming adaptive video content. They used SDN to modify the characteristics of the video flow in order to improve the quality of experience of the users. Furthermore, the authors of [111] designed and implemented a system by analyzing the network load of OpenFlow for an infrastructure-based IEEE 802.11 network. In their approach, the collection of wireless information of the associated end-users has been statistically analyzed in order to detect the traffic load of the APs and improve the end-to-end QoS.

2.7.2 QoE optimization in cellular network

The next generation of mobile networks as known fifth-generation (5G) brings high improvement for massive data rate services for multimedia streaming. Although, the architecture of 5G includes multiple cell ranges. The process of handover among varies cells faces significant challenges, such as weakness parameters of the handover process on selecting an appropriate network destination and tradeoff the network targets. Latest handover techniques are centered in reducing packet loss and delay. One of the things that caused challenges is Inter Cell Interference (ICI). In [112], authors are proposed a coordinated Multipoint-based (CoMP) algorithm focused on minimizing ICI. They employ the user mobility in order to select the best type of cells in the joint TS. Results show an improvement of the Signal to Noise Ratio (SINR) of the channel and the average throughput per user. Moreover, authors in [113] proposed X2-based the implementation employing Software Defined Networks (SDN). Their proposal aimed to obtain the seamless mobility. Authors of [114] proposed an Advanced Mobility Handover Structure (AMH). They employ IPv6 for nodes to communicate with each other and modify the IPv6 address for the device to maintain it after the handover is performed. Using 5G networks, the tendency is to use a greater number of small cells in order to improve the spatial frequency reuse. This leads to an increasement of the handover rates. The topology aware handover skipping solution proposed in [115], which improves the average throughput by 47%. They estimated the trajectory using the information of the user location and the size of the cells. The measures were performed with velocities that ranged from 30 km/h to 240 km/h.

Video transmission in wireless networks is increasing throughout the years. 5G networks will allow improving video streaming but handover delays can decrease the perceived quality of the video. Authors in [116] proposed an Intelligent Network Selection scheme to improve audio and video streaming in the heterogeneous vehicular network. Their vertical handover scheme allows decreasing the probability of connection breakdowns, point-less handover connections, and failures. It also decreases the delay and packet loss ratio.

2.8 QoE Measurement

The approaches are presented to design subjective and objective based on the models to experiment QoE [101]. Although many research works were involved in QoE, it is still an open problem on how to measure QoE quality. Since the user-subjective information such as user preference is highly related with QoE quality, how to balance the influences between objective and subjective factors is a key issue. Procedure and infrastructure are presented next subsections.

2.8.1 Measurement QoE under commercial network

The obtained data from commercial network for measuring QoE is realistic data because of the collected data based directly on user-dependent. A concept of “living laboratory” in which treat to real-life community within a commercial market was proposed in [117]. With a large amount of users were involved, the measurement result becomes more accurate. A distributed architecture to monitor the QoS was presented in [118], the context information and the subjective user experience based on the functional requirements, which is, related to real-time experience measurements in real-life settings. In this approach the produce of measurement as known crowdsourcing environment, subjective experiments can be performed from distance and there is little control over the participant’s environment. It is computer software assisted method generally performed on a web platform. This methodology mainly involves collecting subjective assessment of quality through ubiquitous streaming via the Internet. In addition, the end-users obviously do not want to be disturbed when they are using the applications. Some smart ways should be found to promote the end-users to response their subjective feelings.

2.8.2 Measurement under laboratory network

Due to the difficulties in collecting operators’ actual network data in commercial network environment, there is a big gap between the data actually got and that data ideally needed. Therefore, many researchers incline to establish the laboratory network environment based on their requirement. Researchers have a better control on the whole infrastructure to facilitate QoE measurement. The laboratory experiment provides a controlled environment for performing subjective tests for evaluating the multimedia quality. Different parameters associated with the test like noise level, distance between screen and users, screen size, etc. can be easily controlled according to the requirements. However, lab based experiments have limitations such as 1) high cost in terms of time and labor 2) limited participants diversity. A laboratory experiment takes weeks for preparing tests, recruiting users, scheduling time slots for supervising the experiments, etc. Also users need to be physically present in the laboratory to perform the test. Generally lab tests are performed in university or research laboratory so the participants for the test are the either students or researchers. The QoE unfairness issue is studied by investigating how the segment duration of the video content affects the MPEG-DASH user QoE level. Amounts of experiments are conducted by using different segment duration on an interactive DVB-T testbed [119]. A VMOS model was proposed to predict the video streaming quality. An accurate end-user subjective perception evaluation is claimed on video streaming with low residual error. An adaptive laboratory test environment is also set up for the subjective measurement. And the result indicated that the correlation between the VMOS score and MOS is no less than 0.9. Thus, the VMOS score is a pretty good metric to evaluate end-user perception [120].

2.8.3 Measurement under simulation network

The simulation network environment has huge role to measure QoE, which is chosen by many researchers to evaluate QoE of multimedia streaming. The advantage of the simulation network environment is that the complex network can be set up quickly in order to meet the need of complex experimental environment for researches. NS-3 simulation tool was used to build a simulation platform for mobile streaming network transmission was and added an Evalvid tool to obtain the MOS values of users [121–123].
However, designing testbeds by using the virtualization technique has become a key component of the network testbeds because they allow testing protocols and applications varying the network conditions [124, 125]. Generally, network testbeds have been used to define specific experiments. Once a testbed has been defined with its resources for a specific experiment, it can be used to test the performance and gather measurements to analyze the results. But what benefits the research and industry community most is to have testbeds that allow the performance of several types of experiments. Therefore, a wide range of testbeds such as: OneLab [126], Emulab [127], G-Lab [128], NetKit [129], and PlanetLab [130] have been designed for different goals. Nevertheless, it is quite difficult for these testbeds to reserve enough resources for their own experiments and they have to meet different requirements than those of testing applications [131]. Moreover, virtualizations systems are utilized to reduce the cost of the testbed setup because the hardware resources are reduced and they allow the creation of more complex network scenarios.

2.9 Challenges in QoE

With the move towards converged all-IP wireless network environments, evaluating and managing end-user QoE for measure multimedia streaming based on IPTV and OTT poses challenges.

2.9.1 Challenges in current QoE.

The parameters used to evaluate the delivery multimedia services are still not good enough for the new services especially when it comes to service context and human subjective factor. Therefore, the user experiment to estimate QoE over wireless networks is additional issue, the subjective and objective parameters not designed to the evaluate experiment users when the users at movement. Scarcity of the correction among the parameters is used to design the prediction model regards to accuracy and fastness (Chapter 3).

2.9.2 Challenges due to methodology

In order to accurately evaluate the QoE of HTTP adaptive streaming [32], it is essential to understand its difference to classical video QoE assessment, which is mainly based on the signal fidelity of static multimedia sequences. On the one aspect, in adaptive streaming, there are initial delay, stalling and switching behaviors whose effect takes up to several seconds. On the other aspect, evaluation of such these events has to be considered in a longer time scale than the video encoding related parameters (resolution, frame rate, quantization parameter, bitrate), which can be assessed in shorter intervals of a few seconds. Nevertheless, the current standardized quality assessment methodologies for subjective testing mostly fall short in accounting for these impairments, and in recommending the test design parameters such as presentation modes, number of test video content and evaluation quality scales (Chapter 4).

2.9.3 Challenges due to application service and resources

OTT video delivery can be challenging because of the concurrent viewers, video service providers (e.g., YouTube and Netflix) and network operators (e.g., ISPs) are having a global view of the end-to-end network condition. Therefore, video service providers unable have access to transit ISPs and the last mile network that actually reaches the viewer. Once a viewer is connected to a content server in a CDN operated by a third party service, it is difficult for the service provider to track the network condition during playback. In addition, the content server is rarely switched to another node after the video starts. Thus, it is possible for the viewers to experience oscillation video quality and stalling the video until the end of the video if the network is unstable. Even if the viewers pay for HD videos, they can end up watching low bitrates due to the Internet-side or CDN-side network problems. In order to mitigate these problems, today's OTT video service providers take advantage of ABR [32] technologies where a video player automatically adjusts bitrates depending on the network conditions. Even though the streaming technologies are designed to provide smooth streaming, it does not resolve the root cause of the congestion. For instance, if the main problem is due to the link congestion in wireless network or wide area networks (WANs) or the content server's malfunction, changing the bitrate is not the best way to improve video QoE (Chapter 5).

2.9.4 Challenges due to cost

Measurement QoE under real testbed environment is involved to highly use the resources of human, networks and device equipment. Also, using commercial environment to obtain dataset is accurate because human participate in the tests. But, using real testbeds are costly high and specification of testbeds for specific tests doesn't allow to usable to another tests because of hard to configure. There are wide ranges of real tesbed designed to measure QoE, but, it is quite difficult for these testbeds to reserve enough resources for their own experiments and they have to meet different requirements than those of testing applications [132]. Also, Designing a real testbed to evaluate the performance of networks and heterogeneous video streaming applications is NOT cost-effective, mainly because collecting (or having full access to) the required resources for the testbed development can be a complex task (Chapter 6).

2.10 Chapter conclusion

Multimedia QoE monitoring and assessment is essential to deliver an optimized end-to-end high QoE service. This approach requires a deep understanding and efficient identification of different objective and subjective parameters that impact the experience of users. Multimedia content delivery is a large and continuously move forwarding field that involves various stages from content service providers to content distributors to Internet service providers, and to content consumers (users) themselves. Therefore, a comprehensive QoE assessment requires the understanding the roles and impacts of these process on the providers to delivery content consumption. A multi-disciplinary approach involving different measures at the server, network, application, or user levels for a wide range of objective (QoS) and subjective (user perception) metrics is necessary for building QoE assessment models. A typical process for building such model includes:

- Conducting subjective lab to evaluate user perception in different scenarios. As the number of impacting parameters is relatively high, the objective of subjective tests is to measure user acceptability with respect to a limited number of parameters like screen size change, player buffering strategy, network conditions change.
- Building correlation model to map between parameters (like QoS parameters) measured during the subjective tests with the QoE scores given by subjects. This phase is considered as the learning phase.
- Evaluating the model against user scores to measure its accuracy.

The QoE assessment model requires the extraction of QoS parameters from different points of the network. For this perspective, the measurement of potential QoS parameters plays a key role in providing the required input data for the quality estimation model. Such measurements can be achieved by installing network monitoring on key points in the network infrastructure. Further, the relationship between QoS and QoE is non-linear. To address this issue there are large numbers of intelligent algorithms however, there is still gap for innovative mechanisms to efficiently correlate QoE from QoS and other objectives. Moreover, Efficient QoE management systems aim at reacting before the user even notices the quality degradation. This requires an efficient feedback that can detect and react in real time to degraded network conditions by controlling or reconfiguring different components. Therefore, real network testbeds have used as an accurate manner to provide subjective and objective tests, however, cost and time of tests in real network environment are extremely high. To address this concern, virtualization can be adopted in both designing and implementation phases to reduce cost and times of experiments. It can be used for testing without real world consequence.

The analysis of the relative issues and conclusions that have been written in this chapter have been published in the journal Network Protocols and Algorithms.

Chapter 3. Developed algorithm for evaluating video artifact

3.1. Introduction

The trend towards video streaming with increased spatial resolutions and dimensions such as SD, HD, and UHD. Video communication via error-prone networks suffers from the visibility of data impairment. Therefore, video-streaming quality is affected by the QoS parameters such as throughput, jitter, delay of packets and packet loss. These parameters lead to different types of perception errors on the clients that decrement greatly the QoE of multimedia services applications. However, knowing their effect on video artifacts is of great importance to both multimedia streaming and video applications employed for other purposes such as medical monitoring systems [133]. Video artifacts can be present in many forms [134]. Some of them may display small squares of different colors depending on the type of artifact. Blocking, mosaic patterns basis image and stationary area fluctuations are some of them. Others generate new lines in the form of borders such as false borders, ringing, which creates a halo throughout the border of an object, and false contouring, which displays thin lines where smooth gradients should be displayed. On the contrary, other artifacts make borders difficult to distinguish such as blurring, chrominance errors and color bleeding. Being, the last two, artifacts that create areas of colors that should not be displayed. Finally, motion compensation mismatch is an artifact generated when an object of the video is in motion leaving behind a halo usually with the same colors as the object.

Nowadays, there are two main approaches that allow measuring QoE, which are objective QoE measures and subjective QoE measurement. Objective QoE measures are widely employed due to its small cost in comparison with subjective QoE methods such as MOS. PSNR, SSIM (Structural Similarity) and MSE (Mean Square Error) are some of the most utilized objectives metrics for evaluating the QoE. However, these metrics are unable to identify which video artifact is affecting video playback, which could be very useful in order to avoid the artifacts that may affect QoE the most. Although video artifacts are sometimes used for artistic purposes [135], the manifestation of these errors should be avoided as they can detriment the QoE of the users. In this section, we evaluate all video artifacts that can be displayed when multimedia streaming faces transmission difficulties. And we propose an algorithm to detect automatically error-prone in the relation between video artifacts and QoS parameters.

3.2 Types of temporal and spatial artifacts

Artifacts are first categorized by whether they're time/sequence-based (temporal) or location-based (spatial) as shown in Figure 3.1. An artifact can be noticed when the video is paused, and then it's probably a spatial artifact. If it's much more visible while the video plays, then it's likely temporal. The compression algorithm being used will either utilize the I-frame (intraframe) or P- B-frames (interframe). I-frame-based algorithms like MPEG are less susceptible to temporal artifacts since I-frames are single image encodings, while P-frames and B-frames hold only part of the image information. Therefore interframe algorithms typically show improved video compression rates, but at the expense of propagating compression losses to subsequent frame predictions. This propagation and "rounding on rounding" is the origin of many temporal artifacts.

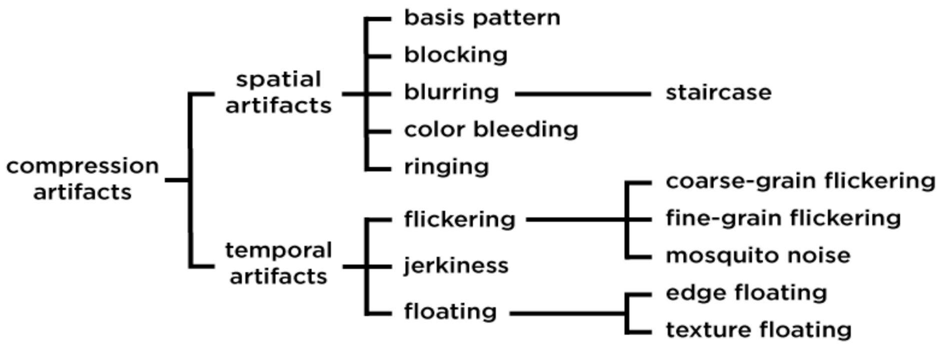


Figure 3.1. Hierarchy of video artifact.

Basis Pattern (Spatial)

The basis pattern effect takes its name from basis functions (mathematical transforms) endemic to all compression algorithms. It usually occurs in regions that have texture, like trees, fields of grass, waves, etc. Typically, if viewers notice a basis pattern, it has a strong negative impact on perceived video quality.

Blocking (Spatial)

Blocking is known by several names; including tiling, jaggies, mosaicing, pixelating, quilting, and checkerboarding. It occurs whenever a complex (compressed) image is streamed over a low bandwidth connection (imagine a golf ball being passed through a garden hose). At decompression, the output of certain decoded blocks makes surrounding pixels appear averaged together to look like larger blocks. As displays increase in size, blocking typically becomes more visible (assuming resolution remains the same). However, an increase in resolution makes blocking artifacts smaller in terms of the image size and therefore less visible at a given viewing distance.

Blurring (Spatial)
Blurring is a result of loss of high spatial frequency image detail, typically at sharp edges. Colloquially referred to as "fuzziness" or "unsharpness," it makes discrete objects as opposed to the entire video appear out of focus.

Color Bleeding (Spatial)
Color bleeding as its name suggests, occurs when the edges of one color in the image unintentionally bleeds or overlaps into another color. Assuming the source video wasn't oversaturated, this artifact is caused by low chroma subsampling.

Flickering (Temporal)
Flickering generally refers to frequent luminance or chrominance changes over time, and is often broken out as fine-grain flickering and coarse-grain flickering. Fine-grain flickering is typically seen in slow motion sequences with large motion or texture details, often appearing to be flashing at high frequency. It can be very eye-catching and annoying to viewers. Coarse-granularity flickering refers to sudden luminance changes in large areas of the video.

Floating (Temporal)
Floating refers to illusory motion in certain regions while the surrounding areas remain static. Visually, these regions appear as if they were floating on top of the surrounding background. This is the result of the encoder erroneously skipping predictive frames, and there are two types of floating: texture floating and edge floating. Texture floating

deals with large areas of texture, like surfaces of water or trees, while edge floating relates to the boundaries of large texture areas, such as the shoreline of a lake.

Jerkiness (Temporal)

Jerkiness, or judder is the perceived uneven or wobbly motion due to frame sampling. It's often caused by the conversion of 24 fps movies to a 30 or 60 fps video format. The process as known as "3:2 pulldown" or "2:3 pulldown," can't create a flawless copy of the original movie because 24 does not divide evenly into 30 or 60. The perception of judder is reduced at higher frame rates because the motion of objects is reduced between frames.

Mosquito noise (Temporal)

Mosquito noise or "edge busyness," gets its name from resembling a mosquito flying around a person's head and shoulders. A variant of flickering, it is typified as haziness or shimmering around high frequency content (sharp transitions between foreground entities and the background or hard edges), and can sometimes be mistaken for ringing.

Ringing (Spatial)

Ringing also known as echoing or ghosting, ringing takes the form of a "halo," band, or "ghost" near sharp edges. Unlike mosquito noise, though, it doesn't move around frame to frame. During image reconstruction (decompression), there's insufficient data to form as sharp an edge as in the original. Mathematically, this causes both over- and undershooting to occur at the samples around the original edge. It's the over- and undershooting that typically introduces the halo effect, creating a silhouette-like shade parallel to the original edge.

Staircase noise (Spatial)

Staircase noise is a special case of blocking along a diagonal or curved edge. Rather than rendering as smooth, it takes on the appearance of stair steps, hence the name. Depending on root cause, stair casing can be categorized as a compression artifact (insufficient sampling rates) or a scalar artifact (spatial resolution is too low).

3.3 Case study

3.3.1 Experimental system description

In order to determine the visual effects of degraded QoS on multimedia streaming content, we have conducted over 100 measures in the real wired network to find out, which appropriate tests of media content can be selected and be analyzed in the experiments. In order to provide the arrangement of the network topology, different equipment are used, as it is presented in Figure 3.2. The server device is described as media provider to the placement of the media content. The provider is connected to switch 1. It transmits the video stream to the client employing real-time transport protocol (RTP). The client is connected to switch 2. Video LAN (VLC) version 2.1.6 is employed on both the server and the client to execute the process of video transmission. In the network, the computer as labeled one, it generates video traffic at seconds 20, 30 and 40 of the video. It is connected to switch 1 and the throughput link of the traffic generator is 10Mbps. When the link is configured to 100Mbps the first traffic generator starts transmitting at the tenth second and the second traffic generator transmits at seconds 20, 30 and 40. The client that receives the traffic from the traffic generators is connected to switch 2. A fourth computer is connected to switch 1 in order to monitor the traffic by using Wireshark 1.10.6. In order to be able sniffing the network, the port connected to the sniffer is mirrored to the one where the link between the switches is located. The link speed between Switch 1 and Switch 2 is 10Gbps. Both the client and the server have an Intel Core i2-2400 and run on Ubuntu 14.04 LTS for 64bits. The video employed for all the tests is Big Buck Bunny at both 30 and 60 fps with resolutions of 800x600, 1024x768, 1280x720, 1280x1024, 1600x1200 and 1920x1080. However, the obtained results for the 1024x720 video at 30 fps and 60fps were selected because artifacts in higher definition videos are displayed more constantly.

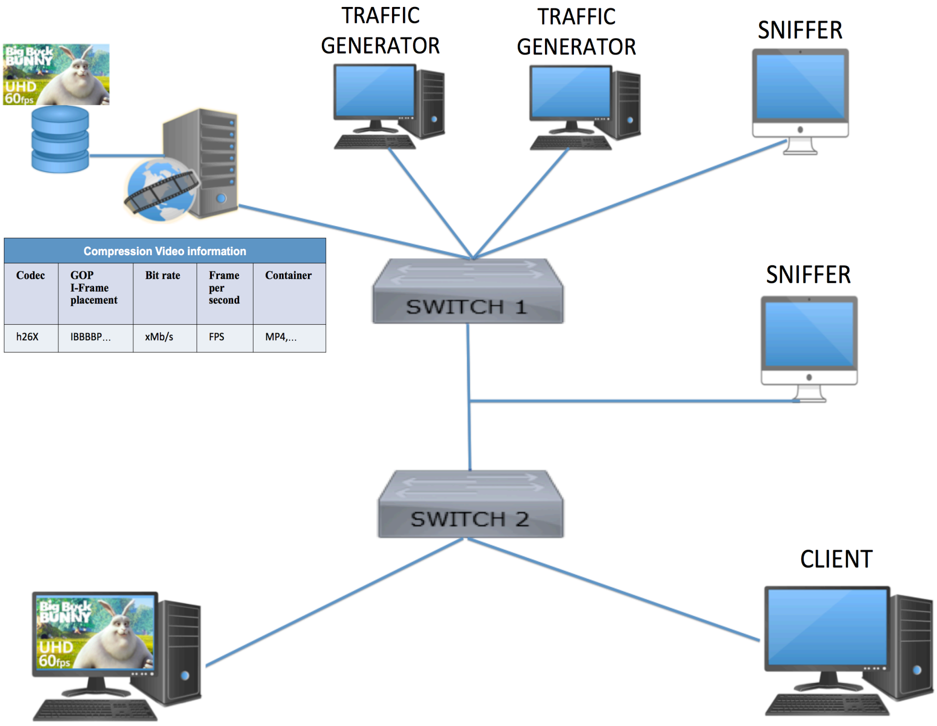


Figure 3.2. Architecture of the experimental setup.

3.2.2 Subjective method and metrics

In order to achieve reliable results, 10 observers selected for the tests. They were non-experts (naïve users) in the sense that they were not directly concerned with television picture quality as a part of their normal work and all of them had correct-to-normal sight. User information as name, occupation, gender, and age are taken. The range of subjects was from 20 to 35 years, including 7 males and 3 females. The average range of the age was 25.

3.3.3 Object measurement

Objective model as a mathematical model can be used for ascertaining the efficient of the algorithm. The objective method based on full reference to help to compute the quality difference by comparing the original video against the delivered video. Every pixel from the source video signal compared to the received video signal, as follows

1) Peak-Signal-to-Noise-Ratio (PSNR)
PSNR gives the average ratio (in dB) between the signals of original video versus the delivered video. PSNR is usually derived via the mean squared error (MSE) between the two signals in relation to the maximum possible value of the luminance of the images. PSNR is accurate to find the perceived quality of the streamed video. The Maximum error varies on color components bits for L component of LUV color space is 100 and 256 for YUV and RGB color spaces. Equation 3.1 define the PSNR formula as expressed as:

$$PSNR = 10\ log_{10} \frac{Max\_Error^2}{MSE} \qquad \textbf{3.1}$$

Where Max_Error is maximum possible absolute value of color components difference, w–video width, h–video height. This metric is equivalent to Mean Square Error, but it is more convenient to use because of logarithmic scale.
The Mean Squared Error (MSE) is measures the average of the squares of the errors. The correct way to calculate average PSNR for a sequence is to calculate average MSE for all frames (average MSE is arithmetic mean of the MSE values for frames). Equation 3.2 describes MSE.

$$MSE = \frac{\sum_{MN}\ [I_1\ (m,n)) - I_2\ (m,n)]^2}{M * N} \qquad \textbf{3.2}$$

2) Structural Similarity Index (SSIM)
SSIM uses a structural distortion based measurement approach. Structure and similarity in this context refer to samples of the signals having strong dependencies between each other, especially when they are close in space. The rationale is that the human vision system is highly specialized in extracting structural information from the viewing field and it is not specialized in extracting the errors. Equation 3.3 gives the SSIM.

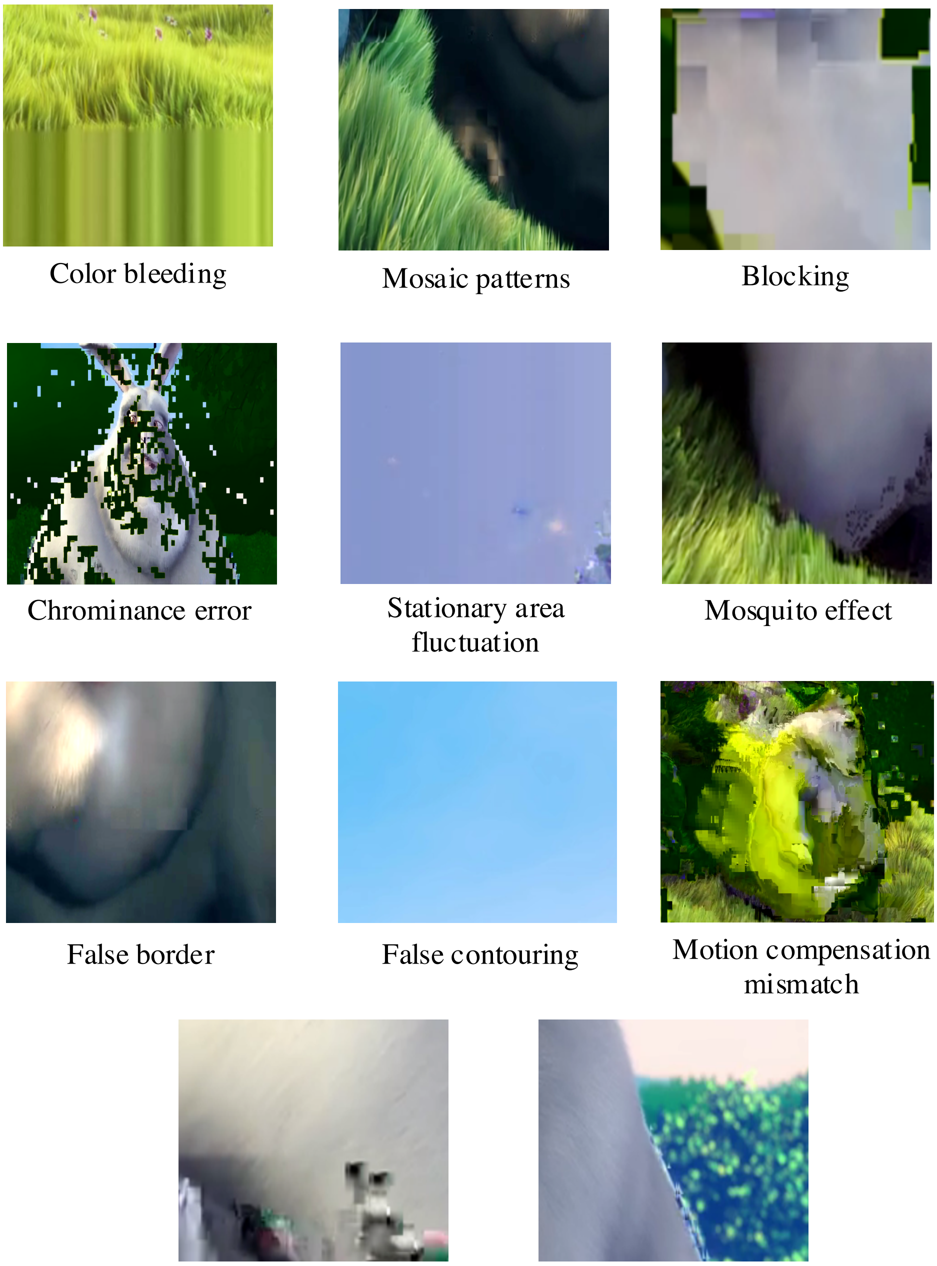
$$SSIM(i,k) = \frac{(2M_i M_k + C_1)(2\sigma_{ik} + C_2)}{(M_i^2 + M_k^2 + C_1)(\sigma_i^2 + \sigma_k^2 + C_2)} \qquad \textbf{3.3}$$

Where $\mathrm{M_i}$ is the average value in the block of the original image, $\mathrm{M_k}$ is the average value in the block of the distorted image, σ_{i}^2 is the variance in the block of the original image, is the variance in the block of the distorted image, and σ_{k}^2 is covariance in the block between the original image and the distorted image. C_1 and $C_2$are the variables to stabilize the division with weak denominator.

3.3.4 Evaluation of experimental results

The presented results are evaluated considering QoS parameters. The objective QoE metrics and the subjective video artifact are identified. All the video artifacts are analyzed in the tests as presented in Figure 3.3. Eleven different types of artifacts are found after performing the subjective assessment. During observation of the tests, many times, several artifacts appeared at the same time on the screen of the devices during the display of video streaming. A similar study was performed in [134]. They identify perceptual artifacts in video streaming explaining the visual peculiarities of each artifact. However, in [134], the QoS was not taken into consideration to realize the effects of QoS on display videos while performing the streaming. In our approach, both perceptual video artifacts and QoS are considered in order to compare the evaluations. The observation of the evaluation is enabled to provide more visual examples of those artifacts. Moreover, the results of artifacts may lead to degradation of perceptual video quality and change according to the characteristic of video and employing different videos. This caused to appear artifacts in the different region of the videos.

In the experiment the places of the artifacts and types of artifacts are depicted in Figure 3.4, at the beginning of the video there was no an artifact to display until the video time reached to the 27th seconds. Different kinds of artifacts appeared according to the figure, therefore, there are the variety of artifact effects appeared at 27th second to the end of the video.



Color bleeding | Mosaic patterns | Blocking

Chrominance error | Stationary area fluctuation | Mosquito effect

False border | False contouring | Motion compensation mismatch

Basic image | Ringing

Figure 3.3. Subjective approach to detect artifacts.

Moreover, when the video time is arrived 49th seconds, the number of availability of artifices is increased. Color bleeding, false borders and blocking were the artifacts that appeared the most. It is also noticeable that the last 10 seconds of the video present a greater number of artifacts matching the moment when the largest amount of disturbing traffic was transmitted of lost packets of the video is presented in Figure 3.5.

Comparing the Figure 3.4 with Figure 3.5, it can be seen that blocking artifacts are related to bandwidth restrictions as they match the lowest bandwidth levels. Basic image artifacts are related to low bandwidth and chrominance errors are related to a high after a small number of lost packets. It is then easy to relate both blocking and mosaic pattern artifact, as the later could be a blocking artifact with more information making it have a wider variation of colors.

Color bleeding and false border artifacts start appearing after the first blocking and mosaic pattern artifacts and continue until the quality of the video worsens. When packets stop reaching its destination larger artifacts begin to appear.

However, as new packets arrive, the errors start disappearing gradually leaving damaged areas that have small artifacts such as stationary area fluctuations and mosquito effects. But, if more packets fail to reach their destination, big color bleeding and false border artifacts may appear. Ringing happens after high packet loss and low available bandwidth, as well as false contouring artifacts.

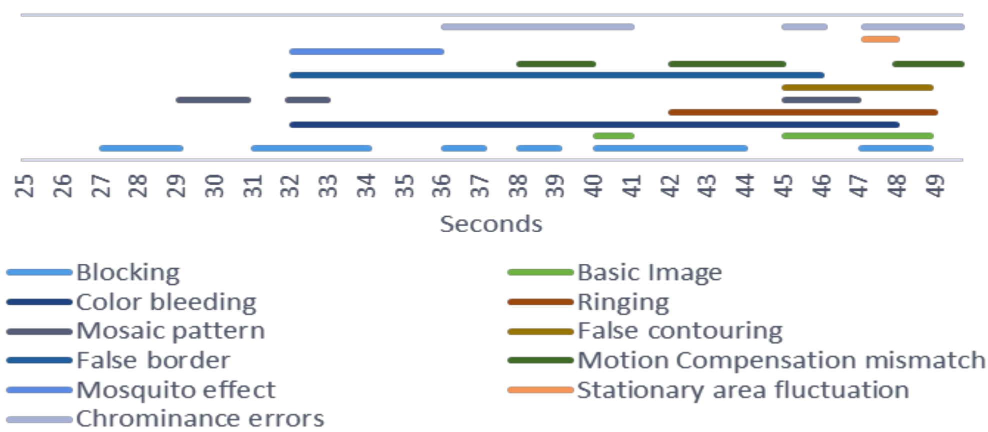


Figure 3.4. Time of emergence of video artifacts of 1024x720 30fps video

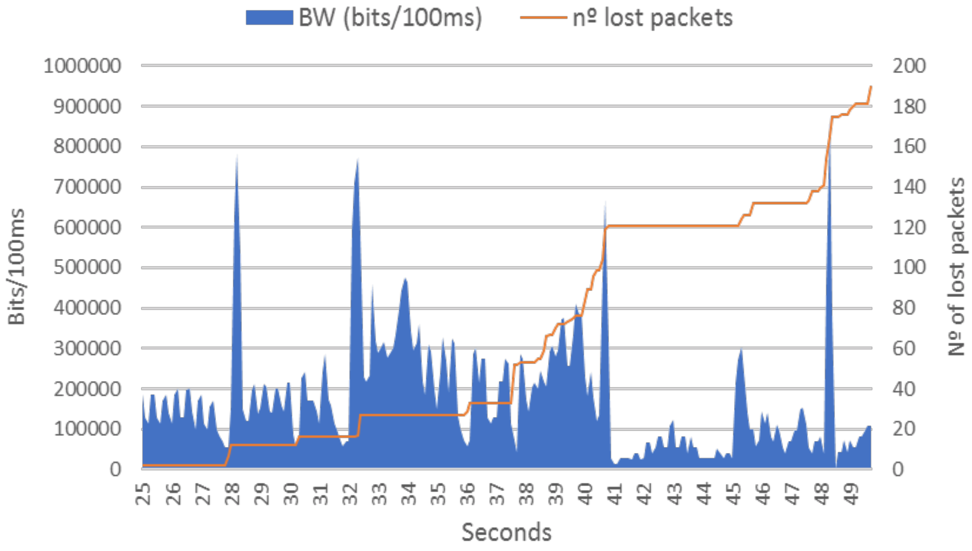


Figure 3.5. Time of emergence of video artifacts of 1024x720 30fps video.

Lastly, when the resolution is 1024x720 and number of frames is 60 FPS. Motion compensation mismatch appears with low bandwidth, which describes a picture in terms of the transformation of a reference picture to the current picture. Low bandwidth is making it difficult for the client to receive more packets in order to solve the errors. As shown in Figure 3.3, the QoE of the user becomes annoyance and it would be harder to distinguish the portion picture appear on the screen and this makes user unsatisfied during watch the video when faced to motion compensation mismatch.

