A Dynamic Adaptive HTTP Streaming Video Service for Google Android

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A Dynamic Adaptive HTTP Streaming Video Service for Google Android

Master of Science Thesis

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To dad.

Abstract

Adaptive streaming approaches over Hypertext Transfer Protocol (HTTP), such as Apple's HTTP Live streaming (HLS) and Microsoft's Live Smooth Streaming, have recently become very popular. This master's thesis project developed and evaluated several media rate adaptation algorithms optimized for mobile networks with a client running on Google's Android operating system. The deployed service supports HLS and the emerging ISO/IEC MPEG standard called Dynamic Adaptive Streaming over HTTP (MPEG-DASH).

Live media was the focus of the evaluation, since this content can not be cached in advance at the user's device, hence the quality of the user's experience will be affected by the currently available bandwidth which the user can utilize. Experiments were performed for multiple scenarios illustrating different network capabilities, especially various amounts of bandwidth available to the user.

This project has produced an implementation of HTTP-based adaptive streaming. This implementation follows the MPEG standard and enables robust and smooth playback of live video content via Google's Android devices. Results of the experiments have shown that the proposed adaptation mechanisms efficiently utilize the available bandwidth of the network. A clear conclusion of this thesis is that adaptive streaming will in fact enable substantial numbers of users to enjoy live media streaming to their devices.

Keywords: HTTP, live, video, streaming, MPEG-DASH, Android.

Sammanfattning

Adaptiv strömning metoder över Hypertext Transfer Protocol (HTTP), till exempel Apples HTTP Live streaming (HLS) och Microsoft Live Smooth Streaming, har nyligen blivit mycket populära. Detta examensarbete utvecklas och utvärderas flera medier algoritmer anpassning av överföringshastigheten optimerad för mobila nätverk med en klient som körs på Googles Android-operativsystem. Systemet kommer att överväga HLS och den framväxande ISO/IEC MPEG-standarden som kallas Dynamisk Adaptiv Strömmande över HTTP (MPEG-DASH).

Direktsendning media var i fokus för utvärderingen, eftersom detta innehåll inte kan cachas i förväg på användarens enhet, därmed kvaliteten på användarens upplevelse kommer att påverkas av den aktuella tillgängliga bandbredden som användaren kan utnyttja. Experimenten utfördes för flera scenarier illustrerar olika nätverksfunktioner, särskilt olika mängder bandbredden tillgänglig för användaren.

Detta projekt har producerat ett genomförande av HTTP-baserade adaptiva strömning. Denna implementering följer MPEG-standarden och möjliggör robusta och smidig uppspelning av direktsänd video innehåll via Googles Android-enheter. Resultat av experiment har visat att den föreslagna anpassningsmekanismer effectivt sätt utnyttja den tillgängliga bandbredden i nätverket. En tydlig slutsats i denna avhandling är att adatptive strömning faktiskt kommer att möjliggöra ett stort antal användare att njuta direktsänd medieströmning till sina enheter.

Keywords: HTTP, live, video, strömning, MPEG-DASH, Android.

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Finally, I would like to express my infinite gratitude to my mother and my brother for their outstanding support in Spain and abroad. And *Kathleen Streit*, with love.

Contents

List of Acronyms and Abbreviations

3GPP	3rd Generation Partnership Project
AAC	Advanced Audio Coding
AVC	Advanced Video Coding
BP	Baseline Profile
CBP	Constrained Baseline Profile
CDN	Content Delivery Network
CODEC	COmpressor-DECompressor
CPU	Central Processing Unit
DF	Delivery Format
DOM	Document Object Model
DSS	Darwin Streaming Server
DTS	Decoding Time-Stamp
DVM	Dalvik Virtual Machine
GOP	Group Of Pictures
GUI	Graphical User Interface
HDS	Adobe's HTTP Dynamic Streaming
HLS	Apple's HTTP Live Streaming
HTML	Hypertext Markup Language
HTTP	HyperText Transfer Protocol
IEC	International Electrotechnical Commission
IETF	Internet Engineering Task Force
IPTV	Internet Protocol Television
ISO	International Organization for Standardization
ITU	International Telecommunication Union
JNI	Java Native Interface
JVM	Java Virtual Machine
LGPL	Lesser General Public License
LSS	Microsoft's Live Smooth Streaming
MEGACO	Media Gateway Control Protocol
MF	Manifest File
MIME	Multipurpose Internet Mail Extensions
MMUSIC	Multiparty Multimedia Session Control (Working Group)
MS IIS	Microsoft Internet Information Services