Therefore, the average of the jitter is presented in Figure 3.6. Blocking artifacts are related to the highest jitter values. Results of the objective assessment such as PSNR and SSIM metrics are presented in Figure 3.7. It shows that the end of the video does not have enough quality to be visualized. As depicted in the figure, when the video playback time arrives to 42th second, quality of the video is highly degraded until the time arrives to 44th second. It matches the part of the video where the higher number of artifacts is displayed. It is also noticeable that the quality decreases significantly when the first artifacts appear.

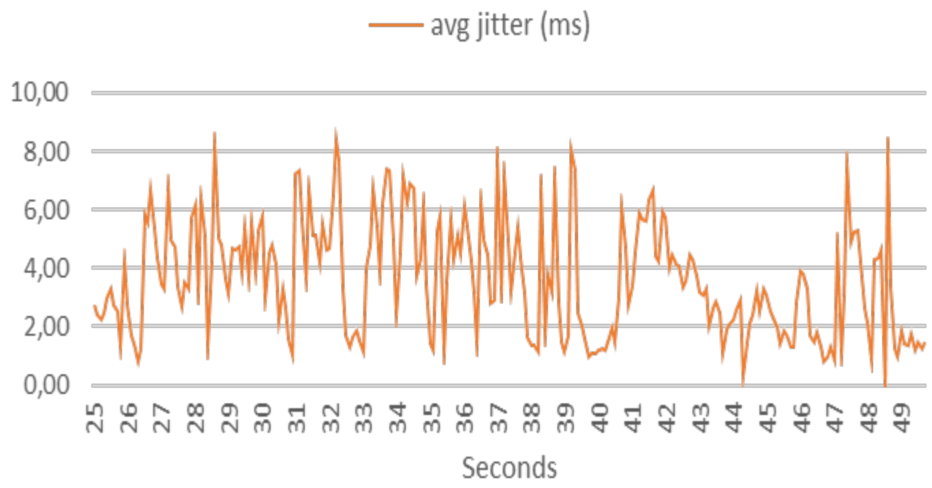


Figure 3.6. Average jitter of 1024x720 30fps.

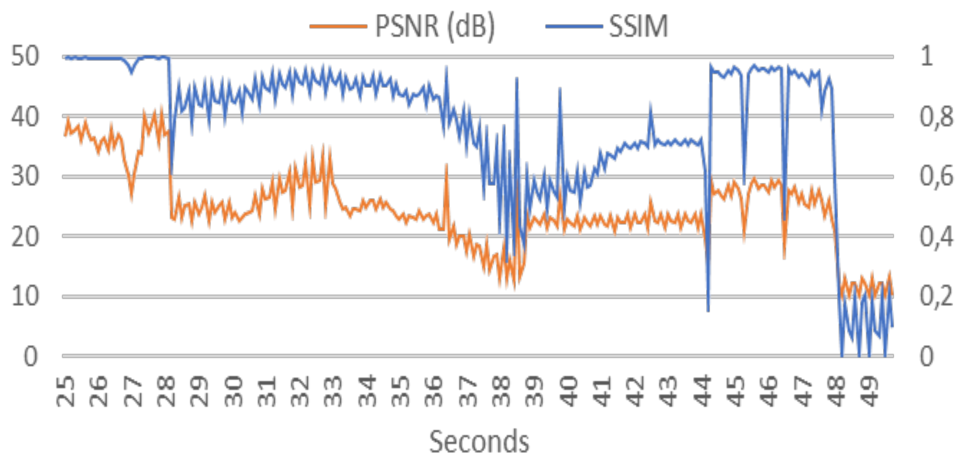


Figure 3.7. PSNR and SSIM of 1024x720 30fps video.

The same process was followed to evaluate the 1024x720 60fps video. Figure 3.8 shows the artifacts that appeared at the last 25th second of the video. The most notorious difference is that, in this case, color-bleeding artifacts did not make the appearance. As in the other video, most of the artifacts are gathered at the end of the video. However, in this video, the perturbance starts earlier and there is a second of good quality video where an intra frame could have solved the previous errors. Motion compensation mismatch and chrominance errors where the artifacts with the most appearance on this video.

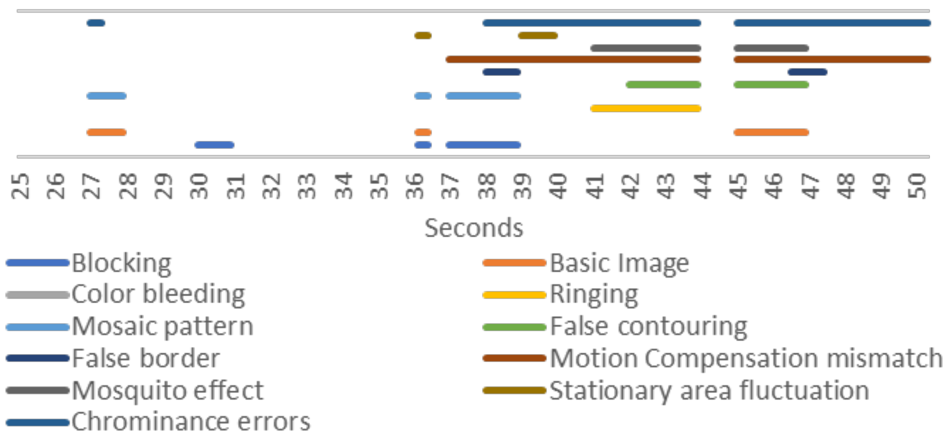


Figure 3.8. Time of emergence of video artifacts of 1024x720 60fps.

The bandwidth and number of lost packets of this video are presented in Figure 3.9. Again, blocking, mosaic pattern and basic image artifacts are related to low bandwidth. Chrominance errors are related to a constant increase of packet loss. Comparing both videos, it is easily noticeable that chrominance error artifacts match the steep regions of the graph. Contrary to the other video, color bleeding does not appear. Artifacts on the second video were bigger leading to fewer artifacts that are caused after other artifacts. Also, in this case, false border artifacts were much fewer. Ringing happens again after a high number of lost packets followed by low bandwidth. The effects of false contouring and mosquito are triggered by motion compensation mismatch artifacts, low bandwidth and high packet loss. The stationary area fluctuations are related to a small number of lost packets.

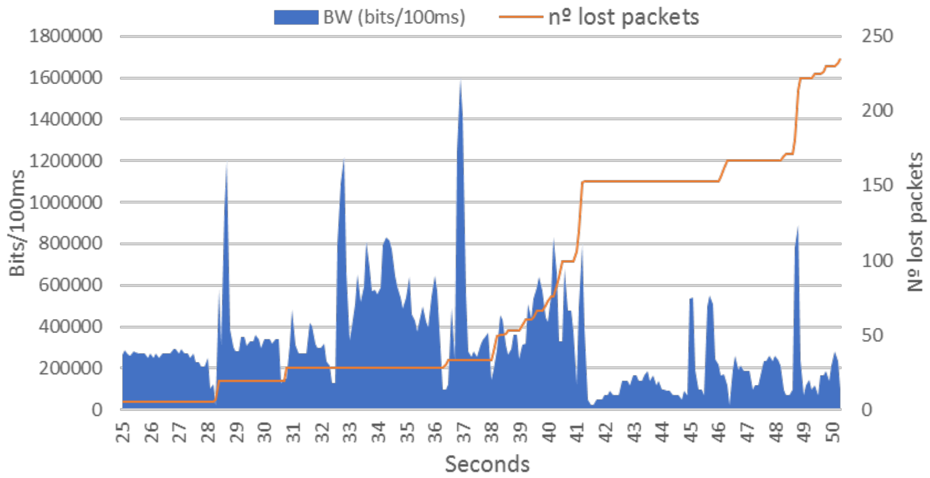


Figure 3.9. Time of emergence of video artifacts of 1024x720 60fps.

The average jitter of the video is presented in Figure 3.10. Blocking is again related to high jitter, as well as mosaic pattern artifacts.

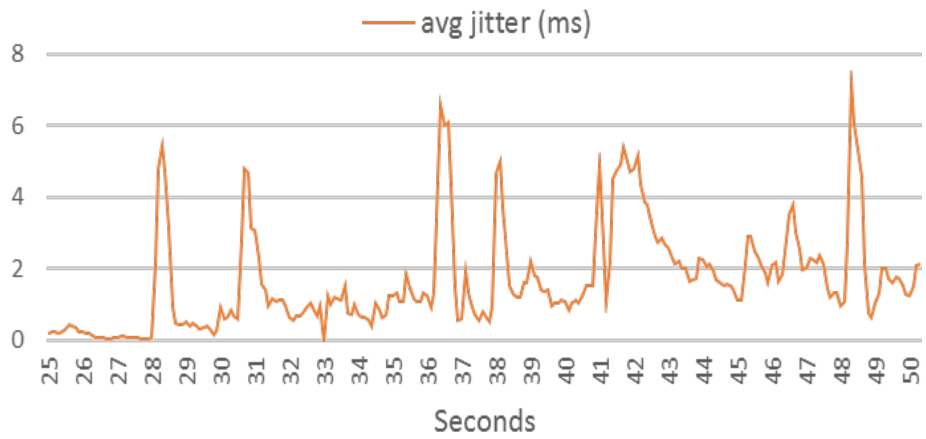


Figure 3.10. Average jitter of 1024x720 60fps video..

Lastly, the PSNR and SSIM measures of the second video are presented in Fig. 10. After the parts of the video were the artifacts disappear, the PSNR and SSIM improve but they decrease again after the manifestation of more artifacts. Motion compensation mismatch artifacts are the ones than decrease QoE most, as they tend to be big and prevent from visualizing correctly the following frames.

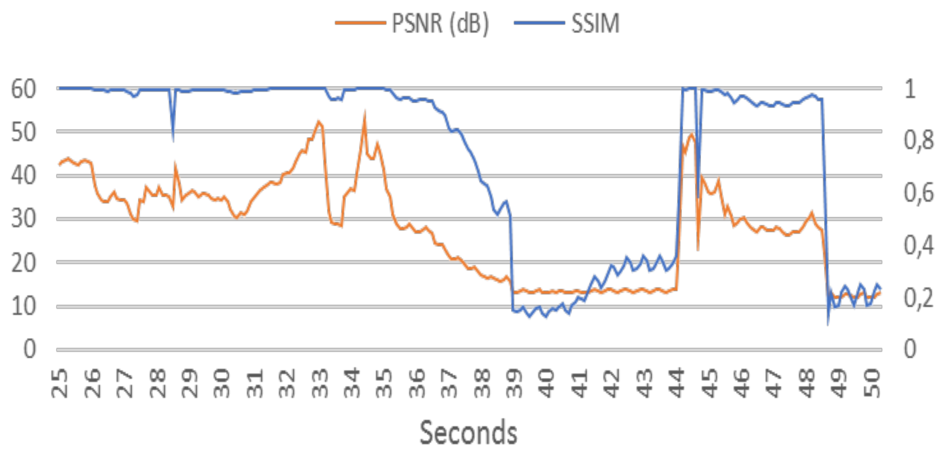


Figure 3.11. PSNR and SSIM of 1024x720 60fps video.

3.5 Proposal of algorithm for detecting artifacts

The proposal of a video evaluation solution is detailed in order to provide the performance of the algorithm and the structure of the error log. The error log contains a record of critical errors that occurred during the artifact process. The proposed system utilizes image-processing techniques in order to detect video artifacts and to classify them according to the different video artifacts detailed in Figure 3.3.

The proposed algorithm is based on a smart approach to learn about the artifact problems and detail of the operation of the algorithm solution is presented in Figure. 3.12

The system algorithm compares both the original and the received video and detects the errors that appear as a consequence of the streaming process. When an artifact is found, the system stores information about the errors, such as the chrominance values, to create a database of video artifacts that allows the system to learn how the error is displayed on the video in order to detect future errors more accurately. After the artifact is classified a record of the error is stored in a database for further analysis. Then, a subjective evaluation is requested. This evaluation is performed in order to avoid mistakes and to allow the system to learn how to classify the artifacts and it is performed following the same method employed the previous section. After an initial learning phase is conducted the learning module can be disabled. When the subjective analysis is performed, the error log is updated and possible mistakes made by the system are replaced with the subjective observations. The structure of the log is presented in Table 3.1.

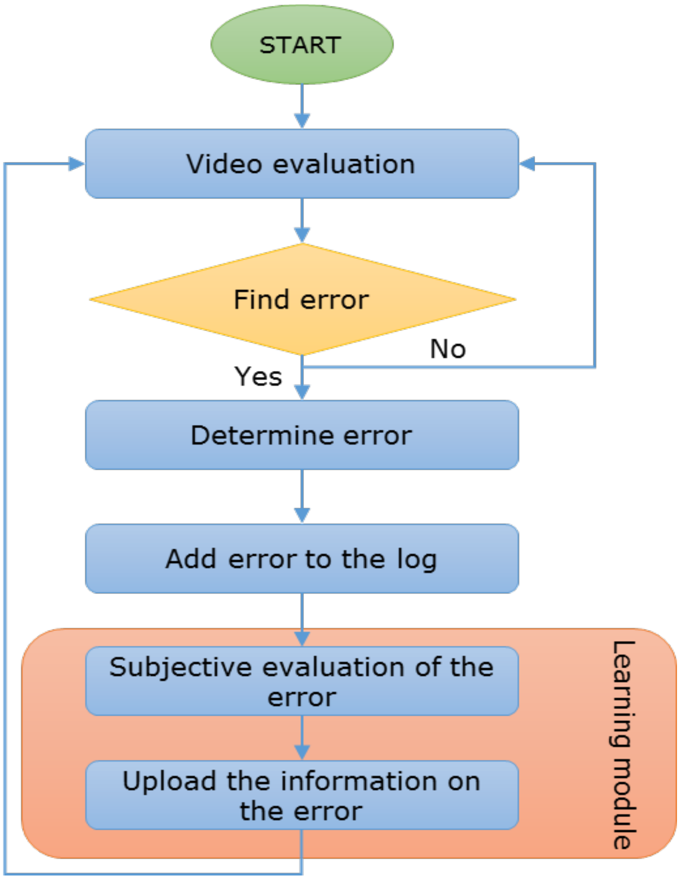


Figure 3.12: Proposal algorithm to detect video artifact.

Table 3.1. Structure of the artifact log.

Error Number	Type	Subjective assessment	BW	Packet loss	Jitter	Delay
Int	Int	Int	Float	Int	Float	Float

The error number is employed as the ID of the log. It starts with the number 1 and it is incremented each new log. Table 3.2 presents the assignations for each video artifact. The subjective assessment indicates the same number as the field type, whether the system performed the evaluation correctly. Otherwise, the number will indicate the correct video artifact employing the same number assignment presented before. Our solution will measure QoS constantly and store the results however, when a new log is generated, the current information on the bandwidth, packet loss, jitter and delay is added to the log for further analysis.

Table 3.2. Assignation types of the video artifacts.

Type	Video artifact
0	Unclassifiable
1	Color bleeding
2	Mosaic patterns
3	Blocking
4	Chrominance error
5	Stationary area fluctuation
6	Mosquito effect
7	False border
8	False contouring
9	Motion compensation mismatch
10	Basic image

3.6 Chapter conclusion

Video artifacts are one of the factors that decrease the QoE of users of multimedia services. We provided two approaches of QoE assessment when the videos are streamed on the Internet. We analyzed the factor are affected on creating the artifacts in the transmission of video streaming while introducing undesired traffic to the network. The subjective artifact data is obtained from subjective identification of video. All artifact types are stored in a database. Objective metrics are used to find the QoE evolution when the artifacts are occurred in video transmission.

Types of the artifacts are compared to the QoS measures in order to evaluate the objective QoE.

Results show that the QoS parameters such as bandwidth, packet loss, and jitter are affected on objective QoE. The proposed algorithm is presented to evaluate the obtained training subjective data according to types of the artifacts are occurred in the video transmission.

Chapter 4. Algorithm for QoE prediction in multicasting

4.1 Introduction

Increasing the interest of playback high quality video definitions such as HD, 2K, and 4K by customers in both schemes of on-demand and live streaming over the current telecommunication systems brings the questions of how the multimedia providers can provide satisfactory streaming of the services to their end-users. One of the important metrics can help service providers to estimate QoE is assessing the perceived quality of delivered video service.

Fastness of UDP as an attractive feature is used for streaming videos over the Internet to enormous clients receivers. However, UDP is not designed to keep track of retransmissions. It is sensitive to delay and loss of packets. In addition, UDP is not always sent packets in order. And lack of communication between devices and stream media over UDP can lead to transmission errors [136]. In wireless connection, using UDP for streaming giant video quality and assessing the received video streaming becomes unpleasant and inaccurate due to the instability of the wireless channels in producing high delay and the probability of a packet loss, which are much higher than that in wired connection networks.

Take into account, the combination of a few parameters is provided an insufficient model for evaluating the QoE. All the parameters such as the characteristic of the video, the service provider, the network service, and the device, are involved to provide better and worse QoE.

Therefore, the common mathematical metrics for evaluating the perceived quality do not perfectly correlate to human visual quality, because of the metrics fail to capture the packet loss characteristics of wireless networks. In this chapter, in order to assess and manage the QoE of the end users in multicast networks, the prediction QoE model based on machine learning model is proposed in order to provide the accurate evaluation of the QoE. Therefore, it allow service providers to adapt their service according to the assessment model is used for QoE estimation in multicast video transmission.

4.2 Metrics for assessing QoE

The main QoE assessment approaches are used to evaluate the delivery media stream over network service, this is led the providers to learn from their users about the service feedback. In general, QoE evaluation methods can be divided into two categories, subjective assessment, and objective assessment. Our proposed framework is designed to make predictions on retrospective QoE scores i.e. the subjective score given by subjects after the video playback has finished. In order to capture both video quality and to predict reactions to playback video stalling, we compute the following types of QoE-relevant input features:

4.2.1 Object metrics

Objective model as mathematical model can be used for ascertaining the efficient of the algorithm. The objective method can be full-reference, no-reference, and reduce-reference. Full reference metrics helps to compute the quality difference by comparing the original video against delivered video. Every pixel from the source video signal compared to the received video signal, The FR is most accurate at the expense of higher computational effort. There are many parameters for evaluation of objective such as DELTA, MSAD, MSE, SNR, PSNR, SSIM (See section 3.1.), VQM, APSNR, and OPSNR. DELTA, MSAD, MSE, SNR and PSNR are mathematically calculated error as a difference between the original and processed pixel [128]. SSIM, NQI, and VQM metrics have quite better performance compared to PSNR and in most cases performs very similar to the Human Visual System (HVS). Select best of them depend on the time and the accuracy of the assessment.

1) Average Peak-Signal-to-Noise-Ratio (APSNR)

APSNR gives the average ratio (in dB) between the signals of original video versus the delivered video. APSNR is usually derived via the mean squared error (MSE) between the two signals in relation to the maximum possible value of the luminance of the images. APSNR is more accurate to find the perceived quality of the streamed video over the wireless network. The Maximum error varies on color components bits for L component of LUV color space is 100 and 256 for YUV and RGB color spaces. The APSR is expressed in Equation 4.1.

$$APSNR = 10\, log_{10} \frac{Max\_Error^2}{MSE\ (all\ Frames)} \qquad 4.1$$

Where Max_Error is maximum possible absolute value of color components difference, w–video width, h–video height. This metric is equivalent to Mean Square Error, but it is more convenient to use because of logarithmic scale. The Mean Squared Error (MSE) is measures the average of the squares of the errors. The correct way to calculate average APSNR for a sequence is to calculate average MSE for all frames (average MSE is arithmetic mean of the MSE values for frames).

2) Mean Absolute Difference (MSAD)

Mean absolute difference of the color components in the correspondent points of image. This metric is used for testing codecs and filters as given in Equation 4.2.

$$d(X,Y) = \frac{\sum_{i=1,j=1}^{m,n} |X_{i,j} \ \ Y_{i,j}\ |}{mn} \qquad 4.2$$

Where $X_{i,j}$ includes the values of image colors in original block and $Y_{i,j}$ includes the values of image color the corresponding pixel in the block being used for comparison, such as Red color, green color, and blue color.

3) Video Quality Metric (VQM)

It is the modified existing DCT (discrete cosine transform) coefficients based on Watson's proposal, which exploits the property of visual perception to correspond to human perception. The VQM performs better than those situations when Root-MSE (RMSE) fails. The light computation and memory load make VQM even more attractive for measure perceived video quality in wide applications. Brighter blocks correspond to greater difference.

$$\boldsymbol{Mean_{\_dist}} = \mathbf{1000} * \boldsymbol{mean}\ (|\boldsymbol{diff}|) \qquad 4.3$$

Where 1000 is the standardization ration.

The maximum distance between the blocks in DCT transformation $Max_{\_dist}$ and the VQM score is expressed in Equations 4.4 and 4.5.

$$\boldsymbol{Max_{\_dist} = 1000 * Max\ (|diff|)} \qquad 4.4$$

$$\boldsymbol{VQM\ = (Mean\ _{-dist} + 0.005 * Max\ _{\_dist})} \qquad 4.5$$

Where 0.005 is the maximum distortion weight chosen based on several primitive psychophysics experiments.

4.2.2 Subjective metrics

Subjective metrics are used to evaluate the QoE of media services according to perceived service by end-users, which can be divided into approach evaluations:

1) Mean Opinion Score (MOS): It is the accurate approach to evaluate the perceived video quality in the domain of QoE. It is conducted based on psychological/visual experiments. However, the quality score given by a human also depends on the evaluator's experience. Therefore, it is most reliable but also the most complicated and expensive method of evaluating user' QoE. The assessment consists in building a panel of human observers, which evaluated the video, depending on the point of view and the perception.

2) Difference Mean Opinion Score (DMOS): The DMOS procedure can help evaluation of the perceptual quality processed through determine how much the differences introduce in the test video degrade subjective assessment picture quality.

4.3. Network measurements

The monitor points allow detecting the quantity of packet is flown in the server and arrived at the receivers. Those points can specify the detail of bit-rate, error rate, latency and variation delay during the broadcast. The real monitoring of the selected points shows the diversity of the quantity of the number of packets in the transmission and realizing the impact of the QoS. These points are described as Bit-rate guarantees to show the maximum and minimum bits transfer during the session for each channel. Moreover, All of the captured data from the points is stored in a database to learn about the measure analyzed. The characteristic data flow in the different aspect of the networks equipment is important in order to understand the precise relation between the QoE metrics.

4.4 Prediction model

QoE-prediction based on Machine Learning (ML) is concerned with the design and development of algorithms for the media's platforms, using ML has capable to automatically improve the performance of the end-user experience over time. Broadly, there are two types of ML, supervised and unsupervised learning. Supervised learning refers to the category structure and hierarchy of the dataset is already known. The learning requires a set of labeled classes and returns a function that maps the dataset to the predefined class labels. Unsupervised learning referred to the process of finding the hidden structure in unlabeled data in order to classify them into meaningful categories. Therefore, the general functions provided by ML are training, recognition, generalization, adaptation, improvement, and intelligibility.

The proposed approach uses the least absolute shrinkage and selection operator (LASSO) to select variables of the characteristic of videos, network service, and device capacity. Obtain information from various resources establishes prediction model for assessing QoE of the end-users.

In LASSO, as Given a linear regression with standardized predictors X_{ij} and centered response values Y_i for i=1,2...N and j=1,2...p, N is being the number of samples and p is characteristic per sample, in our case, different objective QoE assessment such as APSNR, MSAD, ISSM, and VQM is extracted to provide the perdition lasso model. The lasso solves the l_1-penalized regression problem offending $\beta = \{\beta_j\}$ to minimize by

$$\boldsymbol{\beta} = \sum_{i=1}^{N}\left(y_i - \sum_j x_{ij}\,\beta_j\right)^2 + \lambda \sum_{j=1}^{p}\left|\beta_j\right| \qquad 4.6$$

This is equivalent to minimizing the sum of squares with a constraint of the form $\sum|\beta_j| \leq s$. It is similar to ridge regression, which has the constraint:

$\sum_j \beta_j^2 \leq t$. Because of the form of the l_1-penalty, the lasso does variable selection and shrinkage, whereas ridge regression, in contrast, only shrinks.

If we consider a more general penalty of the form, the lasso uses q = 1 and ridge regression has q = 2. Subset selection emerges as q → 0, and the lasso uses the smallest value of q (i.e. closest to subset selection) that yields a convex problem. Convexity is very attractive for computational purposes. It is shown in Equation 4.7.

$$\mathbf{q} = \left(\sum_{j=1}^{p} \boldsymbol{\beta}_j^q \right)^{1/q} \tag{4.7}$$

4.5 Multicast description

In order to bring a solution to our problem, we propose the designed multicast application for streaming the multimedia. We concern with wireless network multicasting of high quality video streaming, which can be raised the following scenarios such as interactive TV live events, broadcast live lessons at university campus, stage performance in concerts, live sport broadcast and live multicast cinema, shown in Figure 4.1.

The provider employs a multicast multimedia application for streaming the video contents over telecommunication systems. The providing multimedia content can be lived or on-demanded. In the live scenario, the action includes recording a real event from the cameras and then the medias are encoded in the standard quality. The on-demand scenario, the media content previously decided to encode in the standard quality. The process of media encoding includes the stream bitrate, frame rates per second, resolution, codec type, and video and audio container.

The Multicast application can stream different events to different environments, each event can describe under a channel as depicted in the figure. The provider can multicast the stream as internal and external e.g. Internet or intranet. Indeed, the service provider connects over Internet to distribute the different events and delivering these events to range of purposed receivers. Therefore, the media providers are using content distributor to provide optimal performance when it is delivering the media to the end-users whether transcoding of the media is necessary.

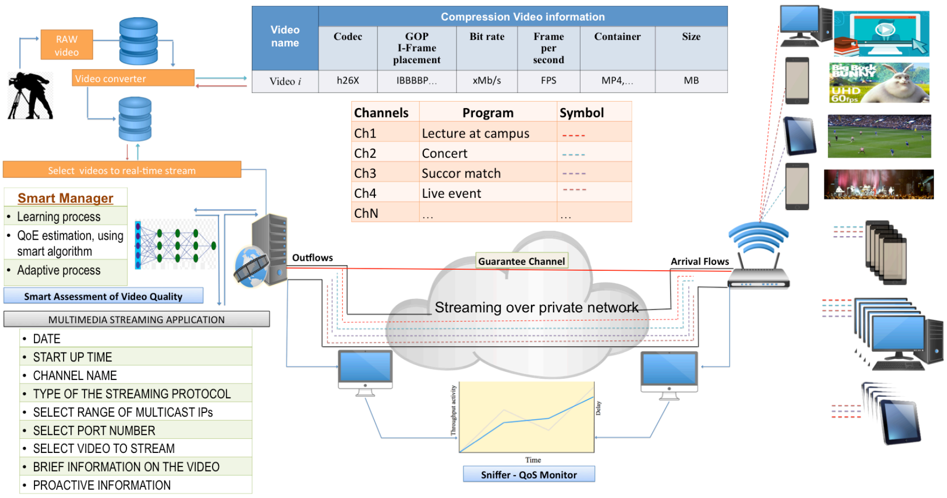


Figure 4.1. Description of the system architecture.

4.6 Case study

In order to provide the multicast video streaming scenarios according to Figure 4.1, the multicast application is designed. The application is based on the using Java language program, which allows the media provider to stream videos to a group or multiple groups of users. The application at application layer reads video data, which has MP4 container. Its transportation layer is based on the UDP protocol in order to provide the technique for one-to-many communication over an IP infrastructure in the network. The application provides some features such as a timetable for multicast videos, range of IP and ports for the multicasts, transcoding of the video if necessary, and a timer to launch the multicast videos. Therefore, the application can launch multiple videos to stream simultaneously to the variety of receivers. Therefore, it uses FFMPEG [137] in the case when the video required transcoding in order to select the correct formats. The clients' applications consist of using the open source VLC as a media player [138] and the other interface window, which allows the clients to send the information of the GOP to the server.

4.6.1 Real Testbed to apply experiments

To provide the experiments, we consider a real testbed in the university's laboratory. The components of the testbed are included heterogeneous devices and equipment, such as the multicast server, fix, and mobility devices, and the hardware emulator device. The real testbed is aimed to observe the experimental tests precisely and obtained the accurate values of the video assessments. The network topology of the real testbed is illustrated in Figure 4.2, which depicts a typical tree-based access network of a multicast server, university Internet, a wireless medium and 20 laboratory devices. The end-users' devices are connected throughout wireless 802.11-access link to receive multimedia streaming.

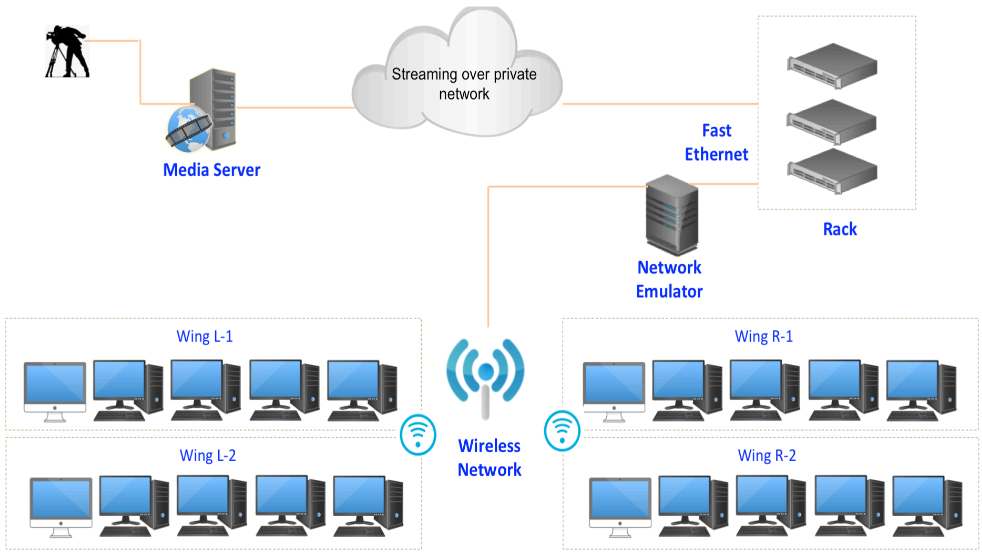


Figure 4.2. The testbed scenario.

The wireless device directly connected to university Internet. The network emulator point is a traffic shaping emulator in the testbed, it controls throughput of available bandwidth by prioritizing network resources and guarantee certain bandwidth based on predefined policy rules. It uses concepts of traffic classification, policy rules, queue disciplines and quality of service (QoS). The emulator is the combination of traffic control (TC) queuing discipline Hierarchy Token Bucket (HTB) and Network Emulation

(NetEm) in order to shape and control the network link's upload, network link's download, delay and packet loss ratio [139]. Therefore, the characteristic of the heterogeneous devices is explained in Table 4.1. Moreover, IP-ERF [140] and ookla speed test [141] are used to measure the connectivity and availability of the networks.

Table 4.1. Characteristic of the testbed equipment.

No.	Technical specification	Characteristics 802.11 (ac)	No.	Equipment Characteristic	MacBook	PCs	Server
1	Frequency	5GHz	1	Version	MacBook Pro retina	Cooler master	Cooler maste
2	Modulation scheme	OFDM	2	O.S	MacOS sierra	Windows	Ubuntu
3	Channel Bandwidth	20,40,80 MHz	3	Processor	2,4 GHz Intel Core i7	2,4 GHz Intel Core i5	2,4 GHz Intel Core i5
4	Data rate	1300Mbp	4	RAM	8 GB 1600 MHz DDR3	8 GB	8 GB
5	Aggregation Data rate	Up to 1.2Gbps (4x4)	5	Graphic card	NVIDIA GeForce GT 650M	GeForce GTX 980	GeForce GTX 980
6	PIRE	<20dBm (PIRE)	6	Screen size,	15 Inch	17 Inch	17 Inch
7	LAN interface	0/100/1000Mbs RJ45 LAN	7	Wireless	Airport Support 5GHz	Linksys Cisco Support 5GHz	Linksys Cisco Support 5GH
8	Dimension (W X D X H)	(28mm x 175mm x 119 mm)	8	LAN	Support 1Gbps	Support 1Gbps	Support 1Gbps
9	Maximum computer per-wireless network	50-70 nodes	9	Resolution, Pixel depth	2880 x 1800 Retina, 32 bit color	2880 x 1800, 32 bit color	2880 x 1800, 32 bit color

4.6.2 Experimental results

In the first experiment, we provide series of videos with different characteristics, the characteristic of the videos are available with vary bitrates, resolutions, frames rates, contents motions and etc., as shown in Table 4.2. The 2000 frames of source raw video under different profiles are encoded by using FFMPEG and each encoded video labels with an ID. IDs are described as ID_i , where i = 1 to 8. For instance, BigBuckBunny as low motion video content and StartWar as high motion video content are encoded [142].

The characteristics of these videos are 2k and 4k resolutions, 30 and 60 FPS, with variable bitrates. The observation of availability of QoS values for this experiment included 280Mbps of bandwidth, 3ms for two ways delay, 0.001 percent of packet loss and 0.001ms for the jitter. The main server streams the videos to the heterogeneous clients over multicast channel. In this scenario, according to Figure 4.2, we select only three devices to capture the transmission information. Therefore, the distances of PC1 and the MacBook are 5 meters away from the main access point. PC2's distance is 10 meters from the access point. All received videos are saved on the clients' sides database. To evaluate the videos streaming quality, the objective metrics of APSNR, SSIM, MSAD and VQM, are selected to observe the degradation of QoE.

Table 4.2. Encoding features.

Feature	Encode version	Encoding quality	Bit rate control mode	Bitrate	Resolution	Frame rate	Buffer level	GOP length	Internal bit depth	Video motion
ID_i	Lib-X26x	Profile	Dynamic Or Static	Kbps	HD, 2k, 4k	24, 25, 30, 60	Second	IBBBP	8, 16, 32	Low, High

The observation results of objective assessment (typically, compared to the original video) allows objective comparison of videos over wireless network and video quality is a characteristic of a video passed through a video transmission as depicted in Figures 4.3, 4.4, 4.5, 4.6, 4.7, and Figure 4.8. The figures show that, the characteristic of the videos and the devices have huge impact on the output of objective metrics results.

Generally, videos characteristics with high dynamic motion frames recorded high oscillation lines than low motion videos content for each metrics of APSNR, SSIM, MSAD, and VQM, as depicted in the figures. And the curves lines in high motion videos are degraded very fast and number of frames suddenly is dropped for both frame rates 30FPS and 60FPS. Thus, the high values of these objective metrics demonstrate better quality of the video also, PC1 and PC2 have same characteristics, but the results are changed according to device availability in all cases.