MPD	Media Presentation Description
MPEG	Moving Picture Experts Group
MPEG-DASH	MPEG Dynamic Adaptive Streaming over HTTP
M2TS	MPEG-2 Transport Stream
MVC	Multiview Video Coding
NAT	Network Address Translation
NTP	Network Time Protocol
OHA	Open Headset Alliance
OS	Operating System
PCM	Pulse-Code Modulation
PIFF	Protected Interoperable File Format
PTS	Presentation Time-Stamp
QSS	QuickTime Streaming Server
RAP	Random Access Point
RTMP	Real Time Messaging Protocol
RTP	Real-time Transport Protocol
RTCP	RTP Control Protocol
RTSP	Real Time Streaming Protocol
SAX	Java's Simple API for XML
SCCP	Skinny Call Control Protocol
SDK	Software Development Kit
SIP	Session Initiation Protocol
SNTP	Simple Network Time Protocol
SVC	Scalable Video Coding
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
URI	Uniform Resource Identifier
URL	Universal Resource Locator
VCEG	Video Coding Experts Group
VLC	VideoLan Player
WAN	Wide Area Network
WAVE	Waveform Audio File Format
XML	Extensible Markup Language

List of Tables

List of Figures

List of Code Listings

List of Algorithms

Chapter 1

Introduction

"The important thing is the diversity available on the Web."
– Tim Berners-Lee

Today streaming is a very popular technology by which multimedia content is delivered continuously from a server to end-users[1]. Streaming methods are constantly being improved since the network capabilities and usage scenarios are quite heterogeneous. The creation of techniques which automatically provide the best possible quality to consumers has become a important challenge [48]. By means of the widely used Hypertext Transfer Protocol (HTTP) [31] which is the *de facto* protocol of today's Internet, new streaming approaches have been developed [48, 55, 59, 69].

Recent studies [24] have shown the crescent diversity of end-user devices. Mobile phones have become immensely popular in the recent years, since they have been significantly enhanced, providing Internet-based services over wireless and broadband connections. *Smartphones* offer capabilities similar to modern computers, as they run more sophisticated operating systems than regular cellphones (allowing the installation of third-party applications). Figure 1.1 illustrates predictions for the next several years in terms of network traffic, suggesting that there will be a considerable increase mobile traffic (estimated to represent the 26.6% of the total network traffic in 2015).

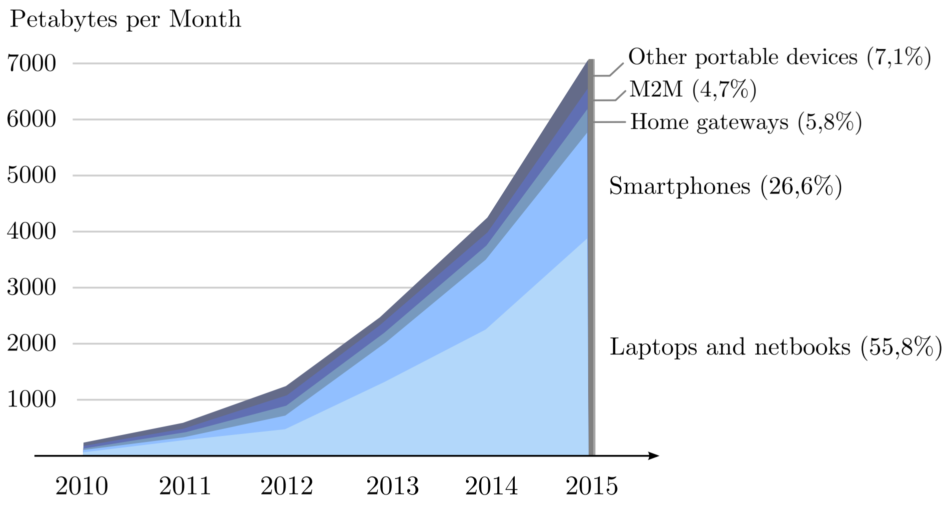


Figure 1.1: Network traffic expected for different devices. Laptops and smartphones lead traffic growth. Extracted from [24] (published in February 2011).

Video data has unequivocally become the predominant type of content transferred by mobile applications. As shown in figure 1.2, video traffic is expected to grow exponentially in the next several years, prevailing (66%) over web (20.9%) and peer-to-peer (6.1%) traffic.

[1] Other approaches [3] employ peer-to-peer overlay networks to provide multimedia content.

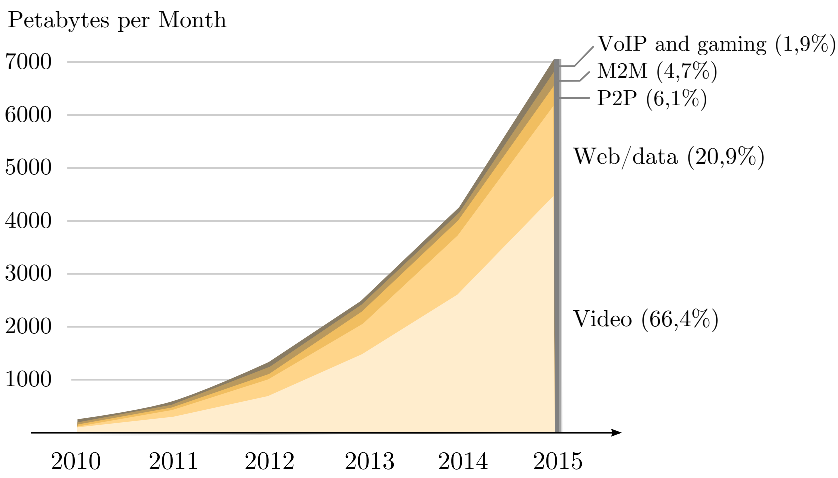


Figure 1.2: Estimate of the type of traffic to/from smartphone. Extracted from [24] (published in February 2011).