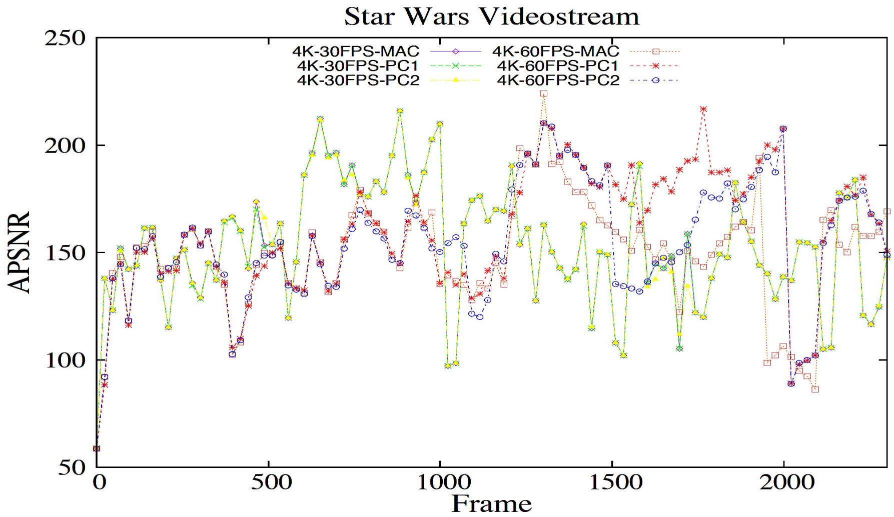


Figure 4.3. APSNR evaluation for Star War video.

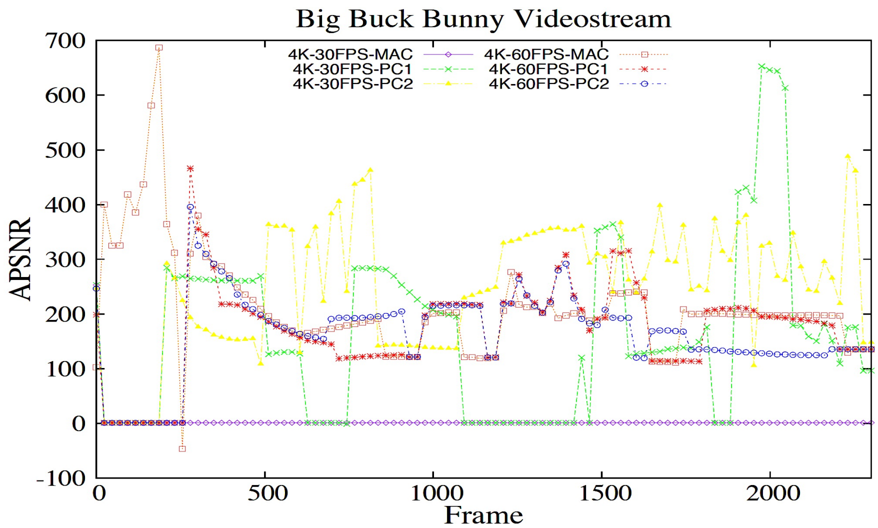


Figure 4.4. APSNR evaluation for BigBuckBunny video.

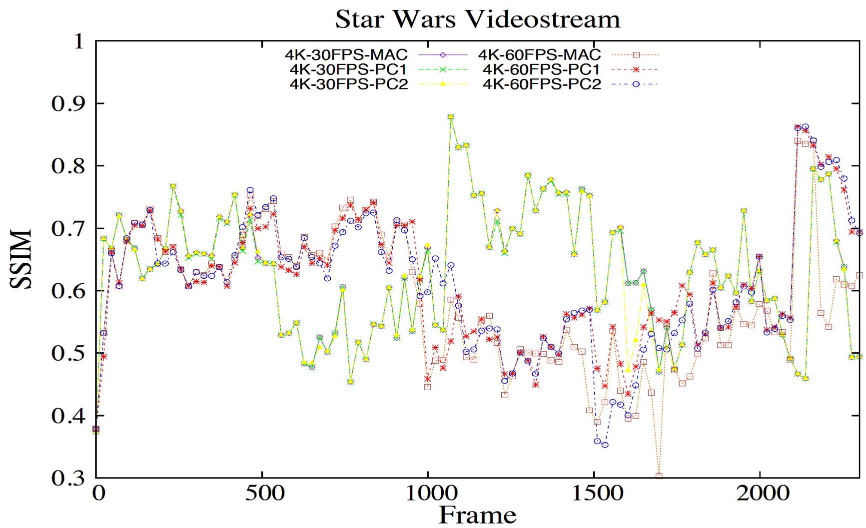


Figure 4.5. SSIM evaluation for Star War video.

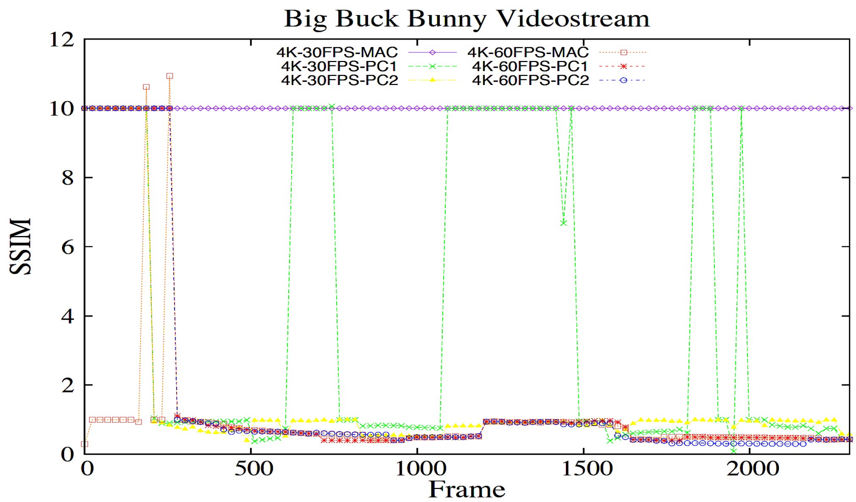


Figure 4.6. SSIM evaluation for BigBuckBunny video.

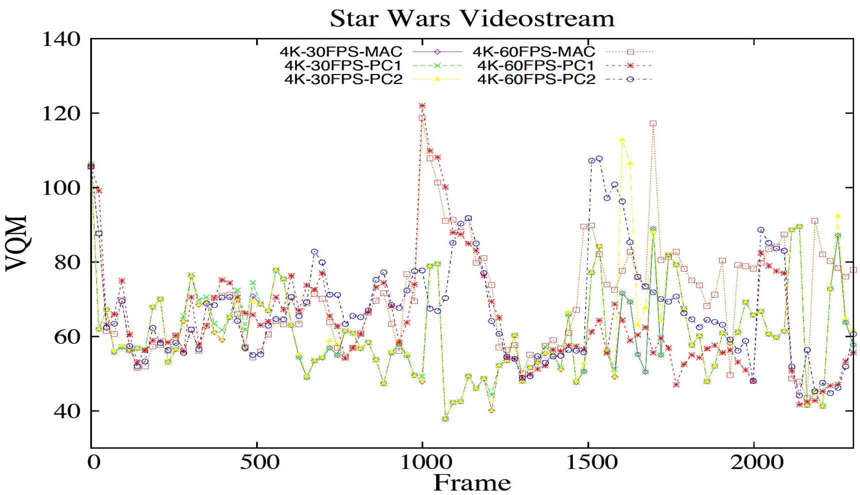


Figure 4.7. VQM evaluation for Star War video.

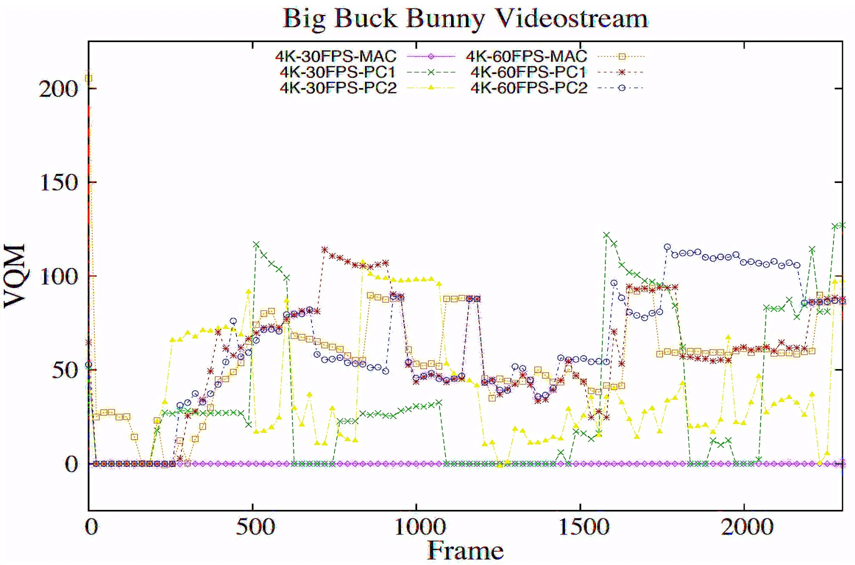


Figure 4.8. VQM evaluation for BigBuckBunny video.

In the second experiment, in order to observe the correlation between the impact of the QoS on the objective metrics, we shape the network delay, jitter and packet loss for the different value. In this scenario, we find both objective and subjective QoE assessment. The subjective evaluation, 80 university participants are involved and they are mixed in gender and their ages between 21-45. We prepared some questions to all participations, which are included feeling of participants to comfortably give feedback on the testing videos and whether they have difficulty of vision of distinguishing between images colors and perceive video degradation. The score of the MOS measurement initiates from 5 to 1, while 5 is indicated excellent satisfactory of the perceptual video quality and 1 points to very annoying. Therefore, the test has been conducted in a controlled laboratory environment. It took 8 weeks to complete the entire subjective test. A particular observation of this experiment is presented in Table 4.3.

Table 4.3. Objective and subjective evaluation.

Video characteristic	QoS		Assessment Metrics				
Extract Features	Parameters	Values	APSNR (μ)	SSIM(μ)	MSAD(μ)	VQM (μ)	MOS (μ)
ID_1 = BigBuckBunny Encode version:**X264** Encoding quality:**4.1** Bit rate control mode: **Dynamic** Bitrate: **Variable** Resolution: **2k** Frame rate: **30** Optimal buffer level: **4000** GOP length: **IB….P** Internal bit depth:**32** Video motion: **low**	Delay On way (msec.)	**100**	**99,910**	**1**	**0,009**	**0,006**	**5**
		250	48,992	0,866	2,008	2,412	2
		500	25,8043	0,722	5,299	4,56	2
		750	19,363	0,605	11,848	7,244	1
		1000	9,4	0,4	17,555	10,01	1
	Jitter (msec.)	**0.01**	**26,318**	**0,736**	**7,738**	**6,266**	**2**
		0.05	20,277	0,630	1,659	7,818	1
		0.10	14,555	0,5	14,54	8,65	1
		0.50	8,555	0,38	16,87	10	1
		1	1,789	0,1	20	12	1
	Packet loss (%)	**0.01**	**84,704**	**0,987**	**0,111**	**0,403**	**4**
		0.05	45,039	0,896	2,437	2,519	3
		0.10	27,517	0,814	4,631	4,791	2
		0.50	18,470	0,624	6,68	7,681	1
		1	15,322	0,515	8,930	9,386	1

We calculated the mean average for both objective and subjective measurements as expressed in Equation 4.8.

$$\mu = \frac{1}{n} \sum_{i=1}^{n} (xi) \qquad 4.8$$

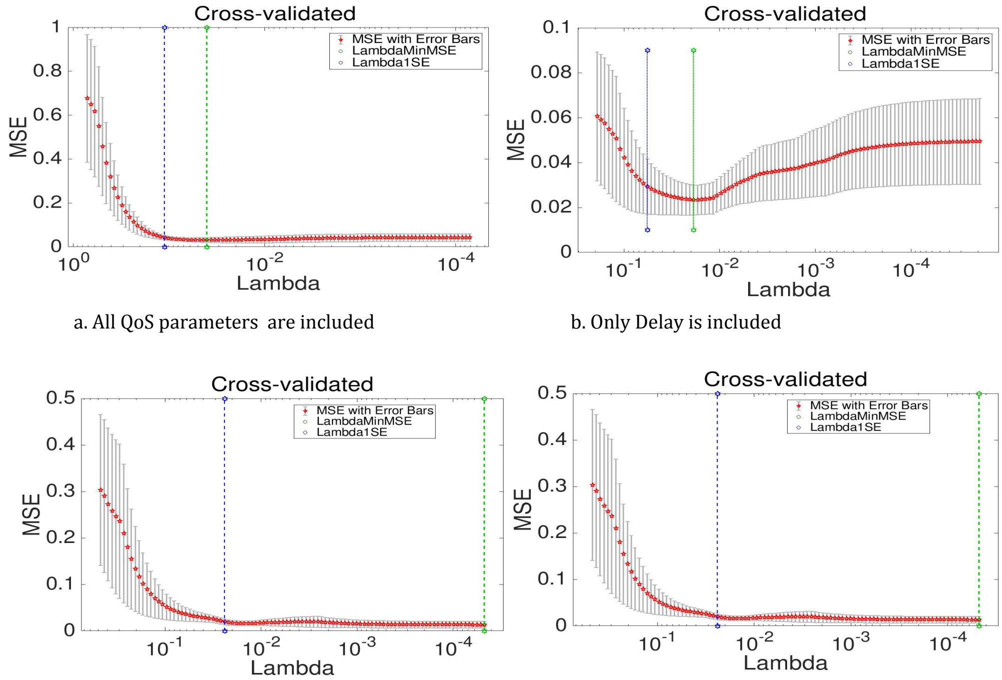
The table shows the correlation between the parameters of QoS and the objective and the subjective metrics. The realized tests for the video ID_1 are depicted in Table 4.2, when the delay equals to 100ms the perceived video quality is imperceptible. When the packet loss is 0.01%, the users are still perceptible for the video but not annoying. However, the jitter of 0.01 is degraded the quality of the video to poor level and the users are annoying. Therefore, when the characteristic of the video escalates to higher frame rate and higher resolution, the MOS becomes degraded, however, the video (ID_1) with frame rate of 60 for delay 100ms recorded high MOS score, when the delay and jitter are reached to 250ms and 0.01 respectively, the quality of the video presents poor quality and slightly annoying.

The mapping is sketched between both types of measurements, objective metrics and subjective metric. This is to compute the mean square error and subsequently for the mapping of the objective metrics to the MOS scale. Although, we extract the restriction parameter among the delay, jitter and packet loss, which is highly affected the results of the QoE. For this purpose, the Lasso regression is used to predict MSE and find the minimum rate of errors, when the objective metrics are used to assess QoE of the end-users for different QoS parameters.

In Figures 4.9, the relationship between λ and the cross-validated mean square error (MSE) is shown. Each of the red dots shows the MSE. The vertical line segments stretching out from each dot are error bars for each estimate. The line on the right identifies the λ value that minimizes the cross validated MSE. The line on the left indicates the highest value of λ whose MSE within one standard error of the minimum MSE.

Figure 4.9 (a) depicts the minimum error rate recorded when the system use entrance parameters of characteristic of video, all parameters of QoS, and characteristic of devices. However, results are different when each QoS parameter separately is validated by LASSO as shown in Figure 4.9 (a), 4.9 (b), and 4.9(c).

According to Table 4.3, the output of LASSO regression for different parameters is shown. As shown in the table, the model learns from different restriction values of QoS parameters i.e. Bandwidth, delay, packet loss, and jitter, are affected on delivery results, the restriction value of the QoS means the restriction parameter has higher effect on the results then other parameters. In our experiment as shown Table 4.3, we calculated lasso regression for each restriction values of QoS. When the model trains all entry parameters as depicted in Table 4.4. The minimum error for that experiment is 0.0036; however, when the packet loss has effect on the degradation of QoE, the minimum error is very high which can be estimated by 4.7237e-05.



a. All QoS parameters are included

b. Only Delay is included

c. Only jitter is included.

d. Only packet loss is included.

Figure 4.9. QoE prediction based on Lasso Regression.

Lastly, to demonstrate the behavior of proposal regression model, we evaluated it using other different types of regression models linear models (Ridge and Lasso regression), Support Vector Regression (SVR) and artificial neural network (ANN).

Table 4.4. Different QoS input to find out regression.

	QoS Parameters			
LASSSO Regression	All (Delay, Jitter , packet loss)	Delay	Jitter	Packet loss
	LambdaMinMSE: 0.0036	LambdaMinMSE: 0.0401	LambdaMinMSE: 0.0187	LambdaMinMSE: 4.7237e-05
	Lambda1SE: 0.4983	Lambda1SE: 0.1116	Lambda1SE: 0.0572	Lambda1SE: 0.0241
	IndexMinMSE: 30	IndexMinMSE: 69	IndexMinMSE: 75	IndexMinMSE: 1
	Index1SE: 83	Index1SE: 80	Index1SE: 87	Index1SE: 68

For the ensemble methods, feature normalization was not required, but we preprocessed the features for all regression models by mean subtraction and scaling to unit variance. Note that we computed the data mean and variance in the feature transformation step using only the training data. For each of the regression models, we determined the best parameters using 10-fold cross validation on the training set. This process was repeated on all possible train/test splits.

To demonstrate the overall improvements delivered by the learned regression models, we also compare lasso regression to other regression models; we use MATLAB for validation results separately (see Table 4.5). In this case, the Lasso regression achieved the highest average performance than other regression models. The performance of the Ridge is higher than ANN model, while SVR yielded the worst performance across all regression models. Clearly, the predicted QoE based on lasso regression significantly improved prediction future QoE with presenting minimum error rate.

Table 4.5. Comparison between Lasso and other models.

Regression Model	Min Mean Square Error
LASSO	0.0036
Ridge	0.0098
ANN	0.3016
SVR	0.6012

4.7 Proposal of QoE estimation algorithm

The management algorithm proposed for this multicast network environment is based on the process of QoE estimation based on Lasso Regression. To do this, the multicast server requests QoS parameters (Delay, jitter, and packet loss) of different parts of the network architecture according to Figure 4.1, which depicted the main points to capture information of QoS.

The main server also captures information of the video's characteristics before it is streamed over different channels.

The multicast server in order to obtain the information of the delivered video to their clients, the guarantee channel is established between group of clients and the server. GOP of the video is sent to the server from the group of users. The information of GOP of the video is saved in the server database; it will be used to predict QoE. From the monitored activity of the network, the system estimates the QoE. If the result obtained is within a limit range, that guarantees the QoE of the user, the system management of the QoE does not take any action and it just continues with the monitoring of the data. Whether, the result exceeds a limit with which it does not the QoE of the user is satisfied, then changes will be made in the system to improve the service of video streaming. These actions can be for example: increase the bandwidth of the channel band, make changes in the video bitrate, and frame rate and video resolution. All these operations are carried out from main server. Detail of the algorithm is explained in Figure 4.9. In the case when the QoE feedback reveals degradation of the perceptual video's quality, the proposed algorithm can manage the video stream to adaptive streaming according to the feature prediction.

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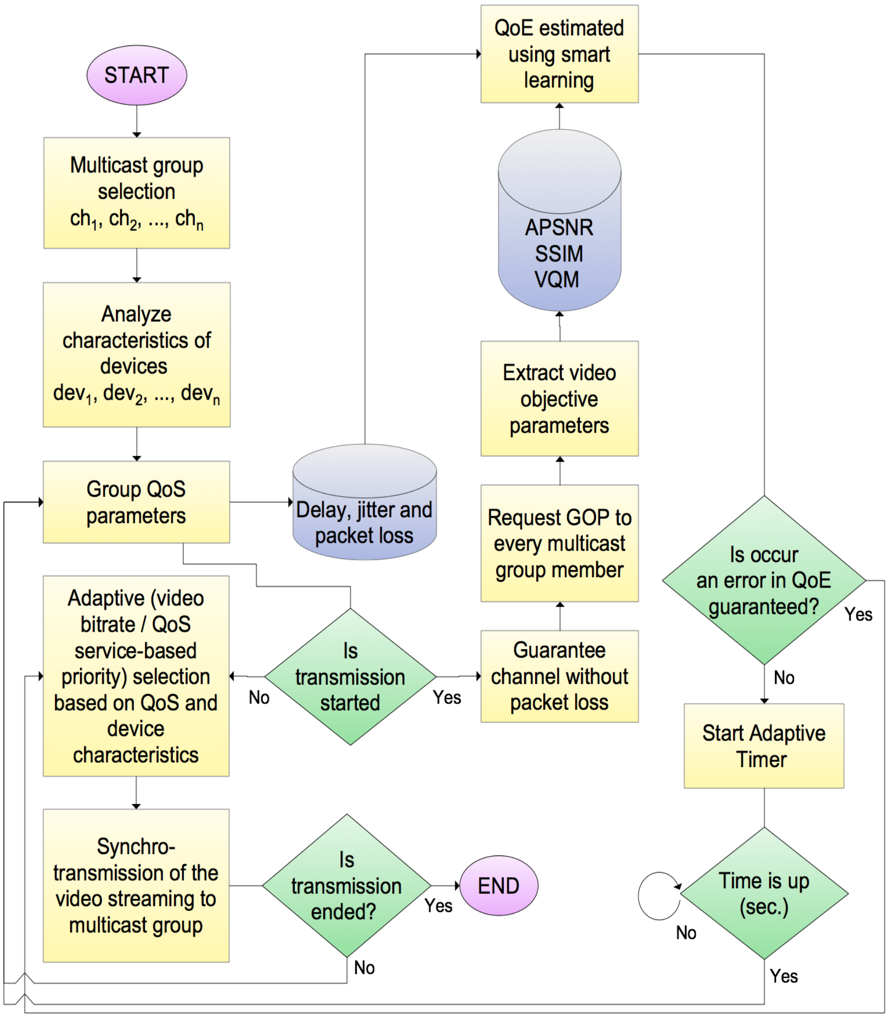


Figure 4.10. The proposed algorithm.

As we have just seen, QoE can be analyzed from two points of view: On one side we have the degradation of the QoE due to the network parameters and on the other we have the degradation of QoE due to user parameters. We need demonstrate that there is a clear correlation between QoS and QoE to justify the use of an inductive method for estimating the QoE whose input parameters are QoS parameters. In subsection 4.2.6, a dataset is provided to demonstrate the influence of QoS parameters on the degradation of the QoE

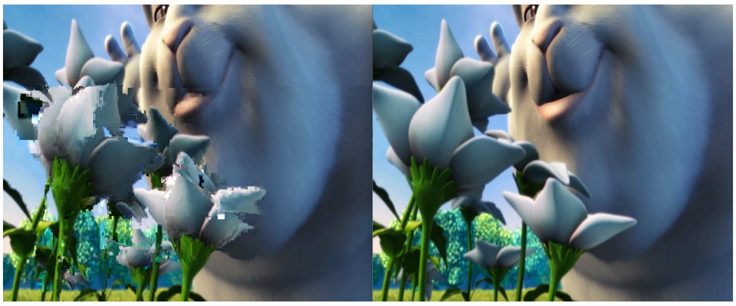
4.8 Benchmark comparison

In order to show the comparison between the traditional multicast streaming and with applied our algorithm, we provide subjective metric to evaluate performance of the algorithm, in this case, DMOS subjective metric is used to observation of the results, in order to apply DMOS, we provide a methodology test, which included 10 people in order to analysis the out of the videos.

In this case, eight videos are selected to provide the experiments, these videos have different characteristics as explained in Table 4.2. The evaluation results are presented in Table 4.6. And the difference between image qualities of the couple approaches is shown in Figure 4.11.

Table 4.6. Evaluation result of QoE Prediction algorithm.

Characteristic of the videos	DMOS- Video streaming based on non-adaptive approach	DMOS-Video streaming based on adaptive approach
ID_1	2	4
ID_2	2	4
ID_3	2	4
ID_4	1	4
ID_5	1	3
ID_6	1	3
ID_7	1	2
ID_8	1	2



a) BigBuckBunny video streaming.

b) Star War video streaming

Traditional approach Adaptation approach

Figure 4. 11. Comparison between traditional and adaptation approach in multicast video streaming.

4.9 Chapter conclusion

This section is described the QoE of multimedia streaming applications in wireless networks.

Therefore, from the obtained results, QoE can be analyzed from three points of view. On one side, we have seen that, the degradation of the QoE due to the network parameters and characteristic of the media. On the other side the degradation of QoE due to the user parameters. We need to demonstrate that there is a clear correlation between QoS and QoE to justify the use of an inductive method for the estimation of the QoE.

In order to develop the algorithm, we investigated the correlation between impacts of the video's characteristic, network parameters and device capacity on the QoE of the end users for perceptual video streaming. A smart learning model based on Lasso regression is proposed to predict the accurate and the efficient assessment of the QoE. The model predictor is used the empirical measurement based on the obtained values of the metrics of the network channel and the objective metrics. The proposal prediction model assesses QoE according to fidelity of results and fast and easy implementation. So, as a result, in the case when the QoE feedback reveals degradation of the perceptual video's quality, the proposal algorithm can manage the video stream to adaptive stream according to the feature prediction and this leads the provider to provide better QoE to the end-users.

Chapter 5. Proposed Methodology Design for HTTP adaptive streaming

5.1 Introduction

The assessment of quality of adaptive multimedia streaming is a great interest to the telecommunications companies that focus to increase expectation of quality by providing adaptable video quality with network condition to end-users. Client's application client receives the video stream by using the adaptation logic. Adaptation logic provides the adaptive video streaming to clients by providing adaptable quality, which is appropriated with network condition. Assessment adaptive video streaming is crucial. On the one hand, using objective metrics to evaluate adaptive video streaming is hard because of different representations of the same video is presented to the users. On the other hand, many researches are investigated on technology adaptive streaming therefore validation of the performance objective QoE models from their proposal is not enough sufficient. Some studies have lack description on QoE evaluation for adaptive video streaming. Deterministic network behavior, such as bandwidth, delay and packet loss, all of them together or separately are high affected on QoE, in the term of QoS many researches are only focused on bandwidth to evaluate QoE of HTTP adaptive streaming. And some parameters such as video characteristic, initial delay, switching, strategy stalls, and chunks size, are not considered as essential requirements to evaluate QoE.

In this section, we provide and develop a methodology to evaluate subjective QoE in HTTP adaptive video streaming. A statistical technique is considered to attempt mapping between the impact factors and QoE. From the correlation a decision deploys to define which parameter of the network behavior influenced highly on the results of perceptual quality. For this case subjective and objective approach is considered to evaluate the QoE. A relation is correlated between subjective and objective metrics in order to find the performance of the interaction and accuracy of QoE prediction in HTTP adaptive video streaming.

5.2 Influence factors on QoE for HAS-client

According to the studies was conducted in Section 2.5. There are many factors influenced on multimedia QoE. Although, the degradation of behavior of network parameters has produced the factors to influence the perceptual quality of HAS. In HTTP adaptive video streaming (HAS), the factors are affected on the perceptual of end-users available in different aspect. Which are classified into three category; initial delay, quality switch and stall frame. In HAS one of the characteristic of the video is chunks. The video includes various qualities and each quality includes a series of chunks, the structure and generate of video chunks are mentioned in the sections 2. They have same length in seconds. But the size can be changes according to the size of the I-Frame and complementary frames such as B-frame and P-frames.

Research questions in the section 2 are depicted on QoE of HAS service on the Internet. We provide the methodology and list the items will be solved in this chapter according to the effect of different parameters on the QoE in HTTP adaptive streaming service.

- Specify optimum initial delays when the users start playing back different chunk size of the video.

- Detect the effective of sharp switches and smooth switch on QoE according to video quality switch down from high quality to low quality and the video quality is periodically change (oscillation).

- Detect the effect of frames stall, stall at low-level quality and stall during switches.

- Assessment QoE in wireless scenario according these researches [32][100][103][118][143][144], they only focused on bandwidth parameter to find performance of QoE for HTTP adaptive streaming, HTTP use TCP, in reality, other parameters of QoS such as delay and packet loss are also affected on the performance of QoE.

- Assessment the QoE of HAS over heterogeneous devices such as PC, mobile and TV.

5.3 Experiment for subjective assessment

The most reliable way to determine the video quality is subjective assessment, which directly contact with humans eyes. To respect different real life network scenarios, special attempt for designing the experiment are taken into consideration, which included the choice of test methodology and evaluation methodology in order to assess service application of HTTP adaptive video streaming by human. In the following, details of these experimental setups are described.

5.3.1 Test methodology

5.3.1.1 Scenario of the tests

With the respect of real life scenarios, adaptive video streaming service users' consume adaptive streaming over heterogeneous devices, which included PC, TV, laptop and smart device or mobiles (See Figure 5.1). Although, these devices access to the service provider throughout different network connection point, it may be cable connection, wireless network connection, or cellular network. Therefore, most users get onto receive adaptive multimedia service over wireless networks with concerning of IEEE 802.11. It may home view, pedestrian in university campus or etc. We consider these scenarios to apply the tests and obtain dataset study. The topology of the network scenarios included the stable condition, where users were consuming video service at home (users use a device such as TV, laptop, or mobile). Later, users were consumed the service at mobility condition such as pedestrian.

We provided a real testbed setup, which covered fives modules. The main server was NGINX web server. Web service application DASH was hosted on it. The server provided encoding video and adaptive video streaming.

Network shaper, it is hardware device with equipped with Ubuntu operating system. The shaper provides shaping QoS parameters available downstream and upstream networking by prioritizing network resources and guaranteeing certain bandwidth based on predefined policy rules. It uses concepts of traffic classification, policy rules, queue disciplines and QoS. This is done in order to shape and control the network's uplink and downlink, delay, jitter and packet loss ratio (See subsection 4.6.1).

In addition to this, in order to shape automating the network parameters, script file created to run automatically on the shaper device when the clients start to watch the video, the all values of bandwidth, delay and packet were shaping according the real scenarios separately.

Network monitor system is based on Ubuntu operating system, which equipped with cacti open-source web tool network monitoring. The monitoring approach was necessary to extract network information by monitoring the HTTP request between client and server, therefore, the network tool was analyzed the performance of captured TCP and establish a correlation between each QoS parameters like bandwidth, delay and packet loss on the performance of the packets transfer between the server and the clients. After each video streaming session, a log file was generated on the client device, including capture information of the video, such as real time of computer, video time, from the comparison real time and video, startup delay, stalls, video bitrates in kbps and quality which is denoted by $Q_{i,i}$, the quality resolution may 4k, 2k, etc.

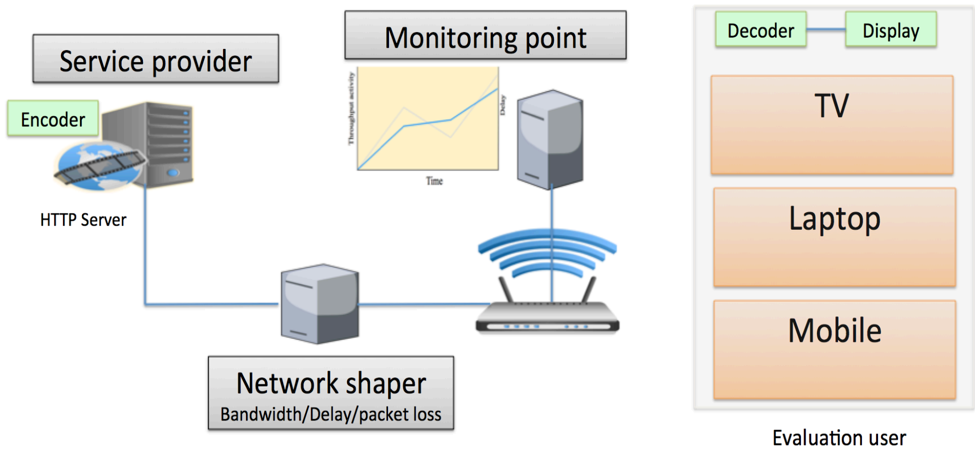


Figure 5.1. The subjecitve assessment testbed.

5.3.1.2 Materials for tests

We select large variety of video sequence in order to cover sufficiently targets of real-life medias and applications regarding to produce noticeably degradation in the process of assessment (See Table 5.1). Therefore, in order to provide accurate assessment, 180 seconds from the raw video of each selected video is encoded. This length is selected to generate higher number of segments and receiving better assessment from the evaluation users. Long sequence time can be fatigued of participants and short sequence time may be presented inaccurate results.

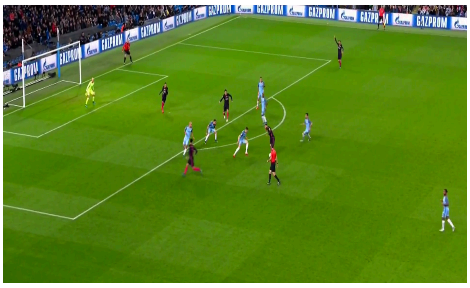
Generally, four kinds of videos are chosen with different characteristics, as shown in Table 5.1 and Figure 5.2. The quality resolutions were 4k, HD and SD. Each video was encoded into 6 representations as shown in Table 5.2 with x264 encoder to cover different quality levels. The choices of bitrate levels were based on the Netflix's recommendation. The test sequences are segmented with GPAC's MP4Box [145] with a segment length of 1, 2, 4, 6, 8, 10, 15, 30 seconds for the following reasons. First, we provide different network profile, in first case only network throughput variable. The delay of network emulator parameters was set to variable milliseconds corresponding to what can be observed within long-distance line connections or reasonable mobile networks, and thus is representative for a broad range of application scenarios. We used 4 network traces for each QoS parameter and the total network traces is 12 as shown in Figure 5.3 that are wide-ranging and representative including stationary as well as different mobility scenarios, such as pedestrian, car, train, etc. The average bandwidth of the network traces varies between 300 Kbps and 20000 Mbps covering all range of bitrates in the bitrate ladder (See Table 5.3).

Table 5.1. Characteristic of the sequences

Code	Genre	FPS	Characterization
1	Tears of Steel	30	High motion fast changing the relatively dark scenes; high disparity
2	Sport Football	30	Soccer; average motion; wide angle camera sequences with uniform camera panning, medium disparity
3	Star war Video game	30 60	Sudden motion High motion fast changing the relatively Dark and white scenes; high disparity
4	Big buck bunny Cartoon	30 60	Smooth motion of objects is dominant; Static background; very low disparity

a. Tears of Steel

c. Star War



b. Succor

d. BigBuckBunny

Figure 5.2. Snapshot of the video sequences.

Table 5.2. HAS representations for test sequence.