1.1 The problem and motivation

The immense variety of end-user devices operating under heterogeneous mobile networks leads to an interesting challenge: produce dynamic and automatized adaptation between producers and consumers, to deliver the best possible quality of content. Multiple constraints are present in the process of content delivery, such as network rate fluctuations or the client's own capabilities. For instance, end-users' devices can be limited by display resolution, maximum video bit-rate, or supported media formats. This master's thesis project focuses on portable devices and considers these limitations.

Figure 1.3 exemplifies the adaptation for similar clients which experience different limitations in the underlying communication network, hence the amount of data supplied per unit time to these clients differ. In this context, *adaptive streaming* [48] represents a family of techniques which addresses the problem of the difference in the data provided to different clients. By means of layered media content and adaptation mechanisms, end-users can perceive the most appropriate level of quality given their current constraints [23]. The most popular adaptive techniques will be introduced in the next chapter (section 2.3).

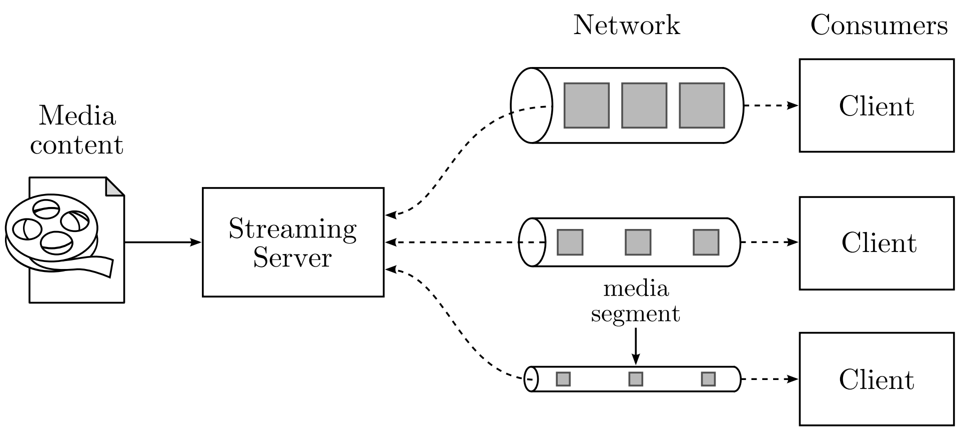


Figure 1.3: A simplified example of adaptive streaming offered to end-users.

In the particular case of live video streaming (that is, non-previously recorded media) there is still a need to evaluate different adaptive solutions under a variety of conditions in wireless networks. Reusing existing protocols to create Content Delivery Networks (CDN) [25] would provide enormous advantages to those wishing to offer live streaming services, as they could take advantage of the optimizations that have been made to efficiently support these protocols and the large investment in the existing infrastructures.

Multiple operating systems (OSs) have been developed for smartphones. Android (explained in detail in section 2.10) is an open-sourced code based mobile OS developed by Google (Android's official logos are depicted in figure 1.4). Recent statistics [33] have shown that Android is the predominant mobile operating system (36% worldwide) followed by Symbian (27.4%), Apple's iOS (16.8%), RIM (12.9%), and Windows Mobile (3.6%) (figure 1.5a). Furthermore, Android is expected to be deployed in almost the 50% of smartphones sold in 2012, followed by Apple's iOS (figure 1.5b).

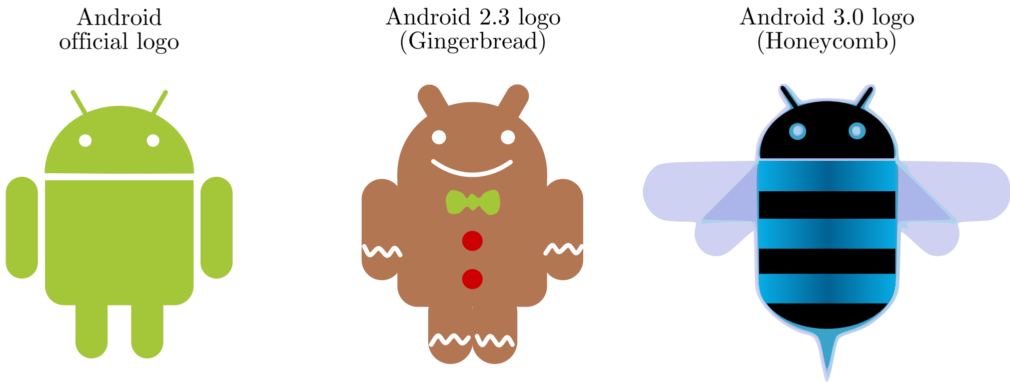
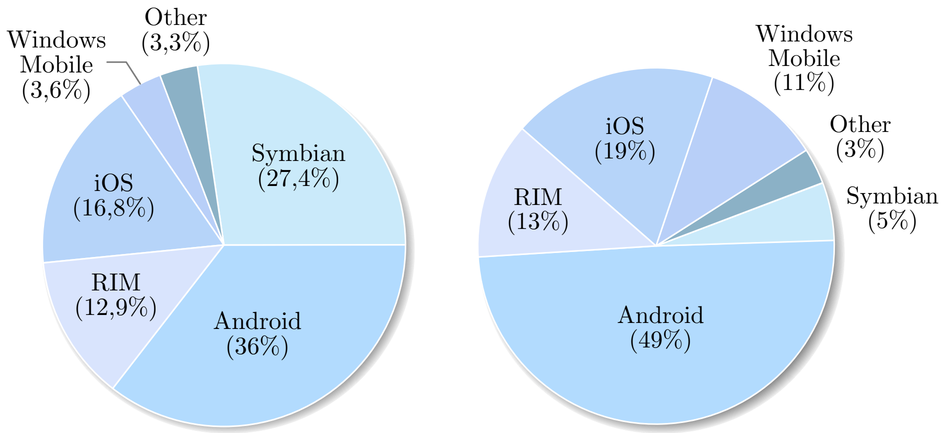


Figure 1.4: Android official logos.



(a) May 2011 **(b)** 2012 estimation

Figure 1.5: Worldwide smartphone sales to end users by Operating System in the first quarter of 2011. Extracted from [33] (May 2011).

At the moment there are few adaptive streaming services for Google's Android, despite Apple Inc. having published and implemented a protocol known as HTTP Live Streaming (HLS) [59] already supported in Apple's mobile phones (the well-known family of *iPhone* devices). Furthermore, Apple-HLS is in the process of becoming an Internet Engineering Task Force (IETF) standard. Other parties, such as the ISO/IEC Moving Picture Experts Group (MPEG), have proposed a standard (that is still in development) for adaptive streaming over HTTP, known as Dynamic Adaptive Streaming over HTTP (DASH) [69].

1.2 Goals

This master's thesis project is motivated by the following goals:

1. Proposal and evaluation of different adaptive mechanisms on the client's side which are able to converge to the maximum sustainable bit-rate. These mechanisms specify the client's application logic, providing efficient use of the available bit-rate in the network. Different procedures will be considered to optimize the use of available bandwidth and studying potential disadvantages.

2. Evaluation of system aspects in heterogeneous network scenarios where network bit-rate, packet delay, packet lost, video quality levels, segment durations, and other aspects differ. This leads to an analysis of potential benefits of the adaptive mechanisms, since they will diverge in terms of performance under different conditions.

In order to achieve these goals, the following tasks are defined in this project:

- Design of a service which supports the fundamental aspects of Apple-HLS and MPEG-DASH. The system produces different quality video levels and segments the media content into small *segments* which are offered to clients.

- Implementation of a prototype client as an application for Google's Android operating system. Achieving this goal requires that we identify what development resources are available for Android and select which of these resources we will use. This leads to an extensive study of the capabilities of the Android's native media framework, *Stagefright*, focusing on live video content. In particular, to analyze which media formats and streaming protocols are natively supported by Stagefright.

- Definition of multiple metrics to analyze during the evaluation. The evaluation will include efficiency, performance, delays, and bandwidth utilization. These metrics have to be defined conveniently in order to efficiently compare the adaptive mechanisms.

1.3 Scope

This project intends to evaluate the performance of different adaptive mechanisms under single end-user scenarios. Therefore, the scalability of the system (i.e., multiple users requesting media content) is not covered by this master's thesis project.

The network communication in this project is based on HTTP and uses TCP as the transport protocol, since it provides reliable byte stream delivery and congestion avoidance mechanisms. The advantages or disadvantages of using other transport protocols are not considered in this work.

1.4 Audience

Software engineers and Android developers interested in adaptive media content delivery could benefit from this master's thesis project. In this work, the most recent adaptive streaming standards (using HTTP as a delivery protocol) have been considered.

1.5 Organization of the thesis

Chapter 2 presents the relevant background, introducing different streaming techniques and the adaptive protocols which have been recently published, such as Apple's HTTP Live Streaming, Microsoft's Live Smooth Streaming, Adobe's HTTP Dynamic Streaming, and MPEG Dynamic

Adaptive Streaming over HTTP. In addition, the capabilities of the Android operating system are introduced, focusing on media formats, coders/decoders (CODECs), and adaptive protocols which are supported in Stagefright.

Chapter 3 summarizes the previous work which has been done in the area of the adaptive streaming, including simulations under heterogeneous network restrictions, performance of the different adaptive protocols, and proposals of adaptation mechanisms.