Quality level code	Resolution	Aspect	Bitrate (kbpss)
1	384x288	SD	300
2	512x384	SD	700
3	1280x720	HD	1500
4	1920x1080	HD	6000
5	2048x1440	2k	11658
6	3840x2160	4k	19684

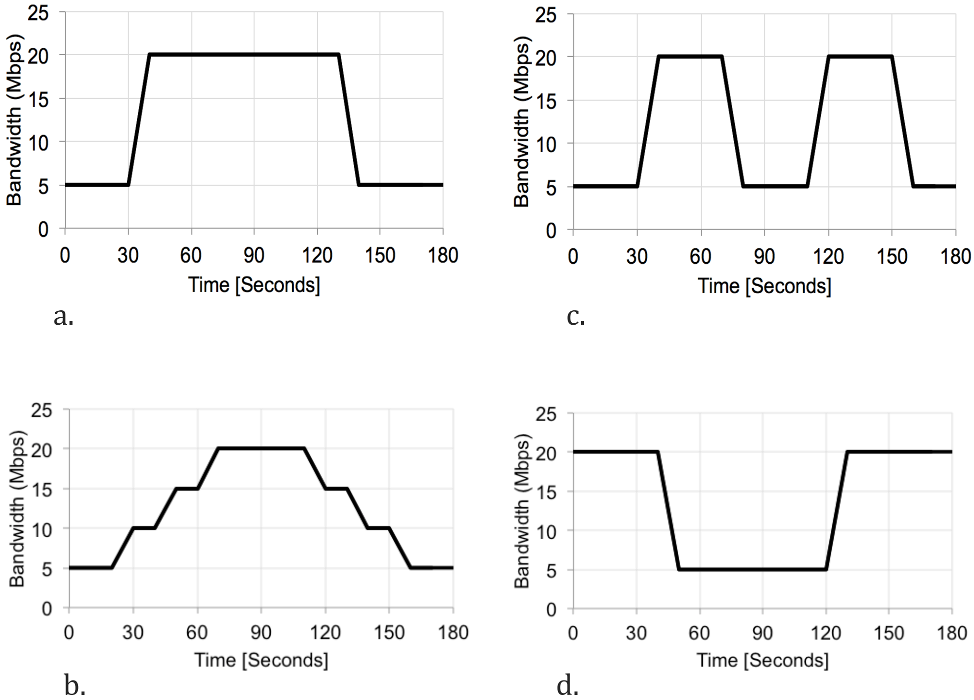


Figure 5.3. QoS profiles: a. Step up and step down, b. Flacutation down, c. ramp up and ramp down, d. Flacution up.

5.3.1.3 evaluation subjective methodology

In order to achieve reliable results, 34 observers selected to the tests. They were non-experts (naïve users) in the sense that they were not directly concerned with television picture quality as a part of their normal work and all of them had correct-to-normal sight. User information as name, occupation, gender and age are taken. Range of subjects was from 20 to 45 years, including 28 males and 6 females. The average range of the age was 25.

Absolute Category Rating (ACR) ITU-R [146] 5-point scale corresponding to the perceived quality is selected to give the participate feedback as shown in Table 5.3. The participants rate the quality of streamed video in three levels; a level for initial delays, other, for quality switching (sharp and frequent switches), last level, for stalling. Participates directly send their feedback to server database when the streaming of the videos are finished.

Table 5.3. Subjective evaluation method.

Session of streaming video	Vote	Vote sends to server

5.3.1.4 Data processing

Subsequently, four participants are removed based on the subject removal scheme, which is suggested in [147], resulting in 30 valid participants. After participants are removed from the scheme, Z-scores are linearly rescaled to lie in the range of [1, 5] (See Table 5.4). The MOS for each individual video is computed as the average of rescaled Z-scores, from all valid subjects.

Table 5.4. Subjective evaluation scale.

CODE	ACR
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

5.3.2 Result Analysis

According to the comments given in section 2, the performance of evaluation of the tests found for different chunk size, where the influence parameters of initial delay, quality switching and video stalling are essential metrics that are impacted on the QoE in HTTP adaptive streaming. Figure 5.4 and Figure 5.5 depict the effect of each metrics on subjective evaluation. According to the figures small segments are rerecorded higher MOS than large segments for both initial delay and video stalling. From segment 1 to 8 the users perceive the quality of the videos. However, the segments length 10 to 30 seconds the perceived quality become degraded and users can no satisfy with the receiving of videos.

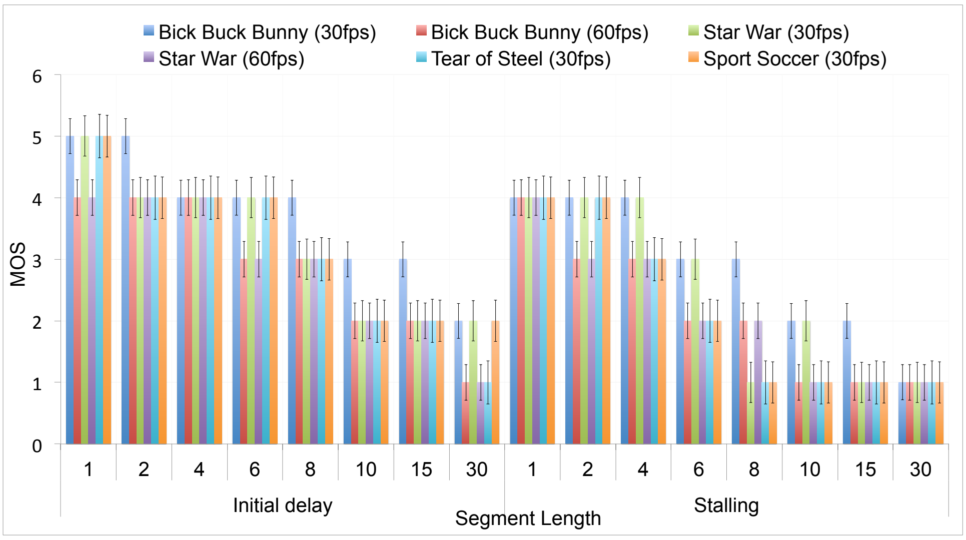


Figure 5.4. QoE evaluation based on the initial delay and video stalling.

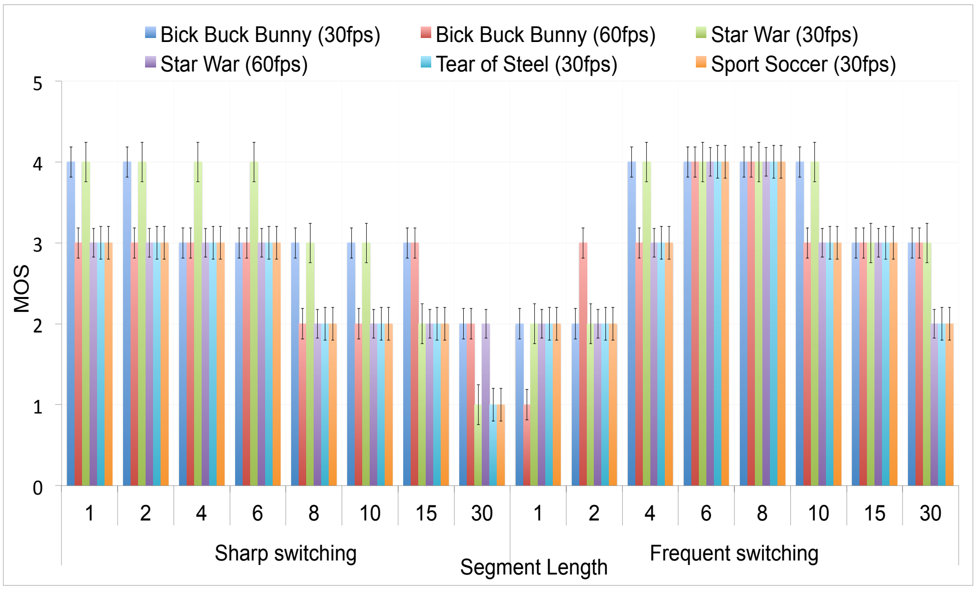


Figure 5.5. QoE evaluation based on the quality oscillation.

According to the Figure 5.5, the subjective evaluation found for sharp switching and frequent switching. The evaluation of the video quality according to quality oscillation has been changed due to characteristic of adaptation login, network throughput and characteristic of the videos.
Frequent switches are very high when the chunk size is small, this is the buffer length will not fill to display the video content on the device also, sharp switches are high in large segments because the large segment have higher code efficiency and users can perceive high to sharp switch from a quality to another quality.

Therefore, we find the average subjective evaluation for all videos, according to the effective metrics on human eyes. According to Figure 5.6, the initial delay is very short in small segments however the MOS value is high. From chunk size 8 seconds to 30 seconds, the initial delay very long the users can perceive high annoyance of adaptive video streaming. Also, video stalling is long when the segments lengths are 10, 15 and 30. Moreover, the sharp switch in small segment is very long and the recorded value for the MOS is low therefore the user feels very annoyed.

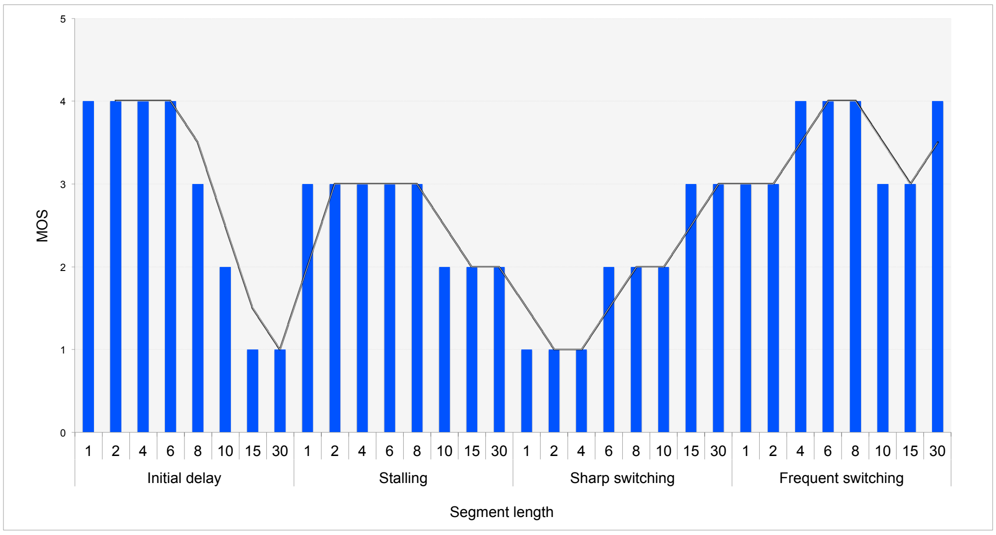


Figure 5.6. Average of the subjective evaluation for all videos.

In order to evaluate the performance of adaptive video streaming on different devices, we select the four kinds of sequences. For this experiment, also we choose the video chunk size with 2 seconds duration. Select the chunks size 2 seconds duration related to the above experiment has done for different segment length according to the proposed assessment methodology. Three types of devices are selected to apply the experiments, such as TV, Laptop, and mobile device [147]. Results as shown in Figure 5.7, the evaluation of subjective metric (MOS) using mobile device is higher than laptop and TV. This is because in mobile devices, users feel less oscillation of video quality, initial delay than other devices such as Laptop and TV, therefore, the characteristic of the video also change the result of the evaluation as shown in the figure for the high motion videos like Tear of steel and football match. (See Figure 5.7).

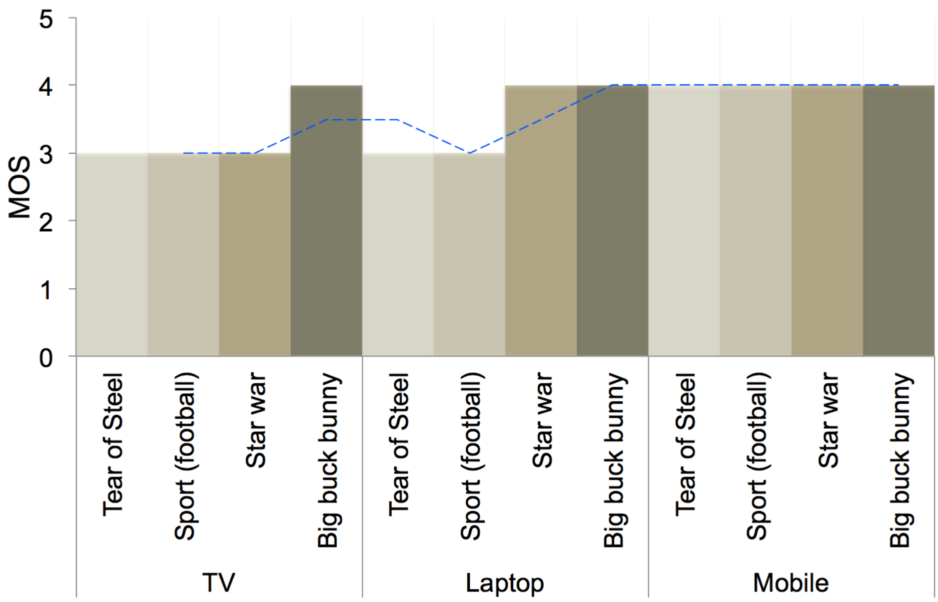


Figure 5.7. Evaluation the QoE on different devices when segment length is 2 seconds.

5.4 Experiment for objective assessment

Motivated by the observation and analysis provided in Section 5.2, we develop QoE model by incorporating the video presentation quality and the impact of initial delay, number stalling and GOP size events. Objective assessment in adaptive video streaming is complex approach. The video content is available with different quality and comparison objective assessment between the original video and the delivered video is hard. The client side may be rendered the video with different quality. Extract a variety of quality of same video is not easy even is not accurate because of the chunks of the video are decoded has different bitrate and different property during the playback time.

In order to obtain the objective assessment, at the beginning of the video preparation, the objective metrics for each quality of the video has been described according the case study. In this case, each video when prepared to being adaptive streaming has brief profile, which described the information of the video. The profile contained such buffer length, time, representation quality, bitrate and resolution, as explained in Table 5.2.

As shown in Figure 5.8, in order to provide the QoE objective assessment, we consider a method to evaluate objective approach. The approach precedes the objective evaluation for each video representation of adaptive video. For this scenario, a network profile is selected, which provided maximum availability of the network behavior where rate of the loss equals to zero. Therefore, the same sequences of the previous experiment have been used to evaluate the objective QoE as shown in Table 5.2. We selected three important metrics such as PSNR, SSIM and VQM for HTTP adaptive streaming [147]. As shown in Figure 5.8, according to Table 5.5 we find only SSIM objective metric for all representations of same video. We want to find out merely effective no-reference image quality assessment algorithm because of in the no reference algorithms, the metric can be used for speed up development process of real time video QoE monitoring and estimation.

Therefore, the objective quality score can either be embedded into the manifest file that describes the specifications of the video. Moreover, the obtained result is labeled to MPD file of adaptive streaming application (See section 2.3). The manifest or metadata file is transmitted to the client side such that its information is available to the client.

Client once reads the manifest file the information of objective metric can be read by the application client. The client requests the segments (GOPs) of video quality, in commonly used streaming protocols such as MPEG-DASH. The series of GOPs arrived to client side, at the beginning of process, the frames of the GOPs decoded and sent for rendering, and then other series of GOPs are requested and placed in buffer for rendering. Viewers see the last successfully decoded frame during the stalling interval.

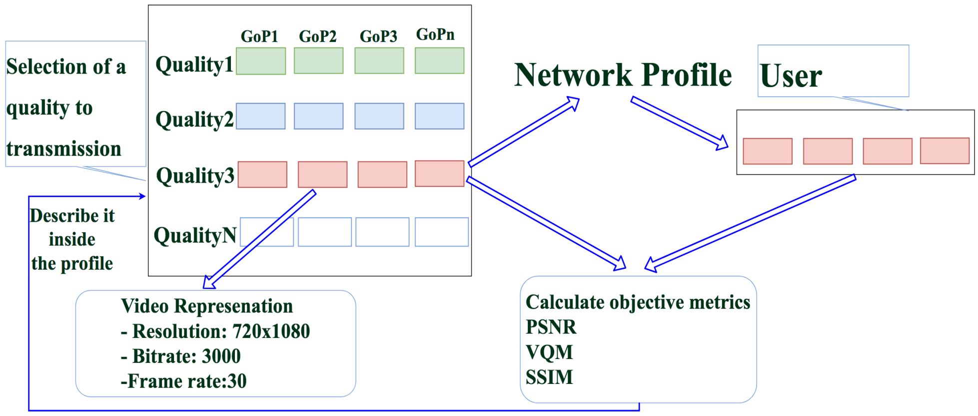


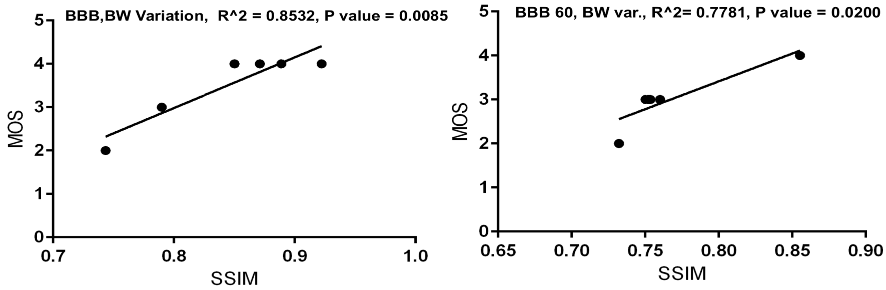
Figure 5.8. The objective evaluation method.

Table 5.5. Evaluation of objective based on SSIM

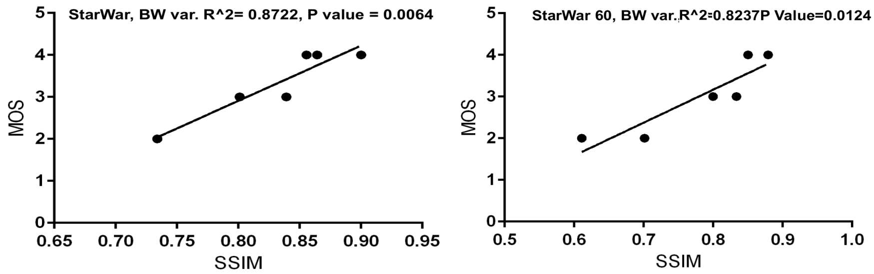
Quality level code	Resolution	Bitrate (Kbps)	SSIM
1	384x288	300	0.94539
2	512x384	700	0.97214
3	1280x720	1500	0.97658
4	1920x1080	6000	0.97898
5	2048x1440	11658	0.98567
6	3840x2160	19684	0.98768

5.5. Correlations between Quality of Service (QoS) and subjective and objective QoE

In order to determine the strength and direction of a relationship between QoS and subjective and objective QoE, we find Pearson correlation coefficient, which attempt to map between QoS and QoE. For this, we provide a massive dataset to predict QoE in HTTP adaptive streaming. According to both cases studied in previous sections the proposed method decides on the evolution of the QoE. As depicted in Figures 5.9, 5.10, 5.11 and Figure 5.12 R and P values are found for both subjective and objective approaches and the correction between the objective and subject are taken closer look and the variation of the parameters of the QoS is highly changing the results.

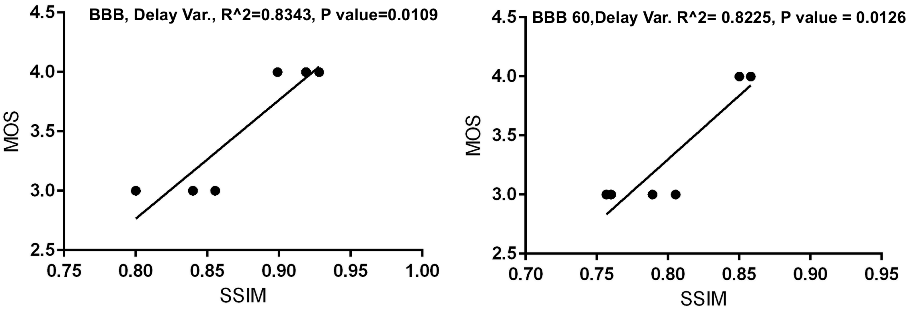


a) BigBuckBunny sequence.

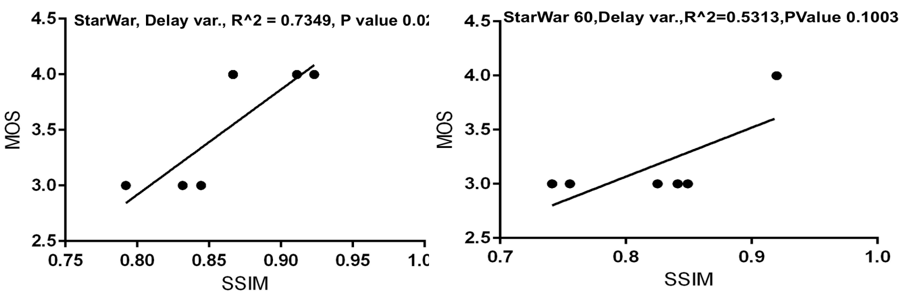


b) Star War sequence.

Figure 5.9. Correlation when bandwidth has a high effect on the QoE

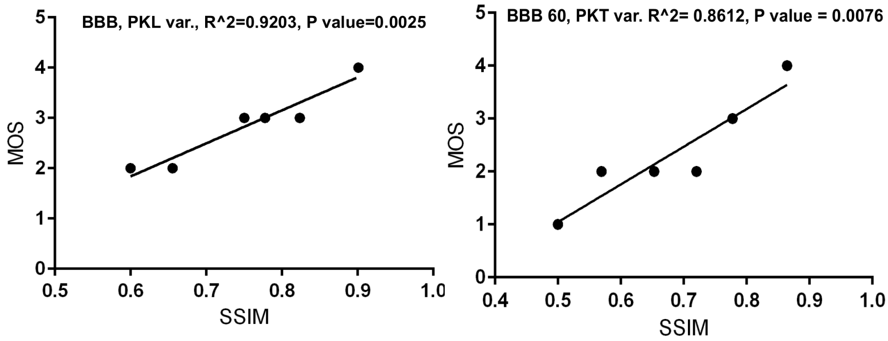


a. BigBuckBunny sequence.

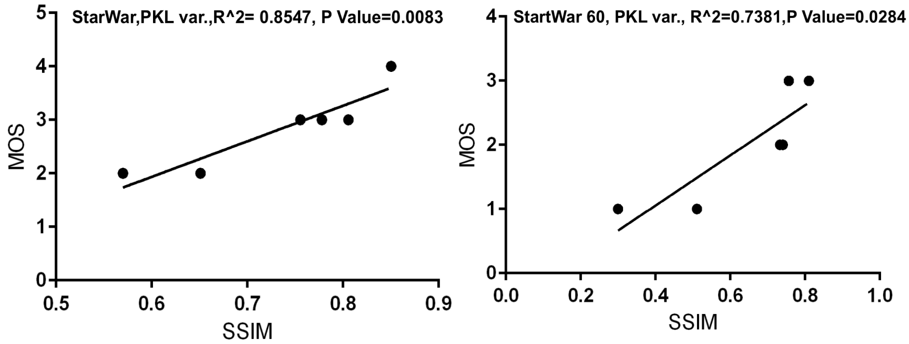


b. . Star War sequence.

Figure 5.10. Correlation when delay has a high effect on the QoE.



a. BIgBuckBunny sequence.



b. Star War sequence.

Figure 5.11. Correlation when packet loss has a high effect on the QoE

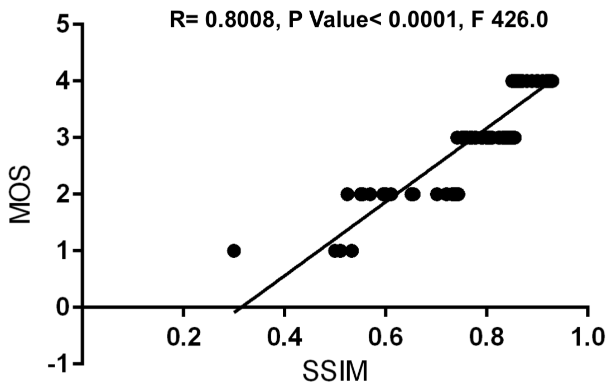


Figure 5.12. The correlation based on the all parameters of QoS.

5.6 Chapter Conclusion

We have studied the influence factors, which are affected on human visual QoE of adaptive video streaming over HTTP. We proposed a methodology to evaluate the QoE. The methodology is based on the different observation metrics: such as chunk size, initial delay, quality oscillation, video stalls. In order to provide the test experiments, different sequences are used to provide the accurate evaluation. Subjective and objective metrics are used to evaluate the QoE. The subjective experiments reveal some interesting relationship between chunks sizes and the impact of stalling, switching and initial delay. Therefore, the objective metrics are carried out to find the evaluation of QoE. Thus, statistical correction is employed to depict the relation between QoS and QoE and relation between subjective and objective QoE. From the correlation approach, our method is the accurate approach to evaluate QoE of HTTP adaptive video streaming and the restriction values of QoS parameters have a huge impact on the prediction of QoE.

Chapter 6. QoE optimization

6.1 HAS QoE optimization in Wi-Fi network

6.1.1 Introduction

Adaptive video streaming providers take advantage of the HTTP protocol to deliver video contents to the applications of the end-users. HTTP adaptive bitrate streaming technology is used for adapting the video quality to the current network conditions in order to deliver the best possible video quality. In this technology, video content is encoded into different bitrates and each bitrate is packetized into small segments. The address of these segments is defined in the XML file known as Media Presentation Description (MPD). Once the client reads the MPD, it decides which segment has to be requested to the corresponding server based on the available bandwidth. The HAS applications are used adaptation logic to adjust video resolution to network user conditions. However, an ABR player may select an inappropriate bitrate during playback, this is due to the lack of direct knowledge of access network performance, frequent user mobility and rapidly changing channel condition [147]. As a result, concurrent user's application attempt to playback higher resolution by requesting higher bitrate video streaming and quality some users are unpredictably degraded in the wireless networks. This caused to high network resource are used and end-user QoE becomes unsatisfactory.

Given that, the issue arises when more than one client requesting adaptive video streaming over a shared bottleneck wireless network, which leads to instability of video quality, unfairness between the video players and unallocated bandwidth utilization.

Moreover, a significant amount of investigations [63] on rate adaptation algorithms in video streaming have been proposed to provide a better decision on improving the playback video quality, reduce instability and provide fairness among users that share the same network.

Generally, in wireless networks, the users' throughput is degraded [148]. Many factors impact on the producing bottleneck throughput in wireless networks such as an increased number of connected wireless clients to the access point (AP), interference with other Wi-Fi signals and non Wi-Fi signal devices, moreover, the distance between the wireless clients and the AP, and the impact of the mobility of the users, which employ mobile devices to connect to wireless networks.

Meanwhile, most of the multimedia stream applications are using an HTTP application layer protocol over a TCP transportation protocol. Using HTTP in Internet has the advantage to eliminate almost all of the problems of NAT, and is traversal and firewall friendly. Thus, when multiple clients stream adaptive video over HTTP in the degraded wireless network (bottleneck connection), the performance of TCP for multiple clients is dropped. Moreover, there are many retransmissions of packets because of the increase of the packet loss rate of the connection. On the other hand, connections based on TCP on a shared channel lead clients to an unfairly playback of the adaptive video quality.

Allocating TCP throughput fairly among adaptive video streaming players in bottleneck connections provides a tradeoff between the perceived video qualities [104]. On the other hand, these clients can fairly playback video content, even when the rate adaptation algorithm of applications is not efficiently designed to provide fair resource utilization [105].

In this section, an algorithm model based on the level of TCP throughput management throughout is proposed using Software Defined Network (SDN) [105] for wireless network devices. It provides a better network resource usage and allocation of the available bandwidth to heterogeneous adaptive video streaming applications. This is done in order to provide stability and efficiency of network resource utilization.

6.1.2 Principles and architectural components

A software-defined networking (SDN) architecture defines how a networking and computing system can be built using a combination of open, software-based technologies and commodity networking hardware that separate the control plane and the data layer of the networking stack. Traditionally, both the control and data plane elements of a networking architecture were packaged in proprietary, integrated code distributed by one or a combination of proprietary vendors. The OpenFlow standard was recognized as the first SDN architecture that defined how the control and data plane elements can be separated and communicated with each other using the OpenFlow protocol. The Open Network Foundation (ONF) is the body in charge of managing OpenFlow standards, which are open source. However, there are other standards and open-source organizations with SDN resources, so OpenFlow is not the only protocol that makes up SDN.

In the SDN architecture, as shown in Figure 6.1, the splitting of the control and data forwarding functions is referred to as "disaggregation," because these pieces can be sourced separately, rather than deployed as one integrated system. The architecture gives the applications more information about the state of the entire network from the controller, as opposed to traditional networks where the network is application aware.

SDN Applications: SDN Applications are programs that communicate behaviors and needed resources with the SDN Controller via application programming interface (APIs). In addition, the applications can build an abstracted view of the network by collecting information from the controller for decision-making purposes. These applications could include networking management, analytics, or business applications used to run large data centers. For example, an analytics application might be built to recognize suspicious network activity for security purposes.

SDN Controller: The SDN Controller is a logical entity that receives instructions or requirements from the SDN Application layer and relays them to the networking components. The controller also extracts information about the network from the hardware devices and communicates back to the SDN Applications with an abstract view of the network, including statistics and events about what is happening.

SDN Networking Devices: The SDN networking devices control the forwarding and data processing capabilities for the network. This includes forwarding and processing of the data path.

The SDN architecture APIs are often referred to as northbound and southbound interfaces, defining the communication between the applications, controllers, and networking systems. A Northbound interface is defined as the connection between the controller and applications, whereas the southbound interface is the connection between the controller and the physically networking hardware. Because SDN is a virtualized architecture, these elements do not have to be physically located in the same place.

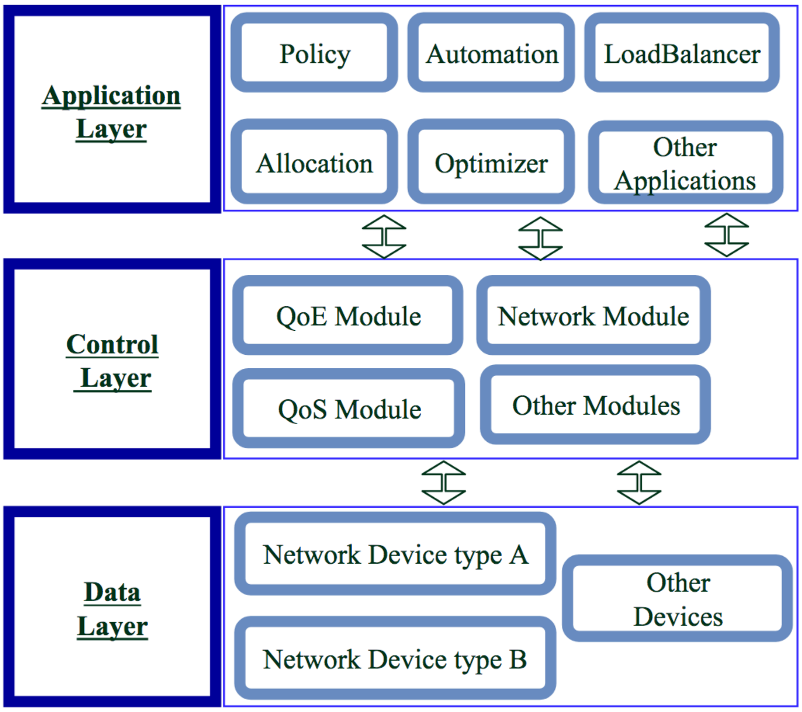


Figure 6.1. Architecture of SDN.

6.1.3 Proposed SDN-based Throughput Allocation Algorithm

The main issue of adaptive video streaming delivery is when many clients access through a network with limited available bandwidth. This situation introduces unfairness between clients causing a decrease in the overall QoE of the users. In this study a bandwidth allocation management algorithm is proposed and developed for wireless networks in order to enhance the perceived video quality. The flow chart of the proposed algorithm is observed in Figure 6.2.

The Software Defined Network controller collects the information of the clients requesting an adaptive video content in the wireless network. The detected information is necessary to determine precisely the bandwidth allocated to each client. The throughput required by the clients is estimated utilizing an SDN traffic-monitoring module. The peak bitrate requested by the clients is monitored to determine the highest video quality in order to operate with those values in the following bandwidth allocation process.

After obtaining the estimated throughput among the involved users performing the playback of the adaptive video streaming, the allocated bandwidth for each user is determined by sending a message to the content provider to allocate the bandwidth. A threshold parameter is employed to determine the maximum bitrate that clients can request in order to have a fair and smooth playback. The maximum threshold is calculated using Equation 6.1

$$\boldsymbol{\theta}_{\boldsymbol{max=\beta*Th_{total}}} \qquad 6.1$$

Where θ_{max} is the maximum threshold, β [0,1] is a parameter utilized to establish a conservative margin. The closer to 1, the less conservative, and Th_{total} is the total available bandwidth of the network.

To determine when should be changed the allocated bandwidth, the conditional presented in Equation 6.2 is evaluated.

$$\begin{cases} \textbf{if } \mathbf{Th_{clients}} > \boldsymbol{\theta}_{\mathbf{max}}, \textbf{Lower allocated BW} \\ \textbf{if } \mathbf{Th_{clients}} \leq \boldsymbol{\theta}_{\mathbf{max}}, \textbf{Maintain allocated BW} \end{cases} \qquad 6.2$$

Where $Th_{clients}$ is the estimation of the throughput required for all of the clients to playback the video with the highest quality.

To allocate the desired bandwidth for each client, Equation 5.3 is performed.

$$\boldsymbol{\alpha} \cdot (\boldsymbol{\theta}_{\mathbf{max}}/\mathbf{i}) = \mathbf{BW_a} \qquad 6.3$$

$\alpha \in [0,1]$ is a parameter employed to establish a safety margin. In order to avoid high competition between concurrent clients, α should be slightly lower than 1. i is the number of adaptive video clients and BW_a is the allocated bandwidth to be assigned for each client.

Test results conducted based on the evaluation obtained by the measurement of the behaviors of the videos when several clients request adaptive video content over a wireless network device are analyzed in the next section. It also gives the detail description of the Tesbed setup and the implementation of the involved parameters.