Chapter 4 explains on detail the proposed system architecture which has been designed and implemented during this master's thesis project.

Chapter 5 covers the overall evaluation performed for the system architecture explained in chapter 4. This chapter includes the definition of the metrics utilized, the input and output parameters, and the results achieved.

Chapter 6 presents a discussion of the results achieved in chapter 5 and the conclusions. Finally, the limitations of this master's thesis project are considered and presented as future work.

Chapter 2

Background

"Any sufficiently advanced technology is indistinguishable from magic."
– Arthur C. Clarke

There are three main methods to deliver multimedia: *traditional streaming* (section 2.1), *progressive download* (section 2.2), and *adaptive streaming* (section 2.3). Section 2.4 describes the evolution of adaptive streaming, using HTTP as a delivery protocol. The most popular implementations of this technique are explained in detail in the following subsections: Apple's HTTP Live Streaming in section 2.4.2, Microsoft's Live Smooth Streaming in section 2.4.3, Adobe's HTTP Dynamic Streaming in section 2.4.4, and MPEG Dynamic Adaptive Streaming over HTTP in section 2.4.5. Two different types of services can be provided: video on-demand or live streaming (section 2.9).

The most relevant video and audio CODECs are described in sections 2.5 and 2.6 respectively, whereas container formats are described in section 2.7. Android operating system capabilities are explained in section 2.10, mainly focusing on the media framework and supported CODECs.

Finally, a brief comparison of the different streaming approaches is presented at the end of the chapter (section 2.11).

2.1 Traditional streaming

Traditional streaming [48, p. 113-117] requires a stateful protocol which establishes a session between the service provider and client. In this technique, media is sent as a series of packets. The Real-Time Transport Protocol (RTP) together with the Real-Time Streaming Protocol (RTSP) are frequently used to implement such service.

2.1.1 Real-Time Transport Protocol (RTP)

The Real-Time Transport Protocol (RTP) [65] describes a packetization scheme for delivering video and audio streams over IP networks. It was developed by the audio-video transport working group of the IETF in 1996.

RTP is an end-to-end, real-time protocol for unicast or multicast network services. Because RTP operates over UDP it is suitable for multicast distribution, while all protocols that are built on top of TCP can only be unicast. For this reason RTP is widely used for distributing media in the case of Internet Protocol Television (IPTV), as the Internet service provider can control the amount of multicast traffic that they allow in their network and they gain quite a lot from the scaling which multicast offers. For a streaming multimedia service RTP is usually used in conjunction with RTSP, with the audio and video transmitted as separate RTP streams.

The RTP specification describes two sub-protocols which are the data transfer protocol (RTP) and the RTP Control Protocol (RTCP) [65, section 6]:

1. RTP is used for transferring multimedia data utilizing different CODECs along with time-stamps and sequence numbers. These time-stamps and sequence numbers allow the

receiver to detect packet loss and perform reordering when necessary and synchronize media streams, among other operations.

2. RTCP specifies the control information for synchronization and quality of service parameters that may be sent. This protocol should use a maximum 5% of the overall bandwidth.

Optionally RTP can be used with a session description protocol or a signalling protocol such as H.323, the Media Gateway Control Protocol (MEGACO), the Skinny Call Control Protocol (SCCP), or the Session Initiation Protocol (SIP).

RTP neither provides a mechanism to ensure timely delivery nor guarantees quality of service or in-order delivery. Additionally, there is no flow control provided by the protocol itself, rather flow control and congestion avoidance are up to the application to implement.

2.1.2 Real-Time Streaming Protocol (RTSP)

The Real-Time Streaming Protocol (RTSP) [66] is a session control protocol which provides an extensible framework to control delivery of real-time data. It was developed by the multiparty multimedia session control working group (MMUSIC) of the IETF in 1998. RTSP is useful for establishing and controlling media sessions between end points, but it is not responsible for the transmission of media data. Instead, RTSP relies on RTP-based delivery mechanisms. In contrast with HTTP[1], RTSP is stateful and both client and server can issue requests. These requests can be performed in three different ways: (1) persistent connections used for several request/response transactions, (2) one connection per request/response transaction or (3) no connection.

Some popular RTSP implementations are Apple's QuickTime Streaming Server (QSS) (also its open-sourced version, Apple's Darwin Streaming Server (DSS)) and RealNetworks' Helix Universal Server.

2.2 Progressive download

Progressive download is a technique to transfer data between server and client which has become very popular and it is widely used on the Internet. Progressive download typically can be realized using a regular HTTP server. Users request multimedia content which is downloaded progressively into a local buffer. As soon as there is sufficient data the media starts to play. If the playback rate exceeds the download rate, then playback is delayed until more data is downloaded.

Progressive download has some disadvantages: (1) wasteful of bandwidth if the user decides to stop watching the video content, since data has been transferred and buffered that will not be played, (2) no bit-rate adaptation, since every client is considered equal in terms of available bandwidth and, (3) no support for live media sources.