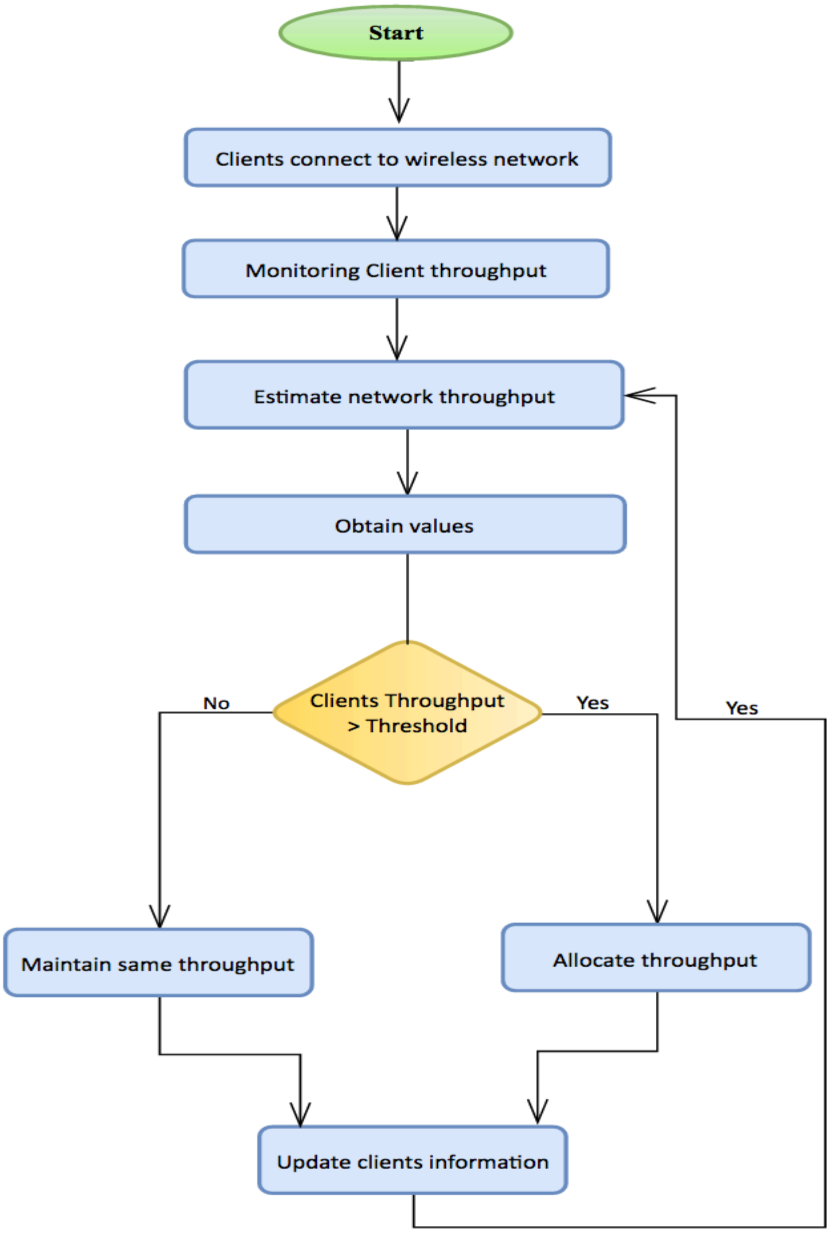


Figure 6.2. Flow chart of the proposed algorithm.

6.1.4 Testbed setup strategy

A platform is for conducting rigorous, transparent, and replicable testing of scientific theories, computational tools, and new technologies. The term is used across many disciplines to describe experimental research and new product development platforms and environments. In this work an experimental testbed that is based on real time video streaming is used to show the perceived quality of the video in the client side.

6.1.4.1 Testbed parameters

In order to measure the effects of client competition, a testbed is employed, which includes a main server as content provider, a cache server to hold the copy of video content and a network shaping to shape and allocate the bandwidth of the clients. These components of the testbed will be virtualized in a physical machine. Moreover, several clients have access to the video content through a wireless network. The SDN controller is associated with the wireless AP to obtain information of the connected clients. The topology of the testbed is shown in Figure 6.3. Two types of videos have been selected in order to obtain accurate results. The selected videos are the Big Buck Bunny and Elephant Dream raw videos.

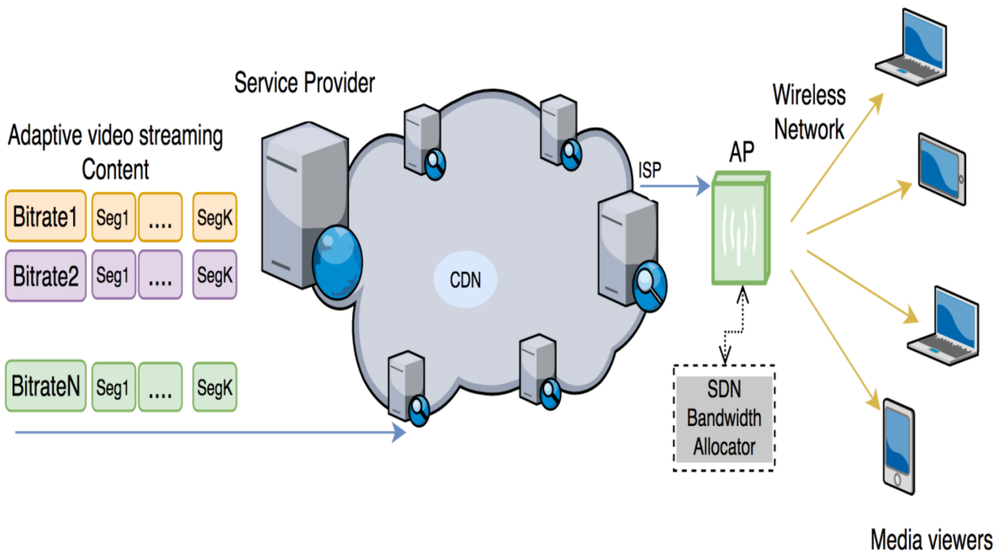


Figure 6.3. The Testbed scenario based on SDN

In order to provide DASH content with the varying bitrates, 2880 frames (120 seconds) have been encoded by utilizing x.264 and GPAC-MP4Box software. Furthermore, the duration of the chunk size for the employed segments is 6 seconds (in order to avoid high oscillation of video quality as explained in chapter 5). Indeed, each video is composed of 20 representations that consist of three levels of video quality. Those qualities can be classified as low, medium, high and high definition quality. Each level of quality provides distinct bitrates, which are sequenced. From low quality, which has a resolution of 320x240 with a bitrate of 47 kbps defined as the baseline profile, to the ultimate quality defined as a high definition profile, which has a resolution 1920x1080 with a bitrate of 4727 kbps.

6.1.4.2 Testbed implementation

Experimentation is very crucial to efficiently and accurately test and evaluate a network-based application. It can be a complex process when trying to experiment with an overlay application involving dozens of machines or more. Using the Internet as a testbed is not practical because parameters cannot be controlled. Setting up a hardware testbed is cost effective and complicated. Furthermore, overlay applications can have very different ways of connecting hosts to each other's and changing the network topology and network parameters of a hardware testbed is time consuming and subject to error. The virtualization technique to generate a testbed can save resources and ease manipulations. It is a proved method for reducing the equipment and space costs as well as the energy consumption of using physical hosts. The solution to overcome the above hardware constraints is thus to build a testbed able to set up virtualized networks. A virtual network uses virtual machines instead of physical hosts and connects them with virtual links in order to build a virtual network topology. The virtual machines of a virtual network can be hosted on one or several physical hosts depending on the number of virtual machines needed and the resources capacities of the physical ones.

To implement the proposed testbed, virtualized networks over Linux are used to virtualize part of the devices in our scenario. The video content provider has Ubuntu 14.10 with NGINX server version 1.8.1. The cache server is a mirror of the content provider. It has the same properties of a content provider. The network traffic shaper operates on a Linux-Ubuntu 14.10 that includes the shell script that automatically emulates the network clients' throughput link according to the bandwidth configuration file. Moreover, the physical wireless AP is a dual-radio 802.11n with data rates of up to 300 Mbps. The SDN controller is a physical machine with Ubuntu 14.10. It is connected to the APs, which are responsible of collecting the information of the wireless networks. The clients have a 2.4 GHz Intel Core i7 CPU, 8 GB 1600 MHz DDR3 RAM and NVIDIA GeForce GT 650M 1024 MB video card. DASHJs player is used as open source application to playback adaptive video streams.

6.1.5 Experimental results

In this subsection, some measurements of the effects of unfairness are presented when sharing a bottleneck as well as the results of the proposed method. Firstly, some measurements of the performance of two different videos are taken. Figure 6.4 depicts the performance of two players sharing a bandwidth of 10 Mbps. The first player starts requesting video segments at time equal to 0 seconds and the second player starts at time 5 seconds. As shown in the figure, the first player reaches the maximum quality from the beginning, but as the second player starts requesting for the maximum quality, the instability of the video quality begins resulting in a competitive state where both players try to reach the maximum quality. Figure 6.5 presents the measures of the second video. As the previous case, player one starts at second 0 and player two at second 5. The situation is similar to the aforementioned case. First player reaches easily the maximum quality and second player starts from a low quality until it reaches the maximum quality, when sufficient bandwidth is not available for both players and the instability begins. The variation of the buffer length for the clients' application is shown in Figure 6.6. The measurements have been performed for two cases, sharing an available bandwidth of 10 Mbps and sharing 20 Mbps. Generally, there is a high oscillation of the filling buffer for different available throughputs. The first player, when using 10Mbps of shared bandwidth, tries to reach the maximum buffer length at the 39th second and 68th second. When they are sharing 20 Mbps, although the first player is trying to reach the peak rate, results in the second player are the opposite, thus, the buffer length is filled for a few seconds.

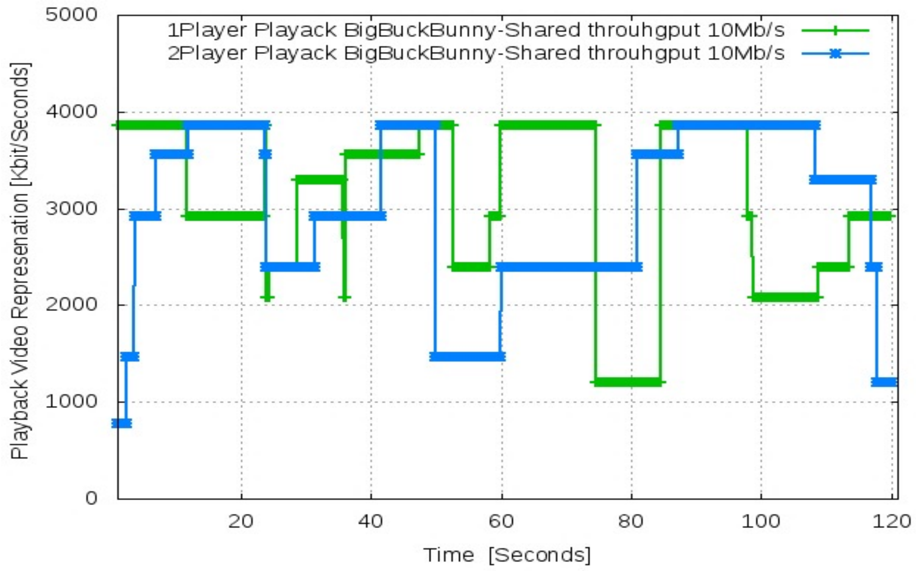


Figure 6.4. The experiment for the BigBuckBunny sequecne.

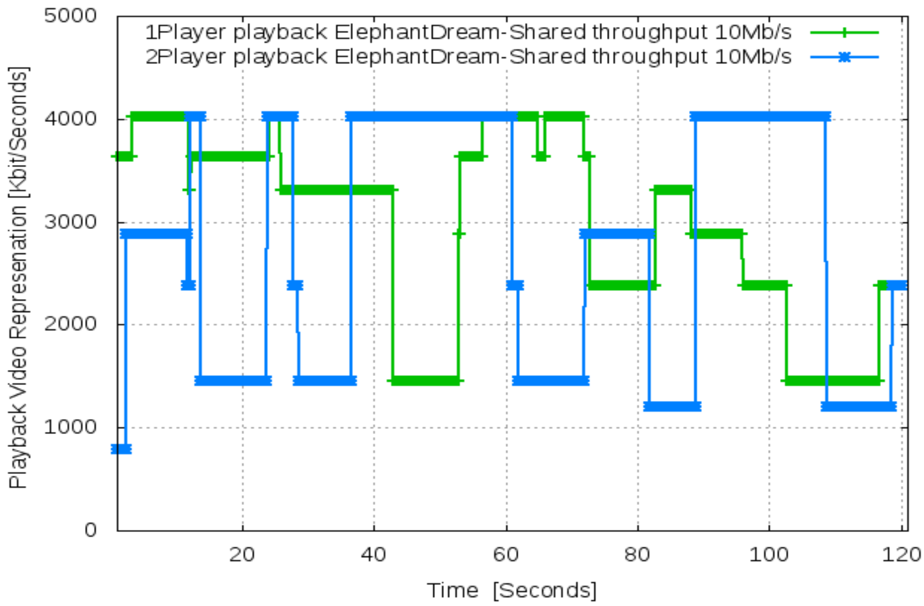


Figure 6.5. The experiment for the ElephantDream sequence .

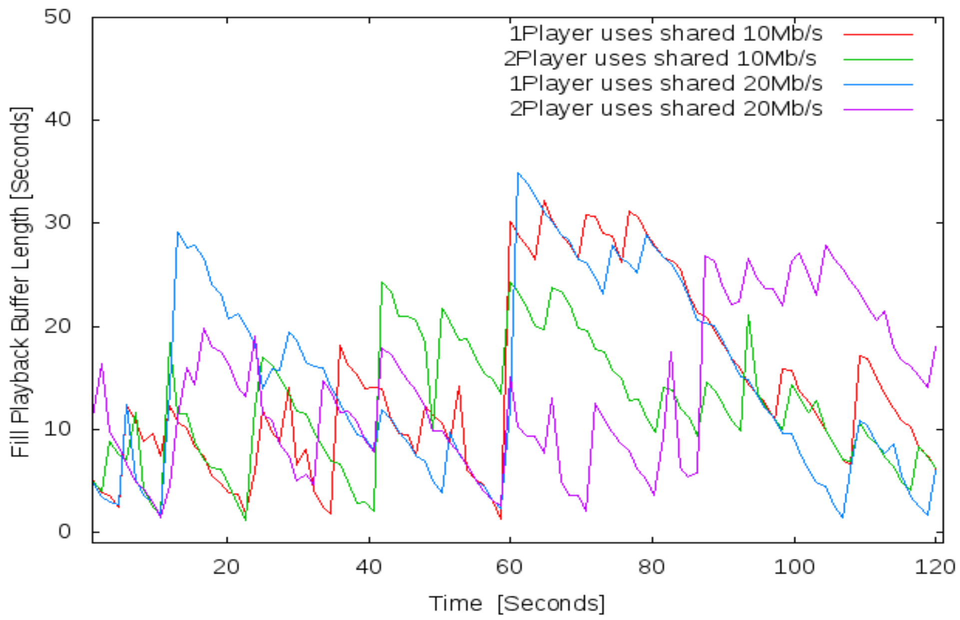


Figure 6.6. Oscillation of buffer length for different videos.

The aforementioned measures have been repeated with the implementation of the proposed method for bandwidth allocation between concurrent clients over a shared wireless network. When the players access to the service provider through the wireless network, they download video chunks into the application's buffer. The SDN controller detects the competition between concurrent clients by the throughput oscillation. After the information has been detected, the controller sends an information message to the content provider to give an alert (two clients have throughput competition). Then, the management allocation algorithm is applied to allocate fairly the throughput among clients in order to have a smooth playback. After allocating the throughput for the video players, the high degradation of video quality is stabled after the 20th second of the playback as depicted in Figure 6.7 and Figure 6.8.

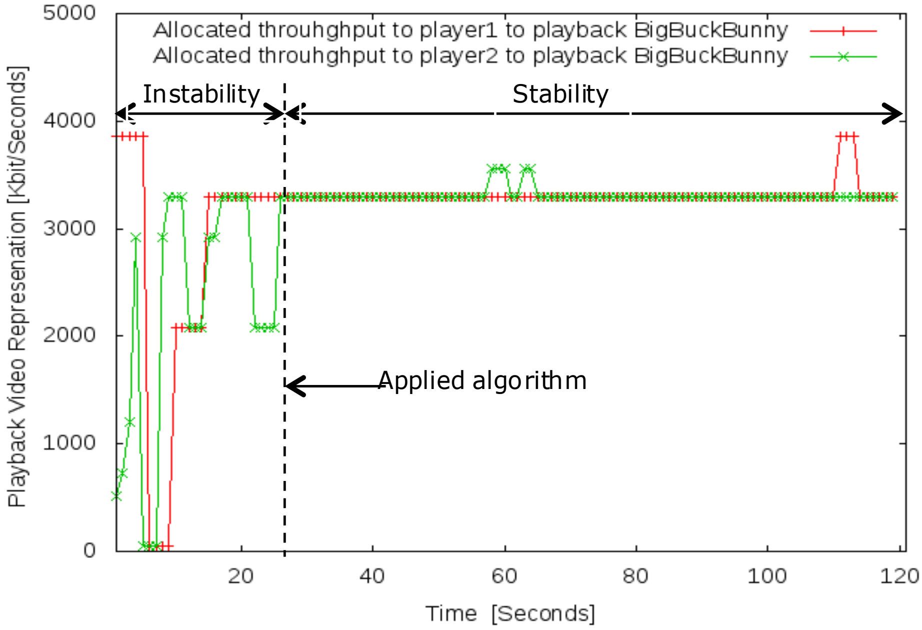


Figure 6.7. Applied algorithm for BigBuckBunny.

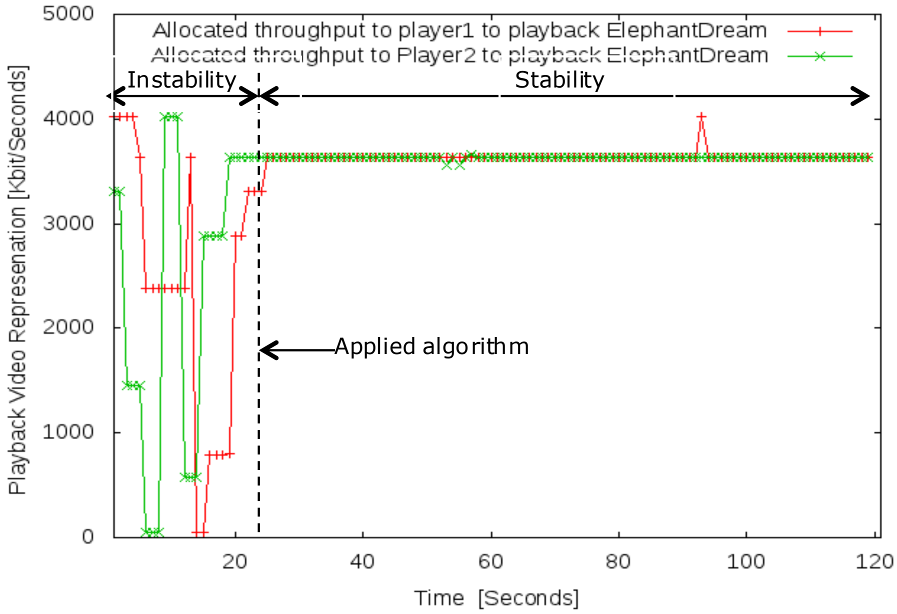


Figure 6.8. Applied algorithm for ElephantDream.

As a result, all players reach the optimum quality and stay mostly on that quality throughout the length of the video. The peaks that differ from the optimum quality occur due to the distinctive characteristics of wireless networks.

The buffer length of the clients was measured again after utilizing the proposed algorithm. The results are shown in Figure 6.9. It allocates the bandwidth for each client in a fair way, which results in a much more stable buffer than without bandwidth and employs the entire available bandwidth to retrieve chunks in order to provide the better QoE it can to the users. In this case, the range of the buffer length in seconds is not very high because the available bandwidth for downloading chunks will only fill the buffer length for that duration.

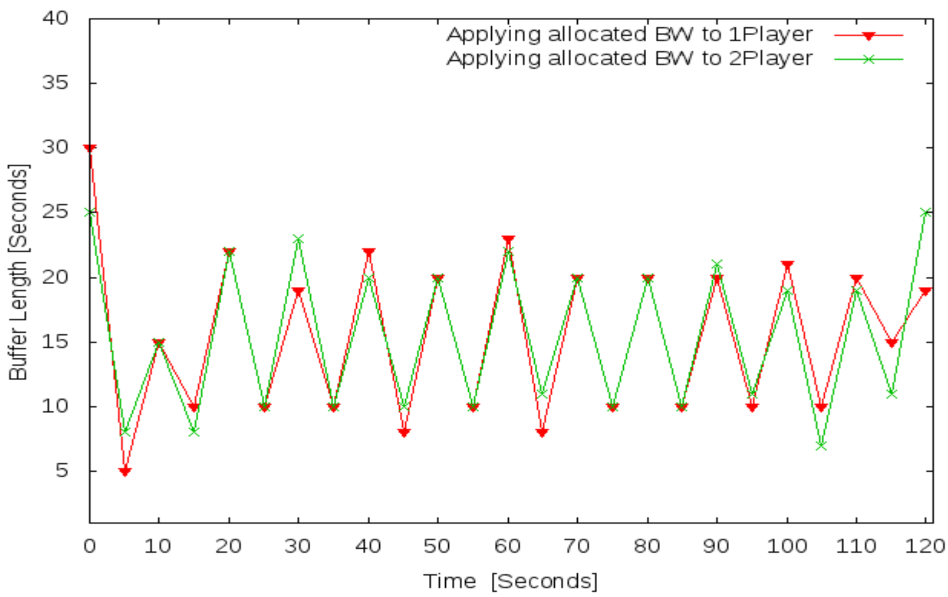


Figure 6.9. Stability of the buffer length.

The message flow between each element of the topology is presented in Figure 6.10. At first, the first client connects to the wireless AP and requests to connect to the server and read the MPD of the adaptive video. After the server provides the information, the client starts requesting video segments. Then, the second client connects to the wireless AP, connects to the server, requests the MPD file and starts requesting segments. After that, the SDN detects the throughput oscillation between the clients and sends a message in order to allocate the fair bandwidth among the clients. Finally, all clients start requesting the segments that allow being fairness among the clients. That implies that the requested segment has a lower quality avoiding the competition between clients. When the streaming is finalized all connections are closed.

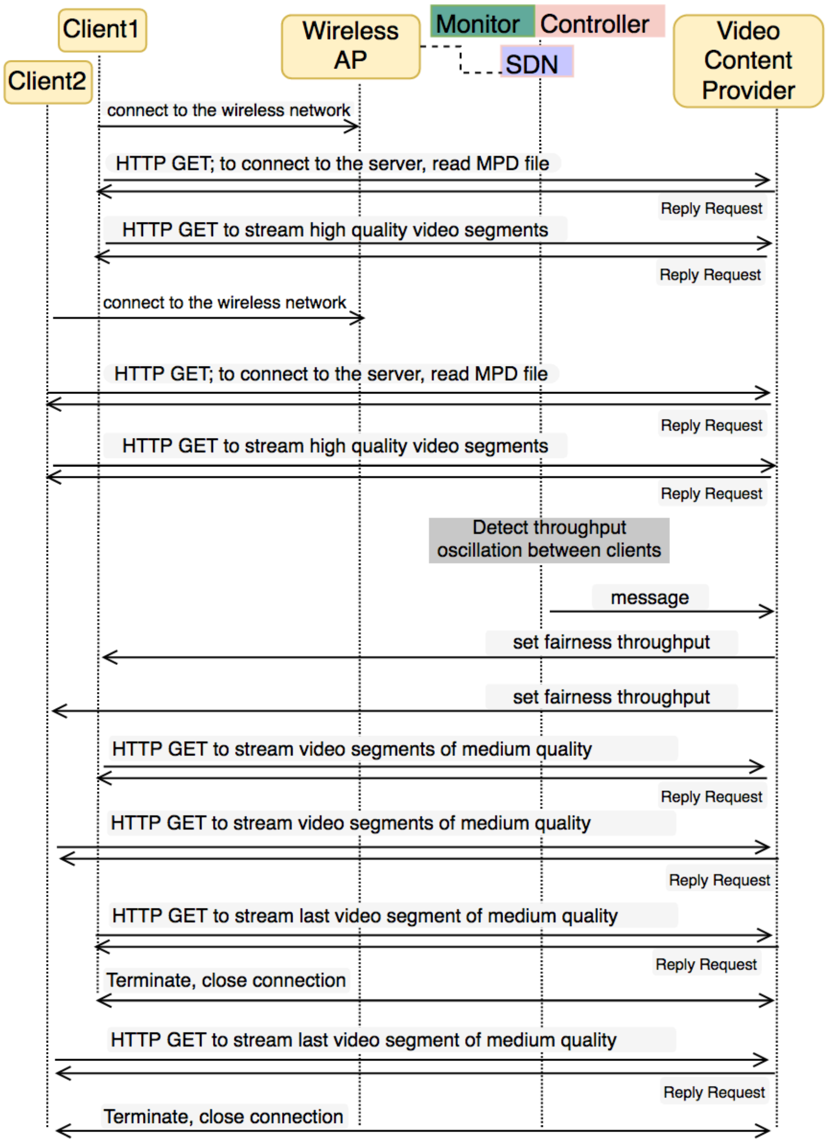


Figure 6.10. Diagram of the message flow.

6.1.5 Result analysis

In order to better compare the results of allocating the throughput of each client from the data collected of the streaming without throughput allocation, we have performed a fairness measure. The fairness index is calculated by employing the highly known Jain's algorithm. With this index, we obtain the difference between the requested bitrate and the allocated throughput. Figure 6.11 shows the results obtained after applying this index to both the measures taken without throughput allocation and with throughput allocation. It can be seen that the fairness decreases as the number of clients sharing the wireless network increases. For the video streamed without bandwidth allocation, the decrease of the fairness index among concurrent users is more abrupt than the case with bandwidth allocation. Utilizing the proposed algorithm, the obtained fairness is very close to the optimal fairness state and the decrease that occurs when the number of clients increases is not significant

Jain's algorithm [149] has been applied to determine the obtained fairness among clients as depicted in Table 6.1. The aforementioned results show an improvement of the fairness of 39% for two competing clients, it has been compared to the traditional method, which provides unfairness. That is shown in Fig. 10. The improvement for three competing clients is 43% and the improvement obtained when four clients are competing for the available bandwidth of the wireless network is 46%. These results show that the perceived quality of the videos is incremented as the number of competing clients increases. The obtained fairness utilizing our algorithm is 99% for two competing clients, 98% when three players share the available bandwidth and 97% when four clients compete between each other.

Table 6.1. Comparison between the tradition and proposal approaches.

Number of the clients	Fairness without applied algorithm %	Improved Fairness %
2	39	99
3	43	98
4	46	97

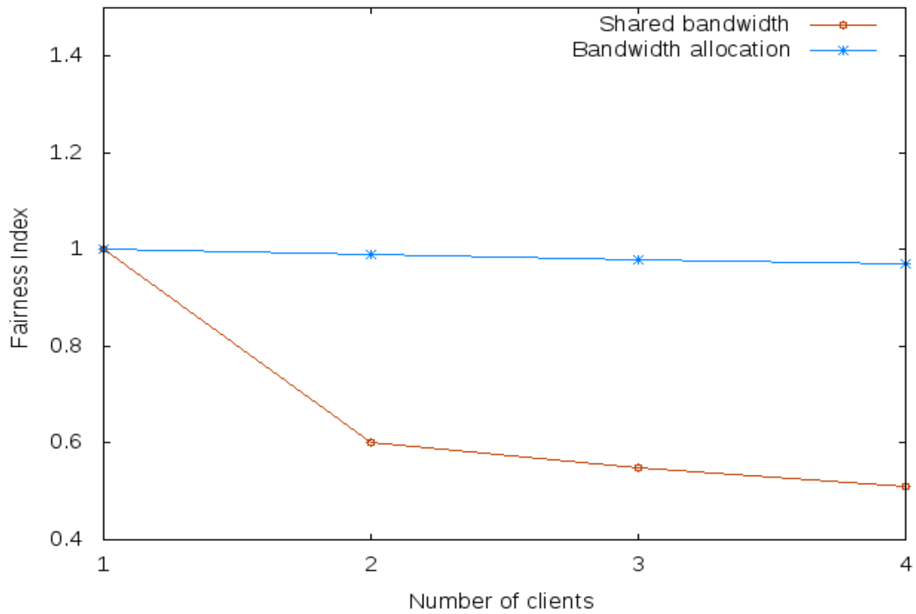


Figure 6.11. Evaluation based on Jain's approach

6.2 Optimization QoE in cellular network

6.2.1 Introduction

Climate change, natural disasters, abuses and the lack of concern of humans for the state of the environment and the preservation of natural spaces has led to numerous environmental problems that require constant monitoring and studying. Monitoring the environment allows to collect diverse data on the environmental condition of an area, as well as the flora and fauna [150]. This allows acquiring knowledge on the evolution of an area, the behavior of the animals and the development of the strategies of rehabilitation of damaged zones. Furthermore, it permits to evaluate the protection measures that are being performed at the areas of study. Employing cameras for monitoring the environment [151] allows performing a more detailed study of the environmental situation of an area providing visual information to the people entrusted with the conservation of the area. With these cameras, it is possible to have access to data that would be impossible to obtain with other resources. Utilizing these means, the process of obtaining content for environmental awareness programs is simplified. Sensitizing children on the importance of maintaining the environment is crucial to promote sustainability of the natural areas among us. 5G networks incorporate improvements regarding previous generation networks and allow the collection of the information from the diverse devices that gather video content. However, these devices may be located on different types of vehicles such as cars, busses or oats, as well as on people. This is a challenge for performing the streaming of the data because handover process between adjacent cells needs to be performed without any information loss. A high loss rate in real time video can undermine greatly the perceived quality of the content.

Throughout the years, several techniques of handover have been developed. These techniques can be classified in vertical handover, where the handover is performed between different technologies, and horizontal handover, where the handover occurs between different cells. The main handover techniques are Hard Handover and Soft Handover. Utilizing Hard Handover, the connection between the device and the antenna is severed before establishing the connection with a new cell. When performing Soft Handover, the connection with the new cell is performed before severing the previous one. Among Soft Handover techniques, the two principal solutions are

Macro Diversity Handover (MDHO) and Fast Based Station Switching (FBSS). A faulty handover process can introduce packet loss, interruptions and interferences.

In this section, we present an algorithm that allows improving the handover process and the load balance between adjacent cells. Our proposal is specially aimed to improve the handover for environmental monitoring devices that obtain stream video information and deliver it to a server. This information can be accessed real time as part of an environmental awareness program for children or in order to obtain data that allows researchers to study the evolution of the monitored area.

6.2.2 Proposed Architecture

In this section, the description of our streaming system and the presentation of the implemented area will be shown.

6.2.2.1 Description of Albufera Protected Area

Natural Park of l'Albufera is a protected area, emplaced in the east of Spain in the Mediterranean coast. It has an extension of 210 km^2, including different ecosystems as dunes, forests, marshes, lakes and agricultural lands. The fauna in the park is mainly composed by birds as herons and ducks that live in the lake and in agricultural land. Some of the fish that live in the pack are autochthonous species that are endangered as Valencia hispanica. The area is included as a Ramsar Site and in the Spetial Protection Areas list. In this park several measures have been taken to restore the ecosystems. High percentage of the protected area is composed by rice paddies. In addition, there are some villages. There are two freshwater lakes, one with 6km of diameter, which is connected to the sea by three channels. The other has an artificial origin and a diameter of 0.5km. Moreover, there are several marshes with smaller area. Several touristic activities are developed mainly in the biggest lake. There, some traditional boats offer different boat trips. In the terrestrial part of the protected area there are different roads and path.

6.2.2.2 Employed video cameras

For the video surveillance of the natural park, different cameras' position is used. The cameras available in variety speed because they can be attached to people, cars or boats. The cameras worn by the workers of the natural park, those placed on the helmets side, the velocity of them is 4km/h. On the other hand, the cameras attached to cars are placed on the windshield.. The average velocity of the cars is 60km/h., although, the installed cameras on the boats have different positions, the velocity of the boat during the trip is 7.5km/h.

In order to implement streaming video content, The GoPro version HERO5 will be employed. They are submergible camera, which are able to record video in 4K. The battery-life of the camera device can reach to an hour and 15 minutes (see Figure 6.12).

6.2.2.3 Architecture

Mobile GoPro cameras will be used to capture video in the natural park. The Go Pro device is incapable to transmit via cellular network. However, they can transmit via Wi-Fi to the smartphone. There are different applications for the smartphone that allow the streaming from the GoPro using the cellular network. Moreover, in the rural, the macro cells of the 5G can be coverage the rural areas. The macro cells coverage up to 10km of diameter. For our project, the antennas with coverage of 10km are used. Figure 6.12 shows distribution of the antennas in our protected area. The antennas are placed to ensure the coverage of the 100% of the protected area (in green). Moreover, the antennas have an overlap of at least 15% between their neighbors to guaranty the handover. Therefore, a total of eight antennas have been planned in the park. Figure 6.13 depicts the proposed architecture for our streaming system. The videos are transmitted from smartphones using the 5G networks to the server, which placed in the research center of the natural park. The researchers can have a global view of the park in real-time. Although, analyzing videos will be a great tool for monitoring the effects of the proposed measures.