2.3 Adaptive streaming

Adaptive streaming [48, p. 141-155] is a technique which detects the user's available bandwidth and CPU capacity in order to adjust the quality of the video that is provided to the user, so as to offer the best quality that can be given to this user in their current circumstance. It requires an encoder to provide video at multiple bit rates (or that multiple encoders be used) and can be deployed within a CDN to provide improved scalability. As a result, users experience streaming media delivery with the highest possible quality.

[1] Actually, cookies can be used to make HTTP stateful [12]. In addition, HTTP 1.1 can use persistent connections as a performance improvement [31, section 8.1].

Techniques to adapt the video source's bit-rate to variable bandwidth can be classified into three categories: transcoding (section 2.3.1), scalable encoding (section 2.3.2), and stream switching (section 2.3.3).

2.3.1 Transcoding

By means of transcoding it is possible to convert raw video content on-the-fly on the server's side. To match a specific bit-rate we transcode from one encoding to another. A block diagram of this technique is depicted in figure 2.1. The main advantage of this approach is the fine granularity that can be obtained, since streams can be transcoded to the user's available bandwidth.

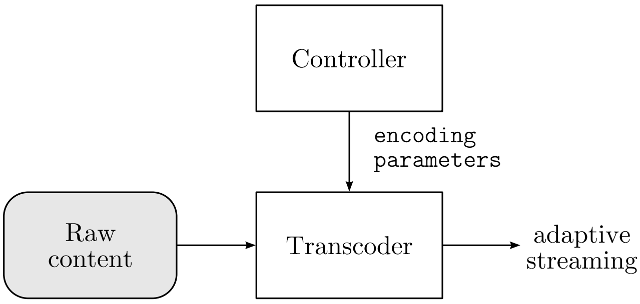


Figure 2.1: Transcoding approach for adaptive streaming. Adapted from [23].

However, there are some serious disadvantages that are worth pointing out. First of all, the high cost of transcoding, which requires adapting the raw video content several times for several requests for different quality. As a result scalability decreases since transcoding needs to be performed for every different client available bandwidth. Due to the computational requirements of a real-time transcoding system, the encoding process is required to be performed in appropriate servers, in order to be deployed in CDNs.

2.3.2 Scalable encoding

Using a scalable CODEC standard such as H.264/MPEG-4 AVC (described in detail in section 2.5.4), the picture resolution and the frame rate can be adapted without having to re-encode the raw video content [42]. This approach tends to reduce processing load, but it is clearly limited to a set of scalable CODEC formats. A block diagram of this technique is depicted in figure 2.2.

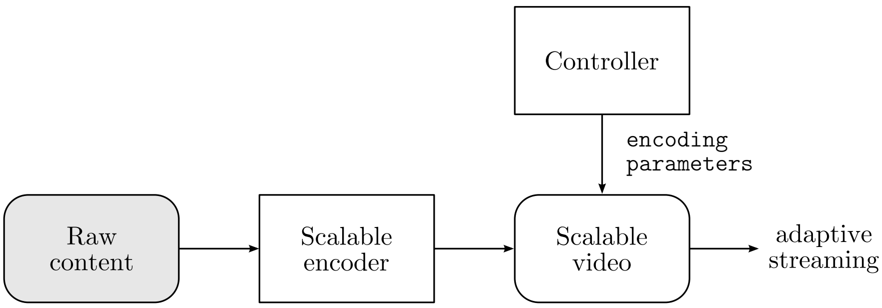


Figure 2.2: Scalable encoding approach for adaptive streaming. Adapted from [23].

Nevertheless, deployment into CDNs is complicated in this approach because specialized servers are required to implement the adaptation logic [23].

2.3.3 Stream switching

The stream switching approach encodes the raw video content at several different increasing bit-rates, generating R versions of the same content, known as video levels. As shown in Figure 2.3, 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 continuous playback.

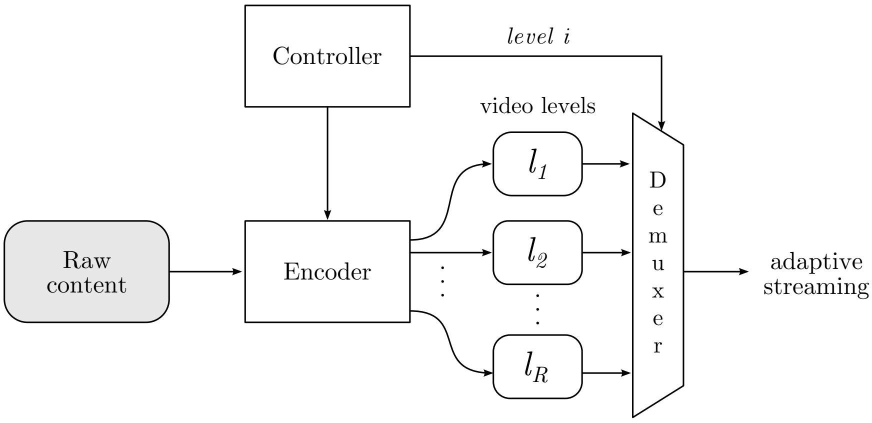


Figure 2.3: Stream switching approach for adaptive streaming. Adapted from [23].