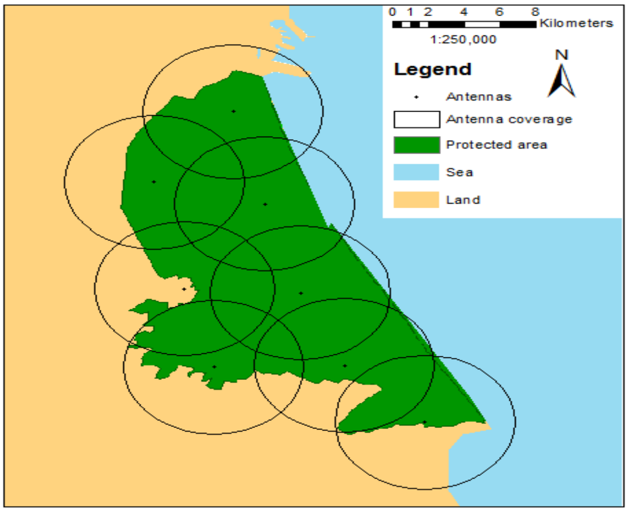


Figure 6.12. Architecutre of the implemention

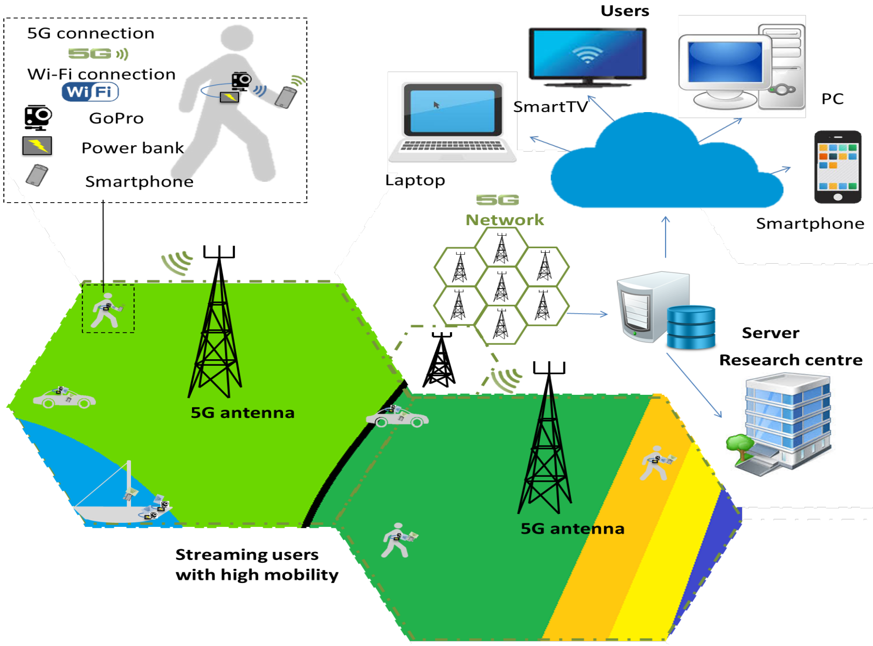


Figure 6.13. Proposed System architecture for handover process

6.2.3 Proposed handover algorithm

The detail of the proposed algorithm will be shown in this section, the algorithm based on a decision, which leads fulfilling the handover process for smart devices that deliver video content in the real time. One of the main issues of delivering video content in real time throughout mobile networks is the service quality of networks such as bandwidth, delay, packet loss and jitter. Moreover, The problem gets worse when there is weakness parameters make the decision in selecting candidate base stations. The inadequate decisions degraded the availability transmission rate of mobile users during the video transmission and this is making the QoE of monitoring video streaming become degraded. Although, this is caused high number of packets are retransmitted in surveillance video applications that is based on using TCP. Generally, in the cellular networks, radio signal strength between both the mobile users (MUs) the base station (eNodeB) regularly has been measured. The mobile device frequently sends update of its measurement to the base station, if the base station detects the minimal level of threshold; the process of handover has been accomplished to new base station.

To obtain the decision of handover process, we employ the following parameters;

Base station capacity.

In 5G, Macro cell nodes avail the networking resources and act as the central coordinate system in the heterogeneous networks, though, the capacity of macro cells can be presented by bandwidth. Number of mobile users and traffic load deteriorate the capacity of the channel. In order to measure the capacity of base stations, we formulate the mathematical equation as shown in Equation 6.4.

$$\boldsymbol{Capacity}_{\boldsymbol{Bs}} = \boldsymbol{Av.connected}_{\boldsymbol{MUs}} * \boldsymbol{th}_{\boldsymbol{MUs}} \;/\boldsymbol{Av.Traffic}_{\boldsymbol{Bs}} \qquad 6.4$$

Where, the throughput has been denoted by *th.*

Movement of mobile nodes

Mobile users during recording video content have variety of velocity. The range of velocity may be changed from one user to others, which is calculated by Meter per seconds or Kilometer per hour. Moreover, the velocity impacts on the distance between mobile nodes and base stations.

Signal strength

Signal power as denoted by φ in (dB) received by a UE from the antenna, the model can be shown in the equation 6.5.

$$\boldsymbol{\varphi}(\mathbf{d}) = \mathbf{k1} - \mathbf{k2}\ \mathbf{log}\,(\mathbf{d}) \qquad 6.5$$

Where (d) is the distance of the UE to the antennas, K1 represents the gain of the transmission and reception antennas, whereas, K2 represents attenuation characteristic of specific environment.

Mobile users traffic

Mobile users generate high video traffic onto base stations. The surveillance devices may be taken video content in the 4K-UHD resolutions, which (3840 × 2160) pixels. Moreover, the strategy of handover can be a network-control, since the UE-control handover cannot control load balance among antennas when the base station is poor. In our algorithm as shown in Figure 6.14, we calculate proactive threshold values to guarantee delivering video content in the handover process and provide better QoE. Where, we take the maximum and minimum ranges. The threshold of the signal strength is shown in Equation 6.6.

$$\boldsymbol{\theta}_{\mathbf{Singal\ strength}} = \mathbf{Min}\ \boldsymbol{\varphi}((\mathbf{d})) \qquad 6.6$$

Where,

$$\begin{cases} \text{if } \theta_{\text{Singal strength}} < \varphi(\mathrm{d}), \text{ signal is high} \\ \text{if } \theta_{\text{Singal strength}} > \varphi(\mathrm{d}), \text{ signal is weak} \end{cases}$$

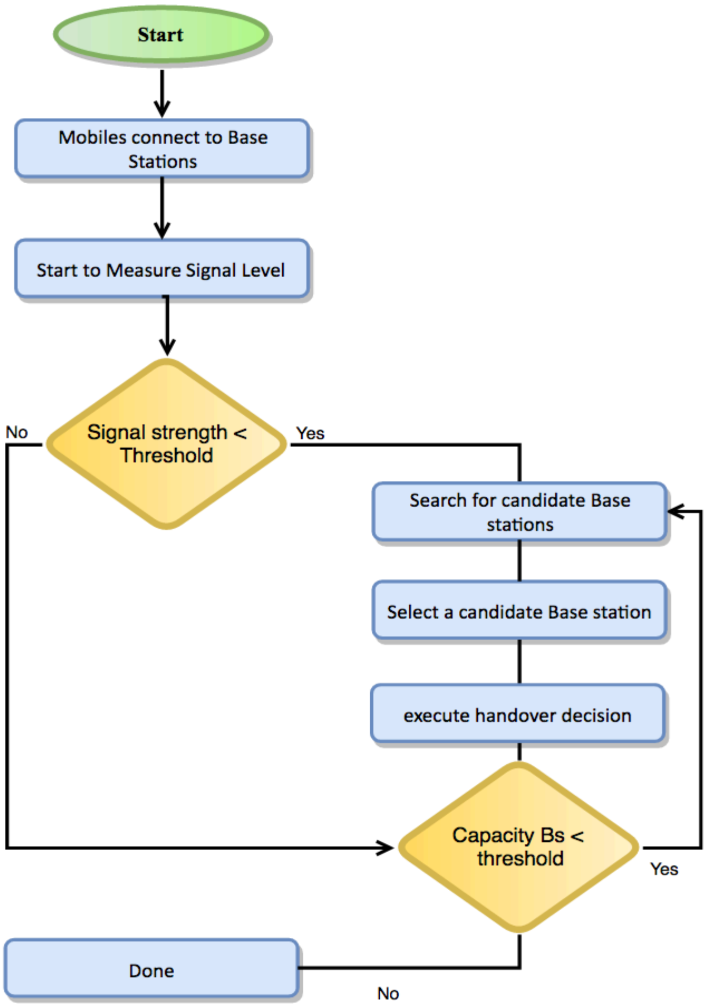


Figure 6. 14. Proposed the handover algorithm.

6.2.4 Performance evaluation

In order to present the performance of our proposed algorithm, we evaluate some measures to show the comparison between our propose algorithm and conventional approach in 5G handover. In the scenario, mobile users capture video streaming for length 300 seconds. A user can generate 1.7 GB of video data for a 5 Minutes. Moreover, a user can transmit real time video data with rate of 350Mbps. Each antenna equipped with high range of mobile users. To obtain the results, First, we measure maximum traffic load of mobile users be transmitted during handover process before and after applying our algorithm for high velocity, as shown in Figure 6.15. In seconds 50, 100, 200, the UEs switched to different base stations, the green line depicts that the number of switches between the base stations keeps better transmission for the user. However, the case is worse in the green line, the user suffers from its transmission rate to stream the video. Moreover, the handover delay will be shown in Figure 6.16. Results of the figure shows that delay of packets from the second handover raised up, the rate of lift red line is three times more than the green line. Finally, The ratio of packet loss has been depicted in Figure 6.17. The packet loss is increased when the number of users is increased, therefore, high congestion on base stations produces high packet loss in both case.

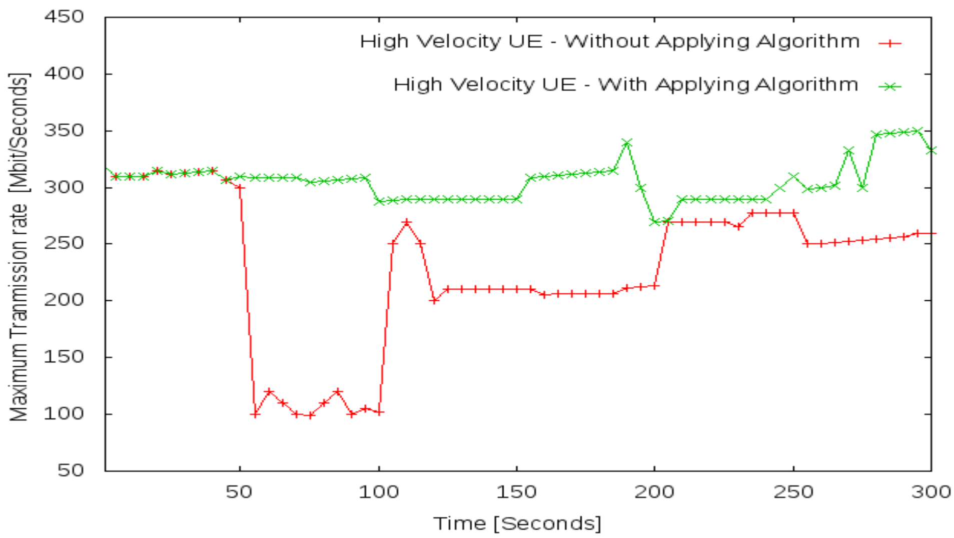


Figure 6.15. Transmission rate of the video.

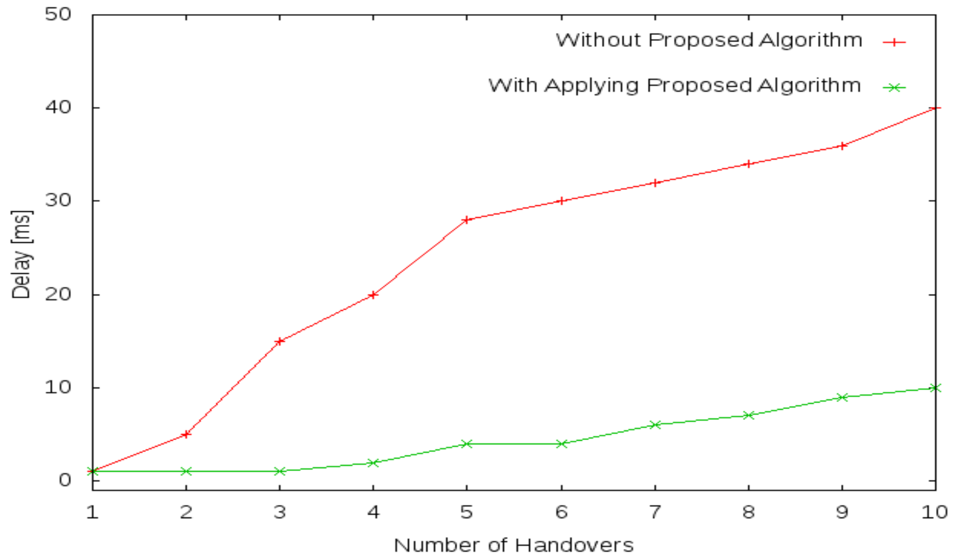


Figure 6.16. Delay of the handover process

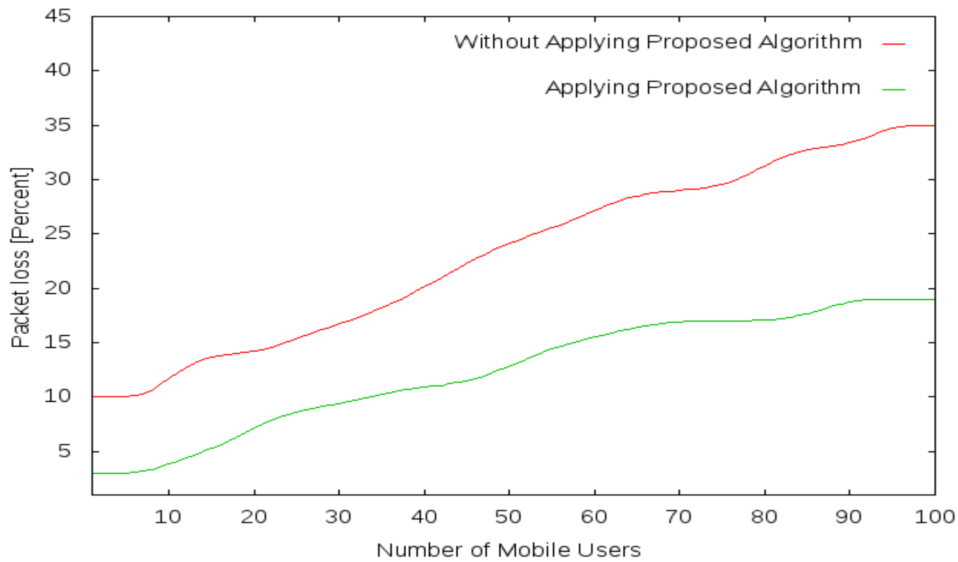


Figure 6.17. Number of packet loss during the handover.

6.3 Chapter conclusion

In this chapter optimization of QoE has been proposed for HTTP adaptive video streaming and surveillance video streaming. In Internet video traffic is increasing over time and as the number of adaptive video streaming users increments, the effects of player competition on wireless networks are more noticeable. We evaluated those effects performing several measures with two different videos. The throughput allocation algorithm based employing SDN to allocate the available bandwidth among the clients fairly is proposed. The results show that our proposal improves the fairness between clients 39% for two competitors, 43% for three and 46% for four competitors.

Therefore, we have shown the importance of video streaming when using mobile cameras in environmental surveillance. An architecture and a handover algorithm have been proposed to provide better QoE and efficiency of network resources by applying the algorithm in the network provider. Moreover, the evaluation for the proposed algorithm is depicted that the proposal algorithm improves throughput and reduce the delay of mobility users and provide better QoE to surveillance users, when transmit ultra definition video quality such as 4K is streamed from handset device to central data base in order to distribute monitored video to other users.

Chapter 7. Virtualized Testbed Design for Evaluating QoE

7.1 Introduction

The increased trend of multimedia technology and smart devices to playback multimedia streaming over heterogeneous networks has become very popular. Although, providers are interested to monitor and evaluate QoE of end-users as explained in section (3), (4) and (5). QoE evaluation needs a test environment to assess quality of the service. However, providing test environment scenarios over using private or public network operators in order to evaluate QoE are high cost and time consumption therefore limitation of using these network components doesn't allow researchers to reconfigure the resources and reusing for other purpose. And it's difficult to develop and verify new approach using real networks operator. Designing testbeds to provide environment tests by using the virtualization technique has become a key component of the network testbeds. They allow testing protocols and applications with varying network conditions. Virtualization testbed is a broad concept in the information and telecommunication area, dealing with the sharing of physical resources. There are already several different virtualization concepts adopted in practice, which target operating systems, hardware, CPUs and embedded systems, networks or storage. The general advantages of sharing resources between different applications are the following ones: reduced number of equipment devices, commoditization of resources, reduced complexity in the management of resources, reduced time needed for deployments using the virtualized infrastructure and flexibility in usage. The economic aspect lies in the optimization of needed resources, and therefore in the reduced Total Cost of Owner Ship. Besides, there are several aspects that have to be taken into account when considering network virtualization, e.g.: better planning of the needed shared resources, additional management for resource sharing, integration of specialized resources requiring higher efforts, and operation and maintenance requiring additional debugging mechanisms.

Generally, network testbeds have been used to define specific experiments. Once a testbed is defined with its resources for a specific experiment, it can be used to test the performance and gather measurements to analyze the results. But what benefits the research and industry community most is to have testbeds that allow the performance of several types of experiments. The advantage of the simulation and virtualization network environments is that the complex networks can be set up quickly in order to meet the need of complex experimental environment for research. Therefore, a wide range of testbeds such as: OneLab, Emulab, G-Lab, NetKit, and PlanetLab have been designed for different goals as discussed in section (2.8). Nevertheless, it is quite difficult for these testbeds to reserve enough resources for their own experiments and they have to meet different requirements than those of testing applications.

In this chapter, we build a virtualized network testbed, which provides different network function to evaluate QoE of multimedia applications. The virtualized testbed provides the complex Internet scenario such as content delivery network (CDN) and wireless network scenarios. The proposal-virtualized testbed presents own mechanisms and functions for distribution multimedia service and redirection the client's request. Therefore, the system allows the multimedia service provider measuring QoE of their end-users by using efficiently QoE monitoring mechanism and from the obtaining results the process of evaluation estimate the future QoE for the different network scenarios according to the subjective and objective evaluation methods as explained in section (3), (4) and (5).

7.2 Virtualized testbed architecture

This section describes the involved components in the architecture of virtualized testbed. This architecture is developed to evaluate quality of experience (QoE) of multimedia streaming in Internet scenarios and wireless environments.

7.2.1 Description of CDN components

The essential components of the virtualized testbed include the scenario of CDN, which are three main modules: main server, cache servers, and clients. Therefore, the system has mechanisms and protocols to distribute media service and content, and redirection end-users to appropriate point. Figure 7.1 depicts the overview of CDN architecture with the mechanisms. The components are further presented in detail as follows,

7.2.1.1 Origin server

Origin server is the main source of content provider. Multimedia providers manage large database of videos at the origin server. The database of the video may contain live and on-demand videos streaming.

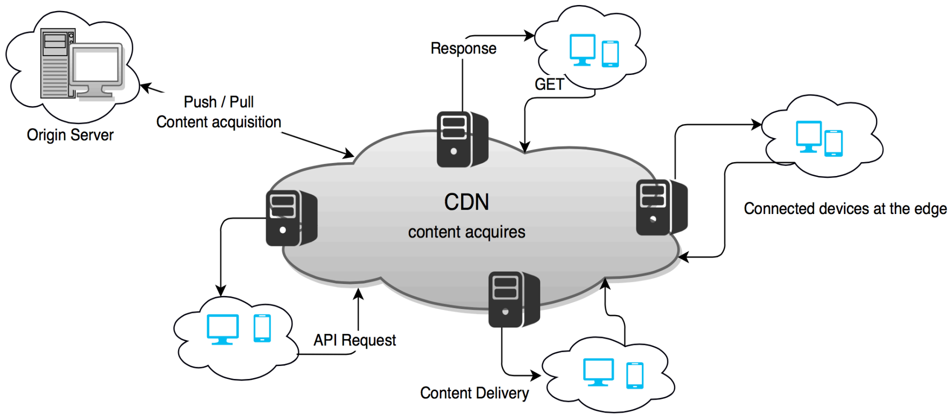


Figure 7.1. Architecture of CDN.

Servers are concerned with updating and publishing videos. Multimedia providers such as Netflix and YouTube are usually relied upon CDN operators for distributing their video streaming to users across various geographic locations.

7.2.1.2 Surrogate Server

The main motive of a CDN operator is to deliver videos to end users on behalf of the origin server while guaranteeing good quality. In order to achieve this, CDN operator strategically hosts numerous cache servers across various locations or point of presents (POPs) and these servers are called surrogate servers. The CDN is characterized by number of surrogate servers and their locations. Based on the type of CDN operator, the technique of deploying and managing surrogate servers varies. The surrogate servers are usually proxy servers that distribute videos on behalf of origin server of multimedia provider. Proxies are broadly categorized into forward or reverse proxies. Both of them perform fundamentally the same caching task but differ in the way they are implemented. Forward proxies are implemented to intercept all web traffic and they can be found in ISP network. Reverse proxies are implemented as CDN's surrogate servers. They are helpful to address content requests belonging to origin servers. Cached content in reverse proxies is under the explicit control of the content provider. The important goals of surrogate servers are to decrease the network traffic, reduce the client perceived latency, reduce load on the origin server, increasing the availability of content and saving bandwidth. In order to achieve these goals, the surrogate servers usually employ caching techniques for storing data. Caching proxies are usually placed close to the users to store most frequently accessed videos. Users request specific content via HTTP requests.

The internetworking and collaboration among the surrogate servers of a CDN can take place in three forms. They are cooperative push-based, non-cooperative pull-based and cooperative pull-based. The Cooperative push based is based on prefetching of content to the surrogate servers. In this approach video is directly pushed or uploaded to the surrogate servers by the multimedia provider and the surrogate servers operate in a cooperative way. The Non-cooperative pull based is based on a pull-based mechanism where a user's request is redirected to the closest surrogate server. In case of cache MISS, the surrogate servers pull or fetch content from the origin server. This approach turns the surrogate server into a standalone server to address requests from users. The Cooperative pull based is similar to non-cooperative pull based approach in the way a request from user being directed to the closest surrogate server. The main difference is in the way the video is pulled or fetched in case when the video is missed in cache. In the cooperative pull, the surrogate servers are cooperated with each other before fetching the content from origin server.

7.2.1.3 Request redirection mechanism

CDN operators use request redirection mechanism to dynamically redirect requests from users to the most suitable surrogate servers. The mechanism is based on different parameters like surrogate server load, network congestion, latency, user access network and proximity to users. Three main methods adopted by CDN operators for implementing redirection mechanism as follows;

1) DNS based redirection: In the DNS approach, servers handle the domain names of multimedia provider website and the addresses of various surrogate servers. Whenever user requests content, the domain name is looked up in the local DNS server and the address of suitable surrogate server is returned. If a cache miss is noticed at the local DNS server, the request is forwarded to the DNS root server, which returns the address of the authoritative DNS server of multimedia provider. The DNS server of multimedia provider then returns the address of suitable surrogate server based on load monitoring and specialized routing. The client finally retrieves requested content from the designated surrogate server.

2) HTTP based redirection: The redirection approach uses feature of HTTP protocol. The multimedia web servers are operated by CDN operators inspecting requests from clients and redirect it to the most appropriate surrogate servers. This method provides flexibility in serving content to users with fine granularity. Users can be served location specific content by redirecting their requests to suitable surrogate servers.

3) URL Rewriting: This method employs a plugged application running on a web server, which is responsible for modifying web URLs. Based on the type of content requested by users, this software rewrites the URLs and points to specific surrogate servers that serve the content better. Using this method, URLs can be rewritten to serve text, images and videos from appropriate surrogate servers.

7.2.2 Network emulation

The network traffic shaping emulator in the virtualized testbed is control throughput of available bandwidth by prioritizing network resources and guarantee certain bandwidth based on predefined policy rules. Traffic Control (TC) is a user-space utility application to the sets of queuing systems and mechanisms by which packets are received and transmitted on a router. This includes deciding which packets to accept at what rate on the input of an interface and determining which packets to transmit in what order at what rate on the output of an interface. Therefore, it entails the shaping, scheduling, policing and dropping of packets and is applied by the kernel. Shaping and scheduling are actions applied to outgoing traffic to delay packets to meet a desired rate or use reordering to induce priority. Policing is used to limit the queuing of arriving data traffic. In the case that packets are not accepted into a queue in either direction, they are dropped by the scheduler. All network interfaces require some kind of scheduling policy, or qdisc, but the structure does not necessarily have to be complex. It may be as simple as a First In First Out (FIFO) queue. By default, Ubuntu uses a priority-based FIFO queue, pfifo_fast, as its qdisc. This queue is a slightly more complex version of a standard FIFO queue, using multiple standard queues to represent different priorities where the first queue is processed before the second and so on. The standard implementation of pfifo_fast uses three FIFO queues, and packets are placed into a queue based on the Type of Service (ToS) octet in the IP header, which may be used to express packet priority. The control path is described in the next.

- **Queuing Disciplines:** There are two main types of qdiscs in Linux. Classful and classless. A classless qdisc can, as the name suggests, not containing classes. Nor it is possible to attach filtering to shaping the network. Classful qdiscs on the other hand may contain classes and subclasses on which can apply more classes or filters for advanced queue management. Incoming and outgoing traffic accepted by or transmitted on a network interface traverses the ingress or egress qdiscs respectively. These are not qdiscs in the general sense but rather locations on which we can apply structures. The egress qdisc also known as the root qdisc may contain any queueing discipline, and allows for much more advanced behavior than its ingress counterpart. The reason for this is that we generally only want to apply scheduling to outgoing traffic and seldom want to apply unnecessary delays on arriving data. Thus the ingress qdisc only exists as an empty root object on which filters may be attached.
- **Hierarchical Token Bucket:** HTB uses the concepts of tokens and buckets along with the class-based system and filters to allow for complex and granular control over traffic. With a complex borrowing model, HTB can perform a variety of sophisticated traffic control techniques. One of the easiest ways to use HTB immediately is that of shaping.
- **Handling a link with a known bandwidth:** HTB is an ideal qdisc to use on a link with a known bandwidth, because the innermost (root-most) class can be set to the maximum bandwidth available on a given link. Flows can be further subdivided into children classes, allowing either guaranteed bandwidth to particular classes of traffic or allowing preference to specific kinds of traffic.

- **Handling a link with a variable (or unknown) bandwidth:** In theory, the PRIO scheduler is an ideal match for links with variable bandwidth, because it is a work-conserving qdisc (which means that it provides no shaping). In the case of a link with an unknown or fluctuating bandwidth, the PRIO scheduler simply prefers to dequeue any available packet in the highest priority band first, then falling to the lower priority queues.
- **Network Emulator (Netem):** Netem is a classless qdisc Linux traffic control allowing the emulation of wide area network properties such as delay, loss, duplication, reordering and corruption and more other characteristics to packets outgoing from a selected network interface. NetEm is built using the existing Quality Of Service (QOS) and Differentiated Services (diffserv) facilities in the Linux kernel.
- **Fair Queue (FQ):** The FQ packet scheduler is a classless qdisc designed to achieve per flow pacing. This pacing can be used to negate the burstiness of some of the TCP congestion control algorithms, which may induce higher queuing delay, more packet loss and a lower throughput. The pacing spreads the transmission of outgoing data evenly over an RTT rather than pushing everything out in a single burst.

7.2.3 Router equipment

Router in the virtualized testbed is a networking device that forwards data packets between equipment of networks. Routers perform the traffic directing functions from main server to surrogate server or among surrogate servers to clients. A data packet is typically forwarded from one router to another router through the networks that constitute an internetwork until it reaches its destination node. The testbed routers use shortest path to deliver media content as fast as possible to among all equipment. E.g. client when request to main server to obtain video stream data, the router connected to the client, forward the packet to shortest router.

7.2.4 Clients

Clients may a physical device or virtualized device, which connected to the system testbed to receive service from the main server or a server of the CDN, which equipment with heterogeneous application to receive media service. Clients may be computers, tablets and mobiles located inside or outside the virtualized testbed. They operate different operating system including Linux, Windows, IOS, etc. They can access the CDN servers to get service through using Chrome browser to playback multimedia streaming or VLC media player. The characteristic of Chrome is to support media source extension (MSE) to handing the media segments together. Therefore users can upload and modify video content in the system as the permission given by servers.

7.2.5 QoE Metrics

Indeed, the virtualized testbed is developed to provide environment of experiment to evaluate multimedia services. According the studies has been done in section (4) and (5). The proposed system is focused on assessment QoE for adaptive video streaming, moreover, evaluation of adaptive video streaming is much more complex than video streaming because vary metrics affected on QoE of end-users as explained in (section 2.5). In adaptive video streaming, subjective and objective are important approaches can be used to assess the delivered video. Indeed, there are two primary aspects to assess video quality, display quality (fidelity), image quality is sufficient for the device's screen size. And transport quality (startup time, buffering, switching and stalling), how long does the video take to start, and does it play smoothly. The assessment model is grouped into parts an objective assessment and a subjective assessment. The following QoE metrics are used to provide the assessment model in our system. It is to evaluate QoE regards to satisfactory and annoyance of the service.

7.2.6 Tools and software

There is variety of tools and software is conducted to be attained in the testbed and to provide experiments approach, providing providers, measure network conditions, preparing streaming content and providing an environment for various processes. In the following list briefly describes the tools have been used in the virtualized testbed setup regarding to its functionality and fulfill process.

- **Operating system:** Ubuntu is a multi-platform open source operating system running on the Linux kernel. All virtual machines of the testbed are set up with a server version and mirror server of Ubuntu. Running the same operating system and kernel version throughout the entire testbed allows maintaining homogeneity on an operating system level. This is an important element in ensuring that the testbed is as deterministic as possible as well as being more easily managed and more maintainable.

- **Tcpdump:** Tcpdump is a tool used for capturing network traffic from a network interfaces. It provides the ability to apply filters to store only the traffic satisfying set requirements such as a specific destination host or a port range. Tcpdump is also capable of saving the network traffic to file resulting in a packet capture (pcap) file, which operates as a command-line program, where the user inputs filtering rules, packet capture is executed, and then logs of results can be output to a separate file.

- **NGINX:** NGINX is a free, open-source, high-performance HTTP server and reverse proxy, as well as an IMAP/POP3 proxy server. NGINX is known for its high performance, stability, rich feature set, simple configuration, and low resource consumption.

- **FFMPEG and X264:** FFMPEG is a multimedia framework with a great number of video management capabilities such as encoding decoding and transcoding among different codec types. It is used to encode the source video material in different resolutions and bit-rates before splitting it into segments.

- **GPAC & MP4Box:** GPAC is an open source multimedia framework with a wide area of application ranging from research to academics to industrial collaborations. GPAC comes with a media packaging application called MP4Box, which provides video management functionality such as conversion and video splitting compatible with the MPEG-DASH standard. The MP4Box is used to split the source material into segmentation and maintain in standard format.

- **Client Application player:** VLC and DASH.js are open source application, which is led to playback adaptive video streaming over HTTP application layer. DASH is an open source DASH framework created by the DASH Industry Forum. They sought to create a production quality framework for building video and audio players that play back MPEG-DASH content using client side JavaScript libraries.

- **Squid cache server:** Squid is a caching proxy for the Web supporting HTTP, HTTPS, FTP, and etc. it is used by hundreds of Internet providers to provide their users with the best possible web access. Squid optimizes the data flow between client and server to improve performance and caches frequently used content to save bandwidth. Squid can also route content requests to servers in a wide variety of ways to build cache server hierarchies, which optimize network throughput.

- **Extract metrics:** The evaluation metrics are based on the segment requests extracted from the NGINX HTTP access logs, JavaScript clients application, capture Network information Although parsing the logs is easily done using simple BASH commands, generating the evaluation metrics, therefore, several hundred log files were easier to implement in python and shell bash script. This application takes the log files as input and output each of the metric results to respective files. Each client is streaming using an HTML document with their hostname ($(hostname).Index) so that they are easily differentiated in the NGINX logs. Each segment request is parsed, containing client id, time of request, segment id, segment layer and bytes downloaded. All this information is stored as a Segment class within its respective host class. This way we have easy access to every segment from every host through a simple host array, along with relevant information. QoE information of the clients extracts from the real monitoring clients during playback of the videos.

- **Virtualized tool :** In order to provide the testbed, Virtual Network over Linux called (VNX) [157] is used to help building virtual network testbeds automatically as shown in Figure 7.2. It allows deployment of network scenarios made of virtual machines for homogenous and heterogeneous types of operating systems. The VNX tool allows the definition of virtual network scenarios and controls their deployment over either a Linux server or a cluster of servers. The user can control how the virtual scenario is distributed over different cluster servers, using algorithms, restricting rules. The main reasons of using VNX in this project are agility, fast, and easy to implement requirements over the Linux operating system.

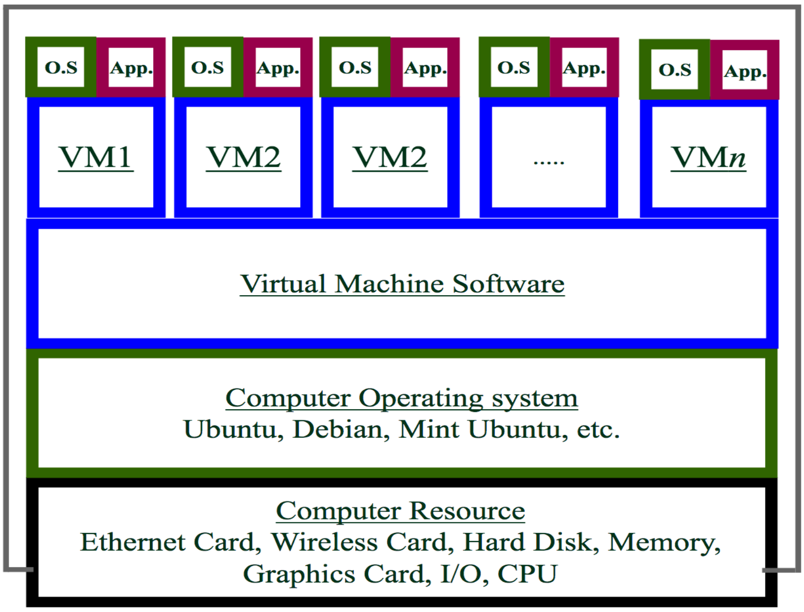


Figure 7.2. Virtualization structure.

7.3. QoE and resource usage metric calculation

In adaptive video streaming, subjective and objective are important approaches that can be used to assess the delivered video. Indeed, there are two primary aspects to assess the video quality, display quality (fidelity), image quality is sufficient for the device's screen size, transport quality (startup time, switching, buffering and stalling), how long does the video take to start, and does it play smoothly. The assessment model is grouped into two parts, an objective assessment and a subjective assessment. The following QoE metrics are used to provide the assessment model. In our system, we evaluate QoE from the point of view of the satisfaction and annoyance of the service, the detail of each metric will be defined and explained clearly next

7.3.1 Initial delay and buffer length

The client application creates a buffer length of Y duration in seconds, the value of Y is based on the video segment length and the maximum availability of buffer size related to QoS factors such as high bandwidth variation and packet loss rate. Buf_sl is denoted as buffer size of segment length, where $\mathrm{Buf\_sl_i} < \mathrm{Buf\_sl_{i+1}}$, i = 1 to M where M is the maximum number of segment length. The client application at time T_0 starts to receive data and store them in the buffer application before being played at time T_n, which had an initial delay or startup time of the video as shown in Figure 7.3.

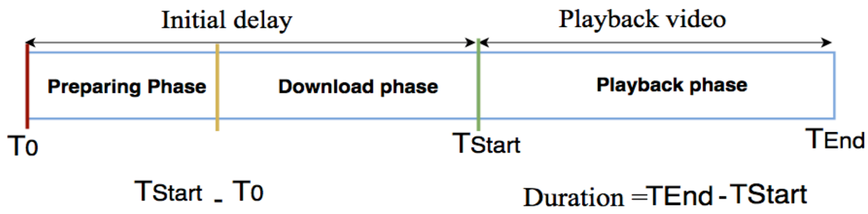


Figure 7.3. Structure of application's buffer.

7.3.2 Oscillation of video quality

The number of switches describes the oscillation of the video session. It depends on the number of bits flowed through the channel per second and the instability of throughput. High value of frequent switching leads to decrease the QoE as described in the Equation 7.1.

$$\sigma^2 = \sum_{i=0}^{n} g(si) \begin{cases} 1 & if \quad i = 1 \\ 1 & if \quad f(s_{i-1}) \neq f(s_i) \\ 0 & else \end{cases} \qquad 7.1$$

$Where,\ s_i\ segment_i \quad i \in [0 - N] \ldots Segment\ Index$

$f_{si} \ldots bitrate\ of\ segment\ _i$

7.3.3 Video accumulative time

Accumulative video time metric is represented as exceed video length versus the default video length. Insufficient bandwidth causes to underrun the buffer of the video and this parameter impacts on the end-user QoE because it is related to the number of stalls and the stall duration. It is shown in Equation 7.2.

$$accumulative\ time = Initial\ delay + \gamma \qquad 7.2$$

Where, $\gamma = \sum_{i=0}^{n} Zi$, Zi is denoted as the duration of each stall, where i is the number of stalls (1,2…N).

7.3.4 DMOS (Difference Mean opinion score)

DMOS is the difference between the MOS value obtained for the original video and the MOS value obtained for the delivered video. So, DMOS value gives the mean subjective value of the difference between the original and the delivered video. A value of 0 means no subjective difference found between the video by all the viewers as mentioned in section (4).

In order to find the threshold values in the proposal model. We take benefit of the subjective assessment. We establish a mapping relation between subjective and objective as it has been studied in sections (4) and (5) in order to automatically assess the QoE. Three important parameters are taken to evaluate the QoE such as initial delay of the video, switch between qualities and video stalling (when client's buffer reach to empty). These parameters are massively annoyed end-users when their values are incremented (according to the research issues in section 2.5). Figure 7.4 depicted the reverse relation between the influence factors and the subjective assessment. The subjective assessment is mapped to the influence factors as described as objective metrics and the indication of objective metrics are (initial delay, stalling and switching), From these metrics, it leaded to know how the objective metrics are annoyed and satisfactory the end-users. Therefore, the threshold value of QoE estimation is based on the subjective evaluation. Also, the range of the subjective evaluations is 1 to 5. The output of the subjective evaluation will be good or excellent evaluation when a user satisfactory is receiving the video with bare initial delay, stalling and quality oscillation and its range can be estimated from 3 to 5, otherwise, the rating of 1 and 2 indicated an annoyance feedback on the perceived quality, when the user faced to high initial delay, massive buffer underrun and high quality oscillation. The median value is set as the threshold to reveal the decision on satisfactory and annoyance of the end-users.

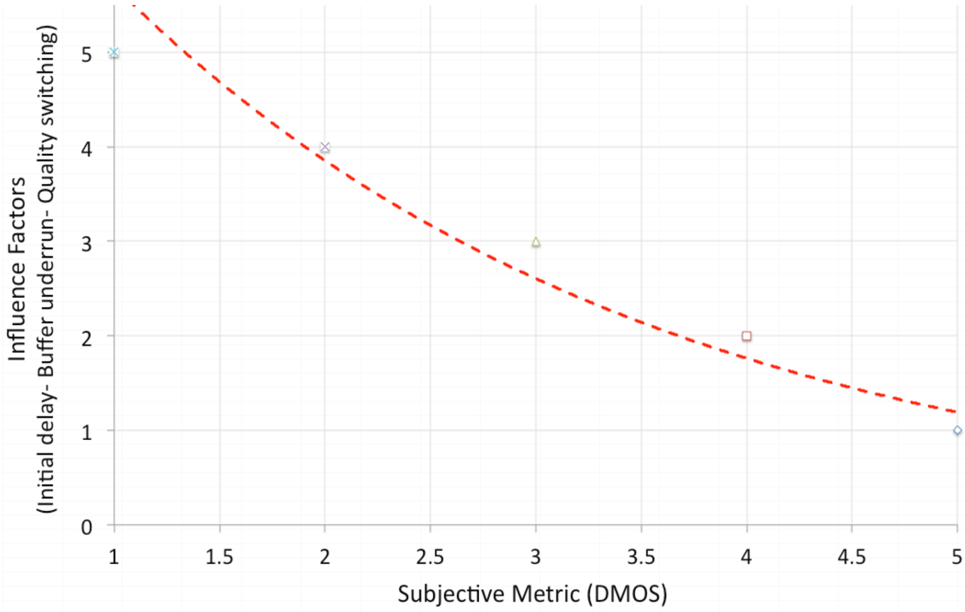


Figure 7.4. Relation between the influence metrics and subjective assessment.

7.3.5 Resource usage metrics

CPU usage and battery power consumption for the VMs and the entire virtualized system are measured by using the Linux monitoring tools. In the VMs [39; 40], the number of cycles executed per second is analyzed when a user watches videos of different segment lengths. The average CPU usage is influenced by some factors such as: high encoded video quality, quality video switching, CPU environment, behavior of the video playback application, and availability of clients' throughput. Therefore, the process of the capturing CPU information is depicted in Figure 7.5 (a). Energy consumption measurement includes real-time monitoring of the virtual devices power usage when playing adaptive videos streaming (see Figure 7.5 (b)). The test when we vary the available throughput, and when there is automatic bitrate adaptation, is used to define how much battery power will be consumed during the oscillation of video quality for different segment sizes. The energy impact will be observed when the fully battery charge has 100 percent. Thus, both aforementioned parameters will be measured for the entire proposed system.

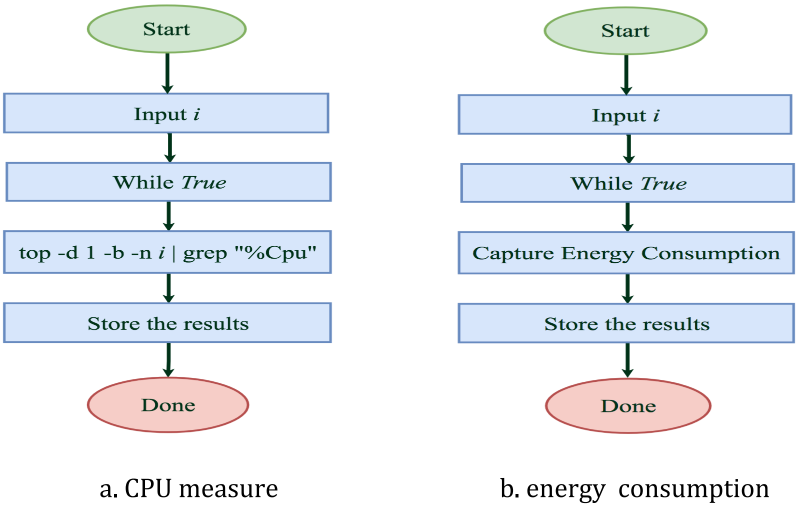


Figure 7.5. Resource usage metrics.

7.4 Configuration of virtualized testbed

In this section, we discuss on the achievement the goal of requiring environment configuration by leveraging host virtualization technology and network configuration at virtual machine layer.

7.4.1 Network topology of the system

The virtualized testbed is configured according to Figure 7.6, which are included the main server and three replica servers with the various routers and shapers. All of them are connected together. The pseudo code of the testbed is presented in Algorithm 1. Which explains the launched scenario for CDN. Therefore, the configuration of each virtual component of the scenario is presented in the next subsections.

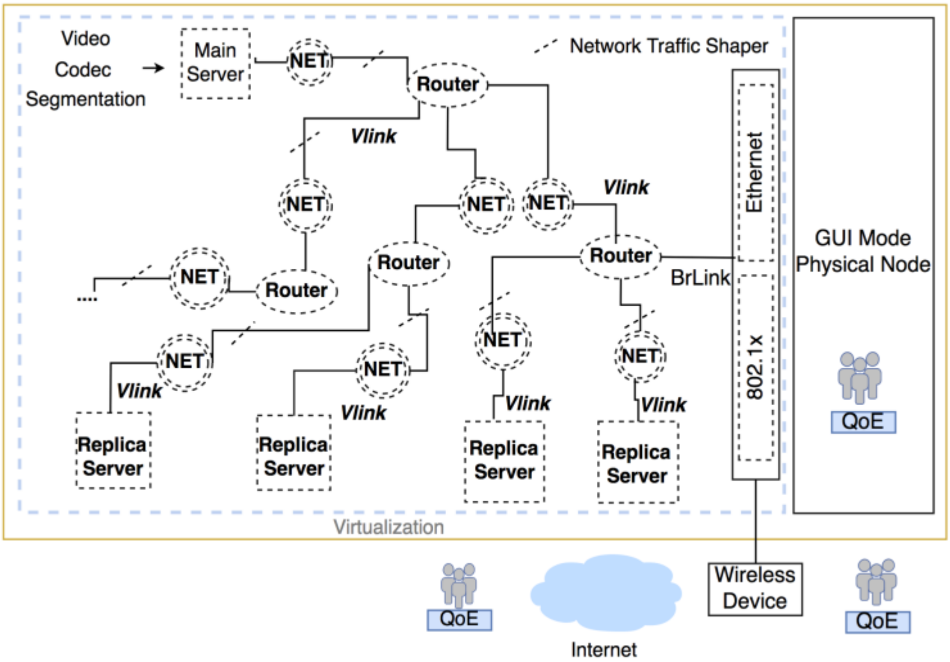


Figure 7.6. Topology of the scenario.

Algorithm 1: Sample code of how to generate a scenario

```
<global>
   <scenario_name>CDN+WirelessNetwork </scenario_name>
<vm_defaults>
     <console id="0" display="no" />
    <console id="1" display="yes" />
<cmd-seq seq="command 1">commandfunction</cmd-seq>
.
.
<cmd-seq seq="command n">commandfunction</cmd-seq>

  </global>

   <net name="Net0" mode="virtual_bridge" />
.
.
<net name="NetN" mode="virtual_bridge" />

<!--Orginal Server -->    <vm name="ORG-SERVER" type="lxc">
 <filesystem type="cow">/usr/share/vnx/filesystems/rootf_ubuntu    </filesystem>
  <if id="1" net="Net0" name="eth0">
 <ipv4>192.168.0.1/24</ipv4>
 </if>    <route type="ipv4" gw="192.168.0.10">default</route>
<forwarding type="ip" />
 <exec seq="on_boot" type="verbatim" ostype="system">
  sudo “Run functions”
  sudo  “Run functions”
</exec>    </vm>

<- Add more devices-- > <vm name="new device1" type="lxc">
<- Add more devices-- > <vm name="new deviceN" type="lxc">

<!-- External Host, include also the application...-->
    <host>
<exec seq="type of funtion" type="verbatim">fucntion</exec>
   </host>
```

7.4.2 Content distribution

The content distribution can be push or/and pull; in our scenario we provided proactive pushing mechanism, which is involved to push the segmentation of the video proactively on to surrogate servers according to the algorithm 2, which is showing the process of the pushing video segments onto surrogate server (cache servers) by using different application layer protocols to place the video segments in order to become closing to end-users. SS*g* is denoted by surrogate servers and, Video_*i*, where *i* is a video with a number of the videos.

Algorithm 2: Pseudo code distribution multimedia service content

---

Original server:

1 Select Video_*i* should be replicated among surrogate servers
2 While True
3. For SS*g* = 1 to N
4. Select application layer protocol to send Video_*i* toSS*g*

Surrogate Servers:
5. Listen to main server requests
6 While True
7 receive the Video_*i*

7.4.3 Client redirection

The redirection mechanism is used to redirect clients request to optimal surrogate server among cache servers, it uses feature of ICMP protocol. The system is calculated latency and packet loss between the main server and surrogate servers to the clients' side. The multimedia web servers are operated by CDN operators inspecting requests from clients and redirect it to the most appropriate surrogate servers by rewriting the URL of the client to optimal surrogate server. This mechanism provides flexibility in serving content to users with better QoS. Users can be served location specific content by redirecting their requests to suitable surrogate servers. In the case, users cannot find the segments of the video, the system tries to rewrite the URL to next the optimal surrogate. The process of client redirection is explained in Algorithm 1 in detail.

Algorithm 3: Pseudo code redirection client request to optimal surrogate server

```
Client side:
Ci sends GET to request multimedia web service

Original Server:
Receive Ci request in http message
Capture Ci IP address
For SSk = 1 to N
Send Ci IP address to N surrogate servers through http
End For

Surrogate Servers:
Receive the http request from original server contains IP of Ci
SSk apply asynchronous task
For SSk = 1 to N
Send 10 packets to client IP address
End for
Calculate average of latency && ratio of packet loss between SSk and Ci

Original server
While SSk do not send response or time out
If SSk has responded
Calculate minimum latency [k]
EndIf
EndWhile
Rewrite URL to surrogate server [k] has minimum latency

Client side:
Receive web service and list of videos from surrogate [k] has minimum latency
```

7.4.4 System function and QoE

The process of service delivery and obtain the information of the QoE is depicted in Figure 7.7. When the video content is delivered to the end-users from one of the content providers, the objective observation parameters such as initial delay, buffer length , quality oscillation and video stalls are saved in a data set in order to evaluate the QoE of end-users. The obtained formation is sent over HTTP protocol to main server in order to provide QoE assessment according to the QoE evaluation as explained in next subsection.

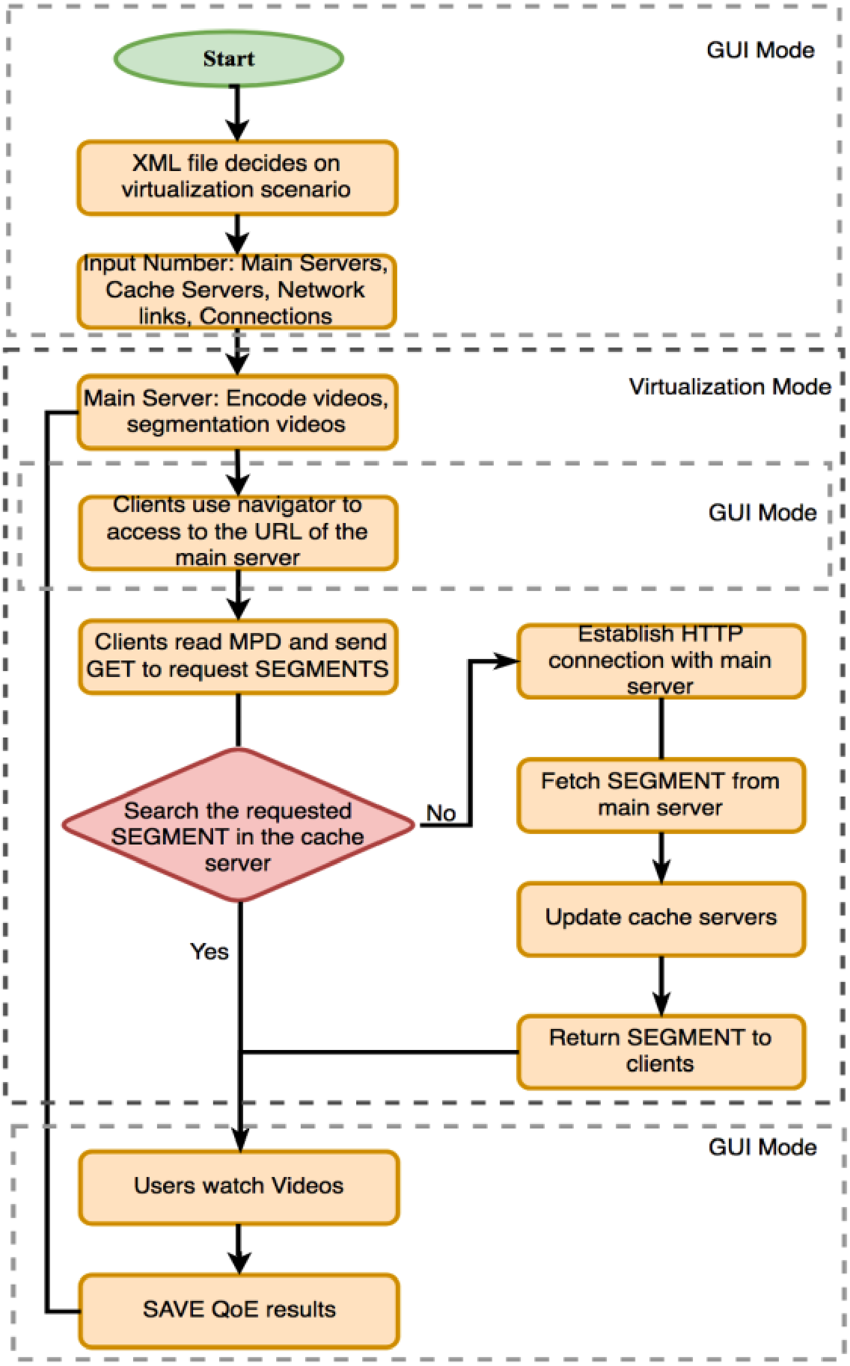


Figure 7.7. System function and capture QoE information.

7.4.5 QoE evaluation algorithm in the system

QoE evaluation is presented in Figure 7.9, according the parameters is found to decide on the users' QoE (See section 5). The algorithm is based on the number of the representation, maximum quality throughput, and the network behavior. If the evaluation of the QoE is high than the threshold values, then the user can be satisfied with the service, otherwise the user can not be satisfied with the service.

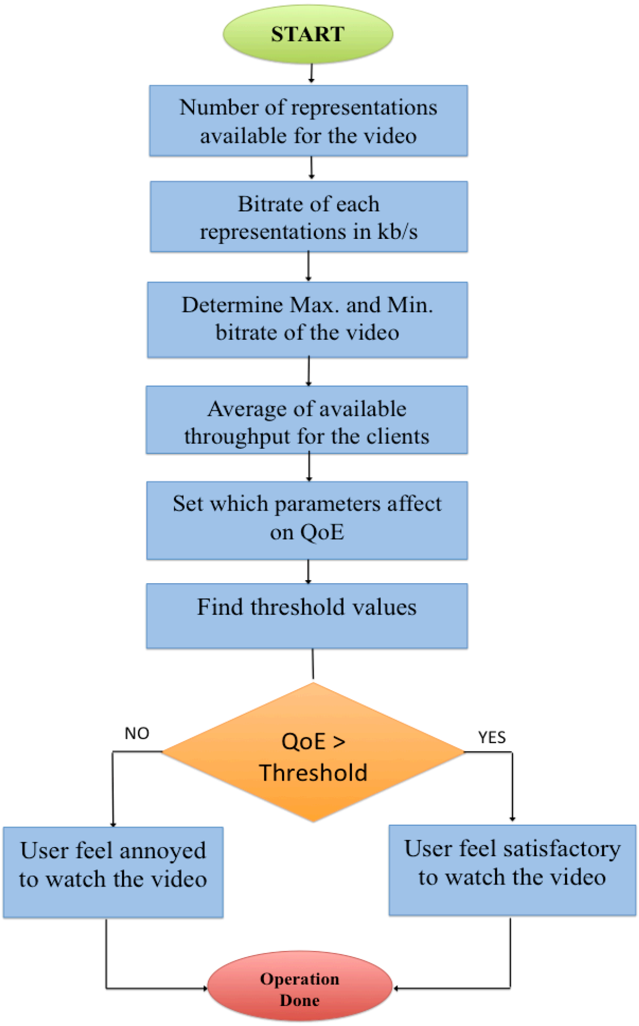


Figure 7.8. Evaluation QoE in the system.

7.5 Experiments and performance evaluation

This section includes experiments and evaluation of the protocols and algorithms, which have been used in the design of the virtualized system testbed. In order to provide clarified experiments for each layer of the system. We provided different subsections to give detail on the experiments as described next.

7.5.1 Experiment 1: Distribution and protocols

In the distribution mechanism, the main server system selects an appropriate application layer protocol that appropriately deliver small chunks of adaptive video streaming among surrogate servers. We select three different application layer protocols that are based on TCP transport layer, respectively includes: FTP, HTTP and web distributed authoring and versioning (WebDAV), in order to choose the better application layer protocol that suit to fast deploy small chunks of media content among surrogate servers. The distribution process includes the utilization command lines such as Cadaver , Curl and FTP client to push the media segments from content provider into surrogate servers. In addition, the video chunks automatically keep in the local cache of each surrogate servers. Therefore we analyzed the performance of TCP of each application layer protocol according to number of the packets that can be retransmitted, and TCP throughput. Furthermore we utilized Wireshark to capture information of sent/received packets in the networks. To apply the experiments we merely determine one surrogate server to replicate adaptive video contents. Network traffic shapers are used to impair the bandwidth, delay and packet loss. Each experiment has been repeated 10 times to get accurate results. So in the first experiment we impair the bandwidth into different ranges that consists of 275Mbps, 45Mbs, 5Mbps and 1.54Mbps in order to find out the performance of which application layer protocol waste less time in the process of replication video contents in the surrogate server, as depicted in Figure 7.9. The blue bar indicates the transmission time of entire chunks that took through WebDAV protocol, the green bar for HTTP and the red bar for the FTP transmission. The blue bar in this experiment wasted less time to deliver entire chunks of the adaptive video streaming, also expanding the bandwidth straightly affected on arrival time of chunks especially in WebDAV.

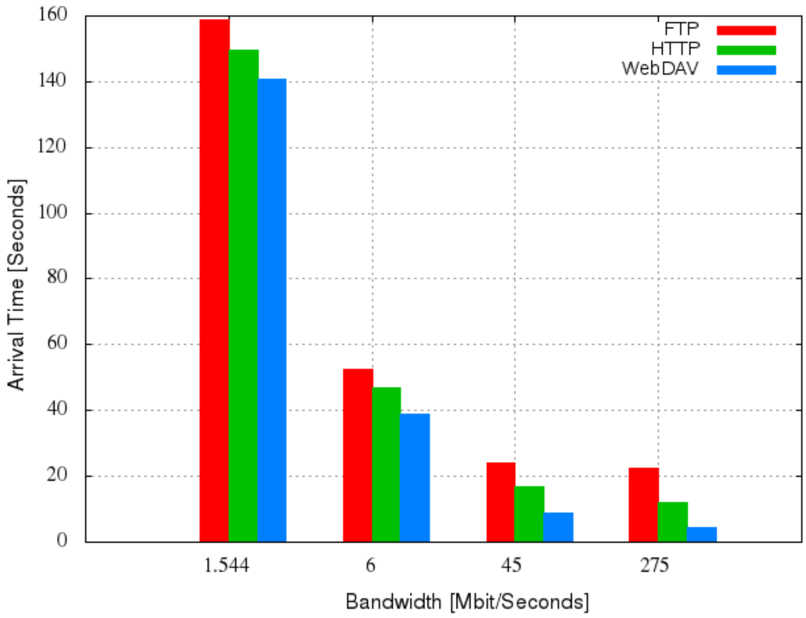


Figure 7.9. Arrival time of segments.

Different values of delays (round trip time) in the next experiment has been proposed as shown in the Figure 7.10, in order to discover the effect of latency on arrival time of replication media segments on the surrogate server. The blue line is the WebDAV protocol that has been moved up slightly from 20ms to 120ms. The red line and the blue line have been turned up highly especially the red line used much more time to arrive video chunks. Therefore, increases in latency between original server and surrogate server have a huge impact on delivery chunks in FTP and HTTP.

In the other experiment, as shown in Figure 7.11, we set the range of packet loss between 0% to 2.5% with 6Mpbs of available bandwidth and delay 20ms. In this experiment, packet loss rates highly affected of the delivery time of chunks through FTP and HTTP than WebDAV. Also WebDAV has barely moved up from its position.

In other experiment, we disclose the performance of transport layer (TCP) for each application layer protocols regarding to bandwidth usage. According to the results are depicted in Figure 7.12. In this experiment we constrain the bandwidth to 6mbps, 20ms delay and 0.1% packet loss as well.

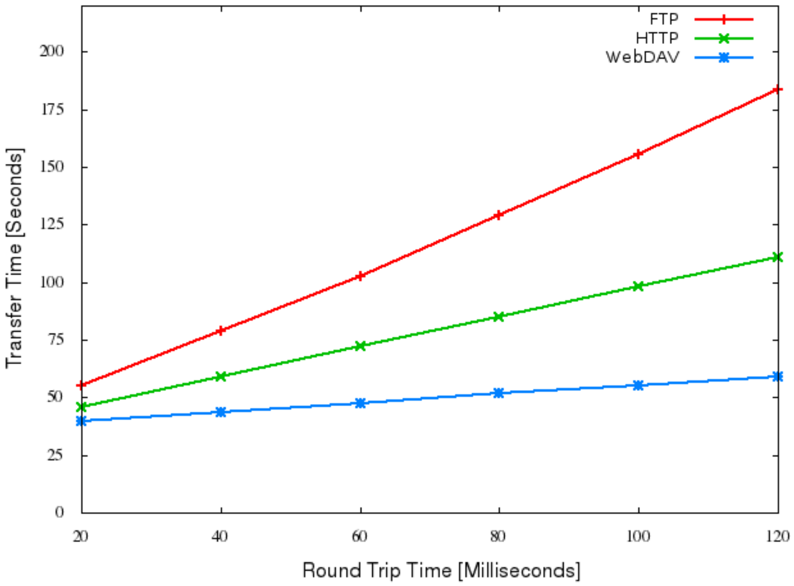


Figure 7.10. Effect RTT on arrival time of segments.

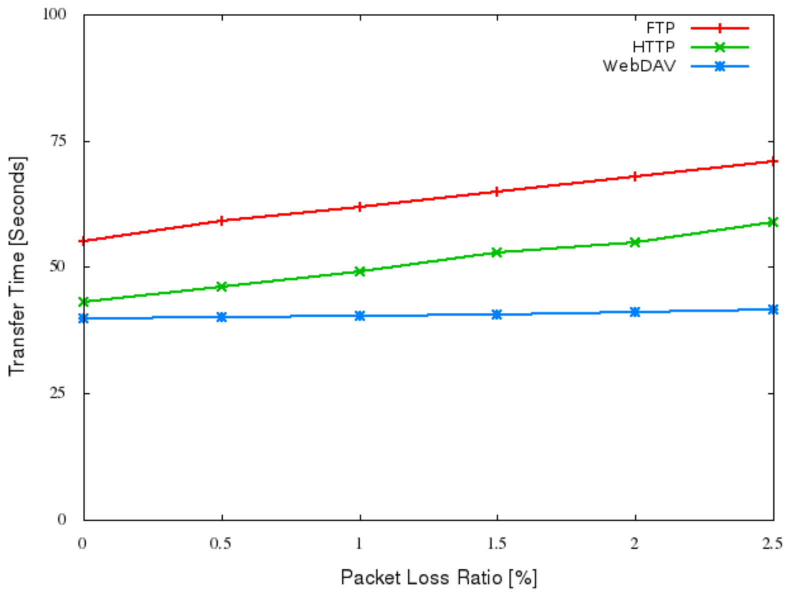


Figure 7.11. Effect packet loss on arrive time of segments.

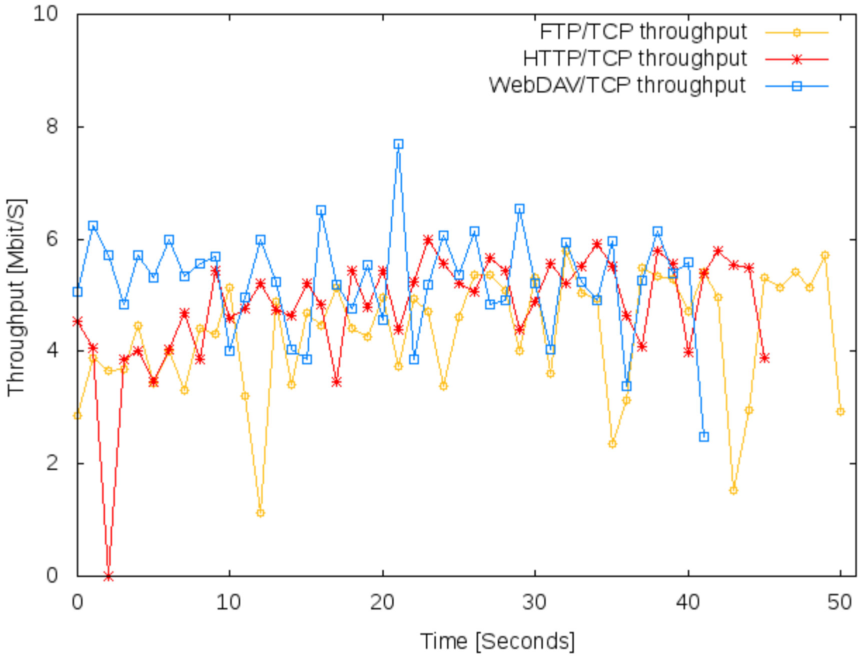


Figure 7.12. Performance of the throughputs.

The blue line uses maximum rate of available bandwidth, it is approximately between 5Mbps to 6Mbps, and its high pick rate in the second 20 to deliver video chunks and the transmission takes 40 seconds to deliver all chunks of the video. The red line in the initial delivery of chunks depredated to low value, this makes the protocol to use fewer packet size to arrive on surrogate server and the duration of whole time transfer of chunks took 45seconds. Also the brown line of the figure is used lower bandwidth to transferring packets. Further the brown line has been dropped to use 1.5 Mbps in second 12 and the total time was 50 seconds. In the other experiment, number of packets is illustrated in Figure 7.13, which describes retransmitted of packets in original server through TCP transport layer of three application layer protocols. For this experiment, 0.1 percent of packet loss is set and 6mbps available bandwidth and 20ms delay in the shaper. In the red line indicates numbers of packets, which have been retransmitted in FTP is higher than the green line. Therefore the blue line indicates TCP of WebDAV non-packet is retransmitted. It means that packet loss and delay barely have effect on WebDAV protocol to deliver small chunks. size longer than HTTP and FTP. Persistent connection, pipelining and permanently connected to destination until task finish makes HTTP and WebDAV fast, easy perform better than FTP

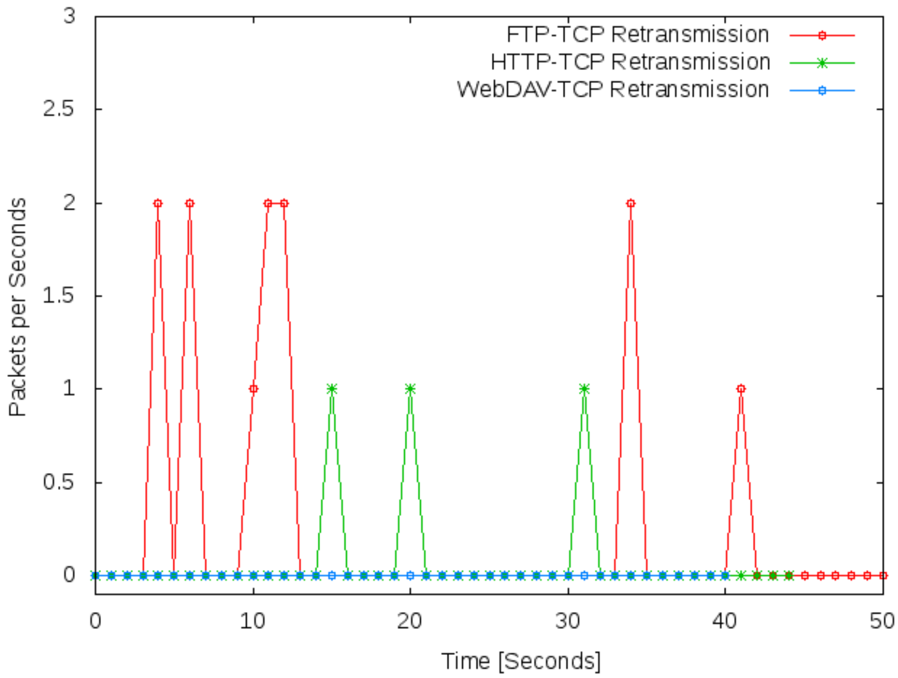


Figure 7. 13. Retransmission of packets.

7.5.2.Experiment 2: Redirection approach

In this experiment client's request to closest surrogate server is observed, the request routing mechanism involves to sending series of IP layer packets through ICMP protocol that reported in PING command to client side, maximum range of packet size is 63353 bytes. Also sending maximum size of packets and small packet size are not the optimal solution in the redirection process because maximum packet size is producing high network congestion and CPU load on between both devices especially in constraint networks. Moreover sending series of small packets sizes are not shown accurate results. In this experiment we employed 128-bytes, including 120 bytes of data and 8 bytes of ICMP header to check the connectivity and calculate minimum average latency between surrogate servers and clients. Once the client sends a request to original server to get multimedia service, the original server sends the client IP address (the client IP address captured through the PHP program) to surrogate servers through HTTP protocol. Surrogate servers use multi-thread process to send asynchronously packets to speed up executes tasks and reduce client page load as depicted in the Table 7.1. Each surrogate server sends a packet of 128 bytes for 10 times and the interval between each packet is 0.2 seconds.

Table 7.1. Page load time.

CDN request route	Client page load
Not apply asynchronous	1.02 seconds
Apply asynchronous	0.45 seconds

Results of packet loss and latency are reported to original server in order to respond to client requests by rewriting the URL to optimal surrogate server address and then the client obtains the web page service application from optimal surrogate server, with the list of available videos. The client can retrieve the media segments of adaptive video streaming from its corresponding surrogate server. On the other hand if the segmentation of video is not available in the corresponding cache server, the same cache server brings it from the original server to respond to client requests. In the redirection mechanism the series of packets send between IP layers of surrogate server and clients to specify which surrogate server become an appropriate server to provide service to end-users. Also clients can see list of available videos that pushed from original server to correspond surrogate server. To evaluate the performance of distribution the segmentation of adaptive video streaming in the CDN testbed we provide different impairment of available bandwidth, high and low distance (delay) between content provider to a surrogate server and then vary range of packet loss ratio in order to find out arrival time of video chunks.

7.5.3 Experiment 3: Simultaneous connection

In to provide speed-up delivery the segmentation of multimedia content streaming from the surrogate servers. The Application's client can request segmentation of the adaptive video streaming simultaneously from different surrogate servers. The application opens multiple persistent Transport Control Protocol (TCP) connections so that different content can be served simultaneously from the same server or multiple servers. Persistent connections allow the same TCP connection to send and receive multiple HTTP requests/responses instead of opening a new TCP connection for every request/response pair. When an application opens fewer TCP connections and keeps them open for a longer period, it causes less network traffic, uses less time establishing new connections, and allows the TCP protocol to work more efficiently. Table 7.2 shows the comparison request time of demanding some segments together due to use simultaneous TCP connections. In the table, the relation between both approaches is shown in arrival time in seconds, as depicted, using multiple connections brings segments to the clients as fast as a TCP per connection.

Table 7.2. Comparison between traditional approach and multiple TCP connection.