The main purpose of this method is to minimize processing costs, since no further processing is needed once all video levels are generated. In addition, this approach does not require a specific CODEC format to be implemented, that is, it is completely CODEC agnostic. In contrast, storage and transmission requirements must be considered because the same video content is encoded R times (but at different bit-rates). Note that the quality levels are not incremental, therefore only one substream has to be requested. The only disadvantage of this approach is the coarse granularity since there is only a discrete set of levels. Additionally, if there are no clients for a given rate there is no need to generate this level; however, this only costs storage space at the server(s) and not all servers need to store all levels of a stream.

Figure 2.4 illustrates the stream switching approach over time, assuming that all segments have the same duration and the switching operations are performed after each segment has been played (not partially). Segments at different video qualities are requested to be played in a sequence. The number of levels and the duration of the segments are flexible and become part of the system's design choices.

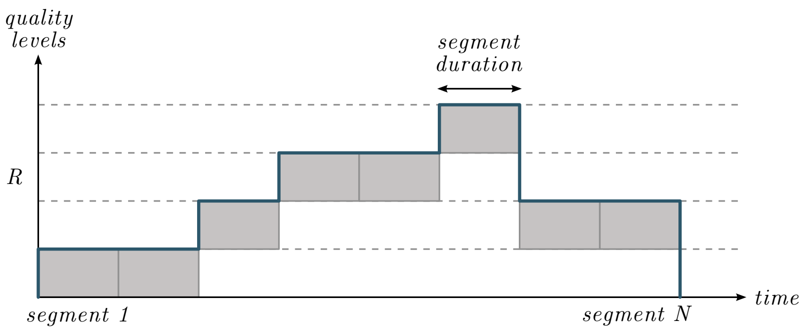


Figure 2.4: Stream switching example over time.

2.4 HTTP-based adaptive streaming

Recently a new solution for adaptive streaming has been designed, based on the stream switching technique (explained in section 2.3.3). It is an hybrid method which uses HTTP as a delivery protocol instead of defining a new protocol.

Video and audio sources are cut into short segments of the same length (typically several seconds). Optionally, segments can be cut along a video Group of Pictures (explained in section 2.5.1), thus every segment starts with a key frame, meaning that segments do not have past/future dependencies among them. Finally, all segments are encoded in the desired format and hosted on a HTTP server.

Clients request segments sequentially and download them using HTTP progressive download. Segments are played in order and since they are contiguous, the resulting overall playback is smooth. All adaptation logic is controlled by the client. This means that the client calculates the fetching time of each segment in order to switch-up or switch-down the bit-rate. A basic example is depicted in figure 2.5, where the *feedback controller* represents the switching logic applied on the client side. Thicker arrows correspond to transmission of an actual data segment.

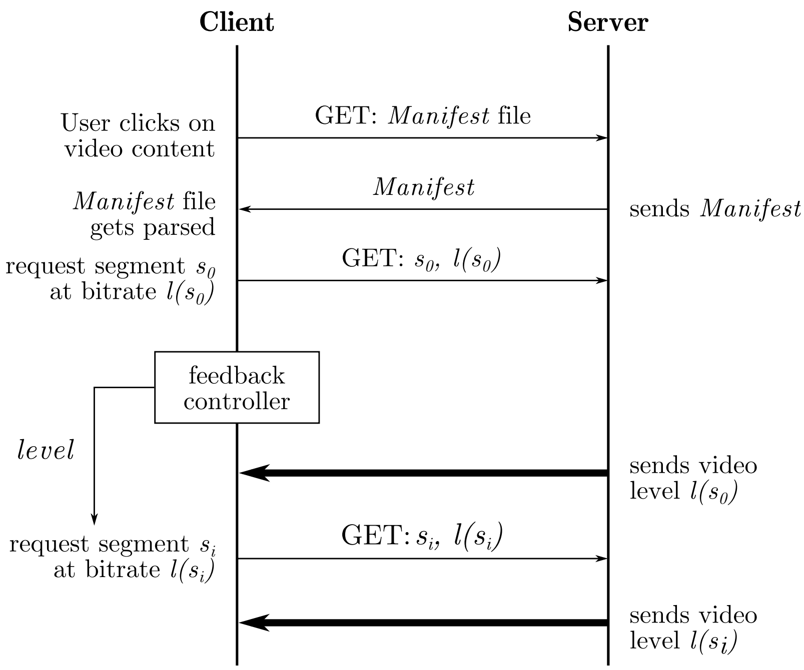


Figure 2.5: Client contract adaptation. Network delays are omitted for simplicity.