Number of segment s	Arrival time Based on traditional (Sec.)	Arrival time based on simultaneous TCP (Sec.)
2	0.10	0.4
4	0.23	0.11
6	0.26	0.13

7.5.4 Experiment 4: QoE assessment

The Big Buck Bunny raw video is used to perform the QoE experiments in the proposed testbed system. In order to provide DASH content at different bitrates, 18000 frames (60 fps) are encoded with Lib-X264 and GPAC-MP4Box. We provide 20 representations that consist of three levels of video quality: low, medium, and high definition quality. Each level of quality provides different bitrates, which are sequenced from low quality, which has a resolution of 320x240 with a bitrate of 50 kbps (baseline profile), to the ultimate quality, which has a resolution of 1920x1080 with a bitrate of 5000 kbps (high definition profile). In fact, we provide 7 versions of the video with different segment duration starting with a segment length of 1 second, and then 2s, 4s, 6s, 8s, 10s and ending up with 15s. Therefore, to conduct these experiments, the output data have been collected from the aforementioned parameters. Information of the video application has been shown in Figure 7.14 and Figure 7.15.

In the first experiment, the initial delay was varied. It depended on the segment duration. This fact is shown in Figure 7.16. The initial delay increased when the segment length increased. Generally, the tendency of these lines is to be uniformly related to the increase of the segment length. Segment with a duration of 4 seconds appeared as inflection point associated with the others. Moreover, few seconds of unrestricted bandwidth led to reduce the initial delay of the playback video in large segment durations compared to short segments as shown by the green line of the figure.

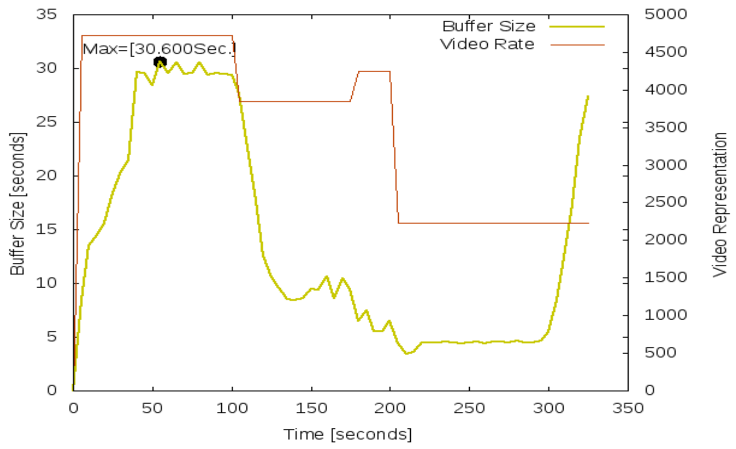


Figure 7.14. Detect buffer length and quality video oscillation.

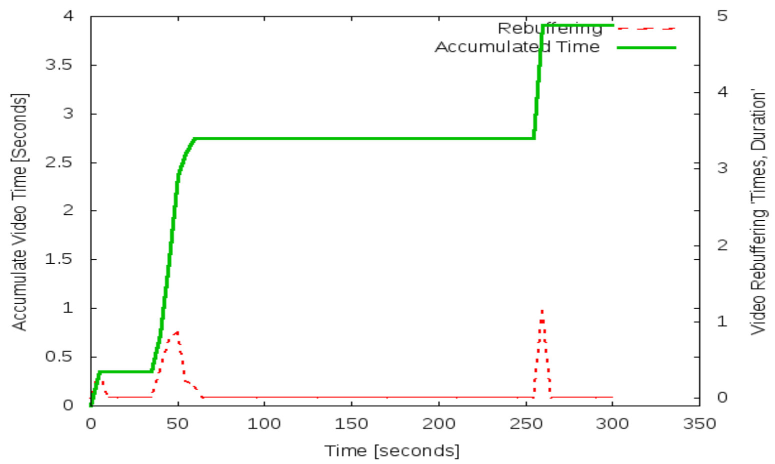


Figure 7.15. Detect stalling and accumulative video time.

In the second test, two different tests are performed with the system as in the previous experiment. The goal was to evaluate the number of switches (for different video qualities). In those tests, the relationship between the different numbers of switches for the 4 seconds segment duration was observed. It is shown in Figure 7 .16. The blue line represents the first test results and the red line represents the second one which quality oscillation has depicted 4 and 6 times with 1 second segment duration after the decreased tendency of the line reached the 4 seconds segment duration. After 4 seconds, the lines of the segment duration coincided until the end of the video.

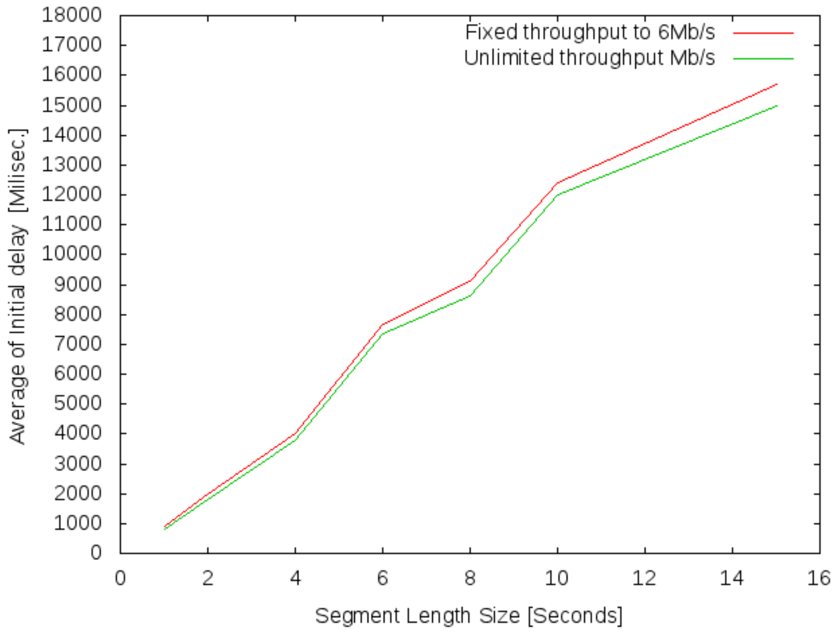


Figure 7.16. Average of initial delay of segments

We used the same procedure in the third experiment, the tests for accumulative video time. The segments with duration of 4 seconds appear to be the inflection point between pitch curves with a low tendency curve as shown in Figure 7.18. From the results, it can be noted that increasing the segment duration for both available network throughputs causes increment of the accumulative time. The accumulative time curve rises up rapidly when the short segment duration is being used. Meanwhile, ascending the curve slowly above the 4 seconds duration means that the accumulative time of short segments is 35% higher than long segments. Figure 7.19 depicts the results described in the fourth experiment, which is performed to measure the effect of the buffer length. The curves show that, after 4 seconds segment length, higher buffer size is used.

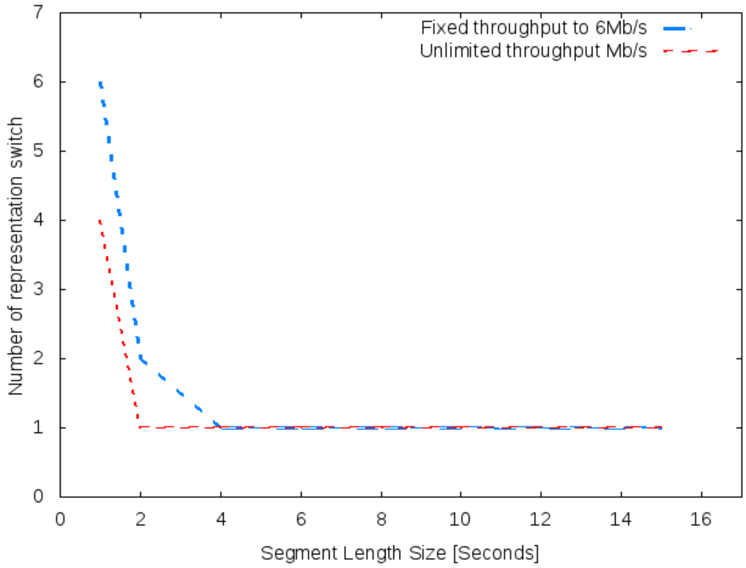


Figure 7.17. Number of switches for the segments

In the subjective evaluation, in order to obtain the DMOS value 10 video experts evaluated the received video. The assessment process of DMOS is classified into several groups; DMOS for the initial delay (play the video during 25 seconds) and DMOS for video stalling and switch quality (play the video during 300 seconds). The classification approach is aimed to find an accurate subjective evaluation. As a consequence, to carry out the process of DMOS, the original video and the received video at the destination are shown to the experts. The ratio scale is indicated from 1 to 5. 1 indicated that most users are dissatisfied, 2 indicted that many users are dissatisfied, 3 indicated that some users are satisfied, 4 indicated that most users are satisfied and 5 indicated that most users are very satisfied. The results of the subjective evaluation are presented in Table 7.2. We observed that when assessing the switch of the quality of the small segments, we obtained worse results than in large segments. Otherwise the initial delay of large segments presented the worst case.

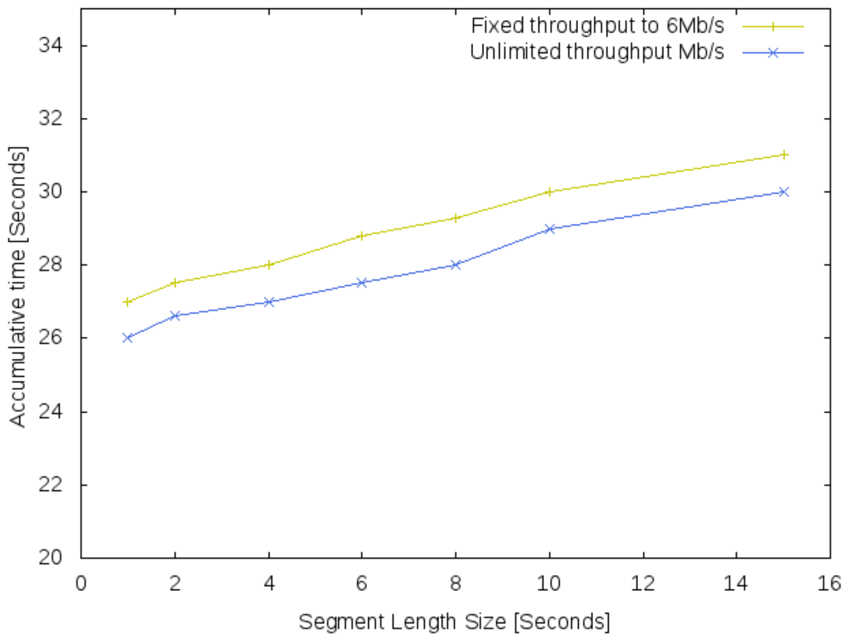


Figure 7.18. Accumulative video time.

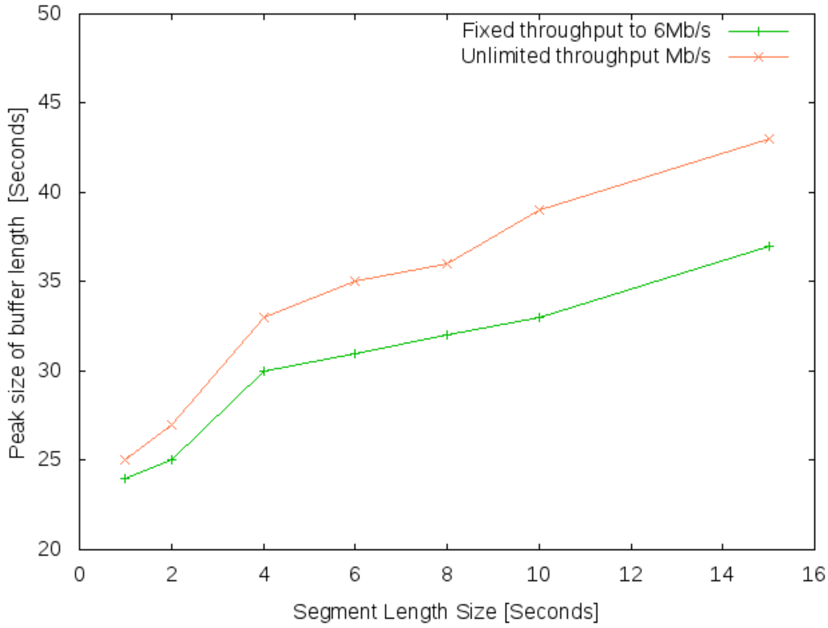


Figure 7.19. Buffer length

Table 7.3. DMOS comparison.

Segment Duration (Sec.)	Initial Delay (Sec.)		Switch quality (Times)		Stalling video (Times)		DMOS Initial Delay		DMOS Switch & Stalling	
	Restricted	Unrestricted	Restricted	Unrestricted	Restricted	Unrestricted	Restricted	Unrestricted	Restricted	Unrestricted
1	1,1	1,01	6	4	7	5	**4**	4	**1**	2
2	2,4	2,04	2	1	3	2	**4**	4	**2**	2
4	4,1	4,02	1	1	3	2	**3**	4	**3**	3
6	7,8	6,89	1	1	3	2	**3**	4	**4**	3
8	9,100	8,79	1	1	3	2	**3**	4	**4**	4
10	15,50	10,3	1	1	3	2	**2**	3	**4**	4
15	16,01	15,2	1	1	3	2	**2**	2	**4**	4

7.5.5 Experiment 5: Resource usage assessment

The CPU usage was also studied with this testbed for both network throughputs. In this experiment, the average of the CPU load is captured during 300 seconds for the segment lengths 1, 2, 4, 6, 8, 10 and 15 seconds. Results are plotted in Figures 7.20 and 7.21. From the obtained results one can observe the following:

- Short segment duration uses shorter peak of CPU usage to request a segment than segments with higher duration.
- Short segments consume upper bound of CPU during decoding the video. This effect is the opposite in longer segments.

The CPU load decreases differently in different network environments (by 60% from 1 to 4 seconds of segment duration). From 4 to 15 seconds of segment duration the CPU load is decreased by 8%. The average energy consumption is presented in Figure 7.22. The energy consumption in small segments is higher than in large segments for both network connections, which can be estimated 30% higher. Retrieving small segments makes the clients to establish many connections to download into buffer's application, so the device consumes higher energy than when it uses large segments.

Finally, in order to quantify impact on the resources because of the CPU usage and energy consumption by virtualization of the devices during the experiments, we have monitored the physical hardware (see Figure 7.23). The number of virtualized devices affects both CPU time and the energy consumption. It is observed that when the number of VMs is increased, the CPU metric values are dramatically increased. Therefore, The power consumption of the raw device is also consumed high rate of the batter life when the all users access to playback the adaptive video streaming.

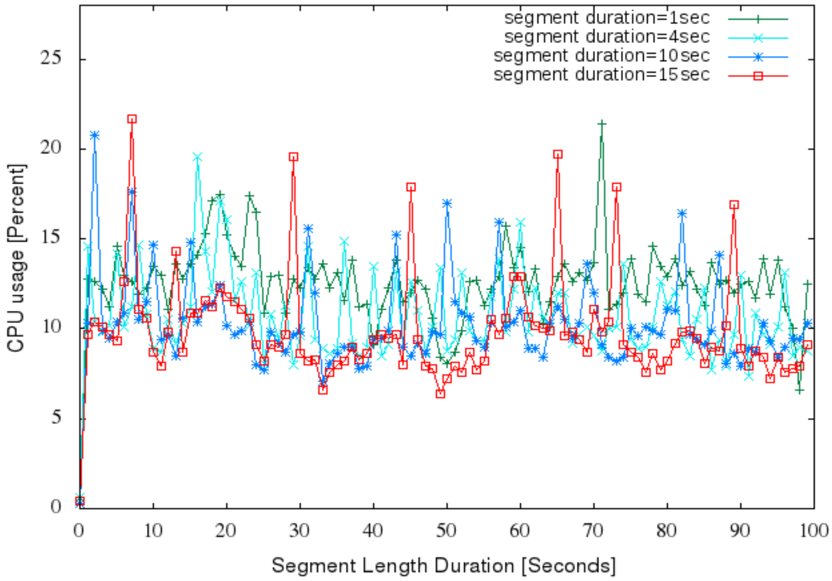


Figure 7.20. CPU information for different segment length

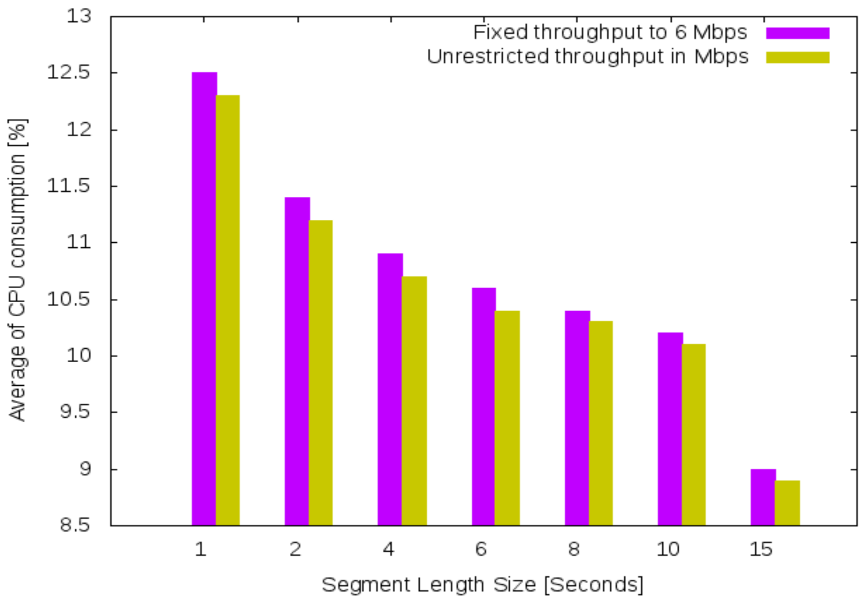


Figure 7.21. Average of CPU usages for different segments.

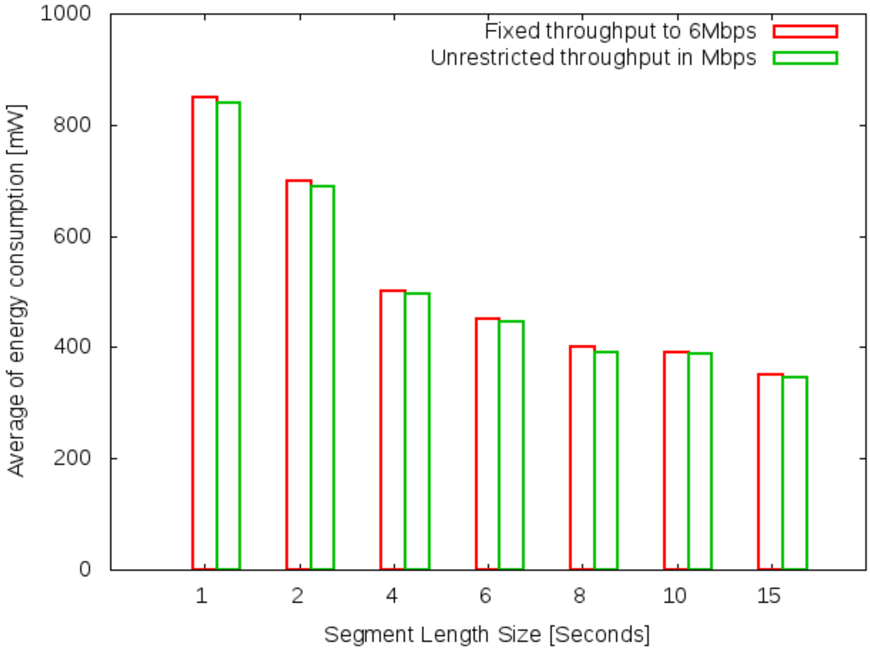


Figure 7.22. Energy consumption of physical devices for different segments

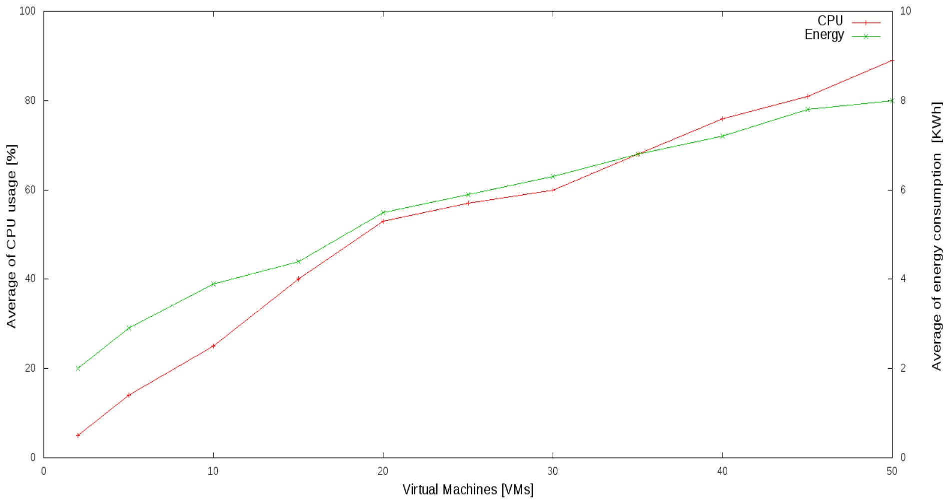


Figure 7.23. CPU and energy consumptions of the system

7.6 Results analysis and benchmark comparison

According to the subjective and objective results obtained from the experiments. In adaptive video streaming, segments sizes are factors that affected on QoE. Segment length can be generated in second, which can be started from 1 second to maximum length. Segments length from 1 to 4 are given lower initial delay however they increased oscillation of video quality and created small buffer size. Segments length of 6 and 8 had some seconds higher initial delay than small segments but they had sufficient buffer size and less oscillation of the quality. Therefore segments length of 10 and 15 are created higher initial delay than segments length of 1, 2, 4, 6, and 8 and they also contained high efficient coding. However, segments of 10 and 15 faced to annoyance end-users when a user had a fluctuation of throughput because of the stall duration of video was longer than previous segments and large segments needs more time to fill buffer size of the application. Therefore, resource usage of the devices are changed according to segments sizes, in restriction and oscillation client's throughput, small segments consumed higher CPU usage and battery power because of the establishing many connections with the server to request the segments. Otherwise, large segments consumed less CPU and energy where there weren't enough bandwidth to fill application buffer length.

Benchmark parameters such as physical size, mobility, scalability, realism of the tests, cost-effectiveness, network simulation and emulation, core network and etc. between our proposed testbed scheme and other testbeds are shown. The impact of factors on the user experience according to the performed tests, which video segment length provides better QoE, are also presented. Table 7.4 shows the similarity and dissimilarly of the network testbeds parameters.

Table 7.4. Benchmark comparison between our proposal of testbed and current testbeds

Testbed name	Simulate testbed Ref. [152]	Real testbed Ref. [153]	Simulate testbed Ref. [154]	Simulate testbed Ref. [155]	Flamingo Ref. [156]	Our Proposed testbed
Tested physical	1 Node	5 Nodes	1 Node	1 Node	1 Node	1 Node
Mobility	Easy	Low	Easy	Easy	Easy	Easy
Scalability	High	Low	Medium	High	High	High
Cost efficiency and dollar-cost per device	Medium	High	Low	Medium	Medium	Medium
QoE Method based on	-	-	-	-	-	Decision algorithm.
QoE Metrics	Only video quality oscillation	Initial delay, video stalls, switch frequency	CPU, TCP.	Video stalls, buffer size, switch frequency, MOS	QoE based only on (MOS)	Initial delay, buffer length, video stalls, switch frequency, DMOS, CPU and energy
Realism	Combine of Simulation and emulation	%100 Real	Simulation (Math. Models)	Simulation (Math. Models)	Combine of Simulation and emulation	Emulation (Live network)
Topology of the core network (routers and switches) based on	Simulation	Real	Simulation	Simulation	Simulation	Emulation
Cache service	-	-	-	-	Support	Support
Machine Learning-based Virtual Network	-	-	-	-	Support	No defined yet
Preferable segment length	Not defined	Small segments	Not defined	Not defined	Not defined	6-8 seconds duration
Language used for describing the simulation	Python	-	-	-	Python	XML
VMs Operating system based	Linux	-	-	-	Linux	Linux
VMs Chain distribution	-	-	-	-	-	Support
Network topology presented for experiments	Simple	Simple	Simple	Simple	Complex	Complex
Description	Virtualized network testbed to perform experiments by Mininet	Real nodes to design real testbed to experiment the QoE	Simulate Platform for mobile video streaming using OPNET.	Simulate Platform for LTE-mobile video streaming using ns-3	Virtualized network testbed to perform experiments by Mininet	Virtualized network testbed to perform experiments and run several virtual nodes on a single physical machine

7.7 Chapter Conclusion

This chapter has proposed a virtualized testbed design to conduct adaptive video streaming experiments in different network environments. The proposed system presented a virtualized complex network scenario and various virtualized nodes with different functionality. The system presented the virtualized network scenario as Content delivery Network (CDN) for distribution and delivery of multimedia streaming service to end-users. Different applications and mechanisms are used in order to provide better performance of delivering video content to end-user with minimal latency.
Therefore, the virtualized testbed is used to evaluate the QoE of adaptive multimedia streaming on the Internet scenario. To select the process of the QoE evaluation, some objective metrics and a subjective metric are presented. The content providers can aware about QoE of its clients. Clients can inform the main servers according to the parameters are used to observe the QoE. The evaluation approach leads to main server to determine users' satisfaction or annoyance. Therefore, in order to evaluate the QoE in the system, different segment size is prepared, from the observation results, the segment length between 6 to 8 was provided better QoE in the system.

Chapter 8. Conclusion

8.1 Introduction

Multimedia streaming has been a very important topic in last years; today and in future, its importance is undoubtedly recognized in the academia and industrial segments of society.
Video streaming has special characteristic that impose to take care when we consider the huge amount of users that are wanting to send and to receive video sequences, using tools as YouTube, Skype, Netflix, Facebook and Hulu, and being served by wireless and wired networks, commonly integrated by Internet, a network disseminated around all parts of the world. These video sequences commonly request a great amount of bandwidth and impose the treatment of real time requirements, which are different from the requirements previously adopted for the design and deployment of Internet.

This dissertation focused on several aspects related to QoE related to users consumption of video streaming. Initially, we've made a bibliographical review on concepts and techniques related to multimedia streaming.

Then, we presented discussions about QoE of Internet multimedia streaming in heterogeneous networks. Therefore, We provided evaluation of multimedia streaming in wireless networks; from the evaluation, we conclude about how the different metrics affect on the QoE of the video streaming.

We provided methodology for QoE evaluation in the social network environment. The subjective methodology evaluates QoE according to the study factors have been investigated for HTTP adaptive streaming. We concluded about how the proposal can provide better QoE evaluation.

We provided algorithms to optimize QoE based on management of QoS in heterogeneous wireless networks; Wi-Fi, and cellular networks. From the control of traffic and resource management, we conclude that how the optimization approach improve the QoE of multimedia service, which offers video streaming as both on-demand content and live streaming.

We provided virtualized system testbed considering massive network scenario. From the plan of shared and management resources with different operations. We concluded that the designed system provides cost-effective for QoE evaluation in multimedia services providers.

8.2. Conclusions and contributions

We consider that the previously established objectives of this PhD thesis were reached, as we discuss in the sequence, talking about these objectives, as well as the conclusions related to each one.

The main objective was to meet the expectations of assessing and managing the QoE of real time multimedia streaming; this objective was broken on specific objectives, as described below.

- Initially, the objective "intensive comprehensive survey of the state-of-the-art concerning QoE of real time video streaming and on-demand adaptive videos streaming." was reached, allowing understanding the subject and indicating the potential ways to address the existing challenges in these regards.

- Then, the objective "Investigation and identification of the key factors are influencing the perceptual quality of real time streaming and on-demand adaptive video streaming for the current service applications." was reached, allowing selecting important metrics to evaluate the QoE. Among these factors, we observed that the QoE metrics such as bandwidth, delay, packet loss and jittering have great influence on QoE; other factors such as device characteristics, multimedia feature and service providers also must be considered for QoE evaluation.

- The objective "Improving the subjective test methodology for the QoE evaluation. Take laboratory controlled approach to collect the subjective dataset with respect different parameters." was reached, allowing evaluating QoE for different multimedia application. The methodology consisted of taking different parameters to evaluate subjective QoE, such as segment size, initial delay, stalling (duration, place, replication), switching (sharp and smooth) are one of the most important and original part of the proposed methodology involves estimate QoE of viewer in better way; several laboratory experiments were conducted, with different kinds of devices, and several types of video sequences. The methodology proved to be efficient for colleting and processing data related to parameters involved in video streaming, according to previously established metrics.

- Then, for the objective "To design a smart system to monitoring and evaluating the QoE, and correlation between QoS and QoE", we study machine learning techniques and investigate on which approach should be selected to be included in our system. This objective was reached, leading to an accurate and robust prediction of QoE, allowing to insert machine learning based algorithm and QoE prediction capability into our system. According to the evaluation have taken in the system, in adaptive video streaming, better segment to generate in content provider is between 1 to 10 seconds duration from the range 1 to 30. Although segment lengths 6 to 8 in the system provide better QoE regarding to consumption of the resource usages of devices.

- Then, the objective "To select appropriate application layer protocols to be used for distribution multimedia content across Internet to provide the best QoE possible" was reached, after comparing different application layer protocols to push video contents, algorithms and mechanisms in a CDN-based approach was considered, jointly with specific mechanisms for distributing multimedia content, accordingly its characteristics of proximity, measured with basis on parameters as latency and number of hops is considered. According to protocols based push-approach, WebDAV is better and faster than FTP and HTTP to replicate video content among the replica servers and the media content available in short period to the end-users which reduce waiting time of playback video streaming. We concluded about the superiority of QoE is high when of the system based in accurate and robust algorithms and protocols. Rate of packet loss in WebDAV is less than both FTP and HTTP. And estimation client request to proximity content provider based on synchronization approach is provided better QoE when a client request video content.

- Then, the objective "Develop cost-effective virtualized system testbed for evaluating QoE." was reached, allowing to provide multimedia service using virtual resources, instead of using real testbeds, commonly expensive. The virtualized system was composed by software components freely available, that were integrated in a system able to collect the video sequences, compress and transmit them, for evaluation of the quality of the received sequences at clients served by heterogeneous networks.

8.3 Future lines of research

As a future work we propose to vary the type of features as shown below

- A sophisticated design will be implemented in order to carry out more complex experiments. We are going evaluate the effects of huge number of clients' competition over different types of wireless networks.

- We are planning to integrate different types of video codecs and video formats to the system in order to evaluate QoE of the end-users for both adaptive video streaming.

- Using smart approach bases on machine learning to monitor health of the system regarding to the resource sharing.

- In adaptive video streaming, region of interest can be interested to provide better QoE. Region of interest can be streamed with high quality and rest of the video adapted to lower quality in order to considering resource usages.

- A security and data integration system for the designed architecture has not been proposed. Future works could design a security scheme that implements, in a distributed way, the security of mulimedia service can be interested to invesitated in the system

- Implementing the functionality On-the-fly transcoding for both live and on demand adaptive video streaming can be interested.

8.4 List of Publications derived from the Ph.D. thesis.

At end, we express the attendance to the mentioned objectives by a set of publications, listed below

Journals	Description
1	Miran Taha; Aree Ali; Jaime Lloret; Laura Garcia, “Adaptive Video Streaming Testbed Design for Performance Study and Assessment of QoE.” *International Journal of Communication Systems,* 2018, DOI: https://doi.org/10.1002/dac.3551
2	Miran Taha; Jaime Lloret; Alejandro Cánovas; Laura Garcia. “Survey of Transportation of Adaptive Multimedia Streaming service on Internet.” *Network Protocols and Algorithms,* Vol. 9, No 1-2, 2017, DOI: https://doi.org/10.5296/npa.v9i1-2.12412
3	Miran Taha; Jose M. Jiménez; Alejandro Canovas; Jaime Lloret, Intelligent Algorithm for Enhancing MPEG-DASH QoE in eMBMS*” Network Protocols and Algorithms*, Vol. 9, No 3-4, 2017.
4	Jaime Lloret; Lorena Parra; Miran Taha; Jesús Tomás. "An architecture and protocol or smart continuous eHealth monitoring using 5G." *Computer* Networks, Vol. 129, No. 2, 2017, PP. 340-352, DOI: https://doi.org/10.1016/j.comnet.2017.05.018
5	Alejandro Cánovas; Miran Taha; Jaime Lloret; JesúsTomás, “Smart resource allocation for improving QoE in IP Multimedia Subsystems. ” *Journal of Network and Computer Applications,* Vol.104, No. 15, 2018, PP. 107-116, DOI: https://doi.org/10.1016/j.jnca.2017.12.020
6	Jose M. Jimenez; Jaime Lloret; Miran Taha; Sandra Sendra, “Interactive Videos in IPTV using Hypervideolinks.” *Network Protocols and Algorithms*, Vol. 9, No 3-4, 2017. http://dx.doi.org/10.5296/npa.v9i3- 4.1254
Conferences	
1	Laura García; Jose M. Jiménez; Miran Taha; Jaime Lloret, “Video artifact evaluation based on qos and objective qoe parameters”, *2017 International Conference on Advances in Computing, Communications and Informatics (ICACCI).* Udupi, India, 13-16 Sept. 2017, DOI: https://doi.org/10.1109/ICACCI.2017.8125980
2	Miran Taha; Laura Garcia; Jose M. Jimenez; Jaime Lloret, “SDN-based throughput allocation in wireless networks for heterogeneous adaptive video streaming applications” 2017 13th International Wireless Communications and Mobile Computing Conference (IWCMC), 26-30 June 2017, DOI: https://doi.org/10.1109/IWCMC.2017.7986416
3	Miran Taha; Lorena Parra; Laura Garcia; Jaime Lloret, “An Intelligent handover process algorithm in 5G networks: The use case of mobile cameras for environmental surveillance” 2017 IEEE International Conference on Communications Workshops (ICC Workshops), Paris, France, 21-25 May 2017, DOI: https://doi.org/10.1109/ICCW.2017.7962763
4	Miran Taha, “A Novel CDN Testbed for Fast Deploying HTTP Adaptive Video Streaming” , MobiMedia '16 Proceedings of the 9th EAI International Conference on Mobile Multimedia Communications, Xi'an, China, June 18 - 20, 2016, PP. 65-71
5	Laura Garcia; Jaime Lloret; Carlos Turro; Miran Taha, “QoE assesment of MPEG-DASH in polimedia e-learning system.”, Conference: 2016 International Conference on Advances in Computing, Communications and Informatics (ICACCI),September 2016, India, DOI: https://doi.org/10.1109/ICACCI.2016.7732194

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