2.4.1 Why HTTP?

HTTP is widely used in the Internet as a delivery protocol. Because HTTP is so widely used HTTP-based services avoid NAT and firewall issues. Because (1) the client initiated the TCP connection from behind the firewall or Network Address Translation (NAT) or (2) because holes for HTTP have been purposely opened through the firewall or NAT service. The NAT or firewall will allow the packets from the HTTP server to be delivered to the client over a TCP connection or SCTP association (for the rest of this thesis we will assume that TCP is used as the transport protocol for HTTP). Additionally because HTTP uses TCP it automatically gets in order reliable byte stream delivery and TCP provides extensive congestion avoidance mechanisms. HTTP-based services can use the existing HTTP servers and CDN infrastructures.

Finally, the *streaming* session is controlled entirely by the client, thus there is no need for negotiation with the HTTP server, as clients simply open TCP connections and choose an initial content bit-rate. Then clients switch among the offered streams depending on their available bandwidth.

2.4.2 Apple's HTTP Live Streaming (HLS)

In May 2009 Apple released a HTTP-based streaming media communication protocol (Apple-HLS) [10, 11, 29, 52, 59] to transmit bounded and unbounded streams of multimedia data. Apple-HLS is based on the Emblaze Network Media Streaming technology which was released in 1998. According to this specification, an overall stream is broken into a sequence of small HTTP-based file downloads, where users can select alternate streams encoded at different data rates. Because the HTTP clients request the files for downloading this method works through firewalls and proxy servers (unlike UDP-based protocols such as RTP which require ports to be opened in the firewall or require use of an application layer gateway).

Initially, users download an extended M3U playlist which contains several Uniform Resource Identifiers (URIs) [14] corresponding to media files, where each file must be a continuation of the encoded stream (unless it is the first one or there is a discontinuity tag which means that the overall stream is unbounded). Each individual media file must be formatted as an MPEG-2 transport stream [43] or a MPEG-2 audio elementary stream.

Listing 2.1 illustrates a simple example of an extended M3U playlist where the entire stream consists of three 10-seconds-long media files. Listing 2.2 provides a more complicated example, where there are different available bandwidths and each entry points to an extended M3U sub-playlist file (depicted in figure 2.6 on page 13).

Listing 2.1: Example of an extended M3U playlist which contains three 10-seconds-long media segments.

```
#EXTM3U
#EXT-X-MEDIA-SEQUENCE:0
#EXT-X-TARGETDURATION:10
#EXTINF:10,
http://www.example.com/segment1.ts
#EXTINF:10,
http://www.example.com/segment2.ts
#EXTINF:10,
http://www.example.com/segment3.ts
#EXT-X-ENDLIST
```

Listing 2.2: Example of an extended M3U playlist which contains several sub-playlists, consequently providing alternate stream at different qualities.

```
#EXTM3U
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=1280000
http://www.example.com/low.m3u8
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=2560000
http://www.example.com/mid.m3u8
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=7680000
http://www.example.com/hi.m3u8
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=65000,CODECS="mp4a.40.5"
http://www.example.com/audio-only.m3u8
```

The overall process performed in an Apple-HLS architecture is shown in figure 2.7 on page 13. From the server's side, the protocol operates as follows: (1) the media content is encoded at different bit-rates to produce streams which present the same content and duration (but with different quality), (2) each stream is divided into individual files (segments) with approximately equal duration, (3) a playlist file is created which contains an URI for each media file indicating its duration (the playlist can be accesed through an URL), and (4) further changes to the playlist file must be performed atomically.

From the client's side, the following actions take place: (1) selection of the media file which shall be played must be made and (2) periodically reload the playlist file (unless it is bounded). It is necessary to wait a period of time before attempting to reload the playlist. The initial amount of time to wait before re-loading the playlist is set as the duration of the last media file in the playlist. If the client reloads the playlist file and the playlist has not changed, then the client waits a period of time proportional to the duration of the segments before retrying: 0.5 times the duration for the first attempt, 1.5 times the duration for the second and 3.0 times the duration in further attempts.

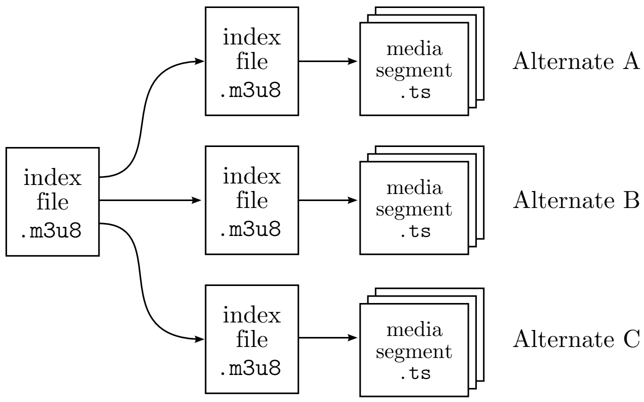


Figure 2.6: Alternate index files to offer different streams. Adapted from [11].

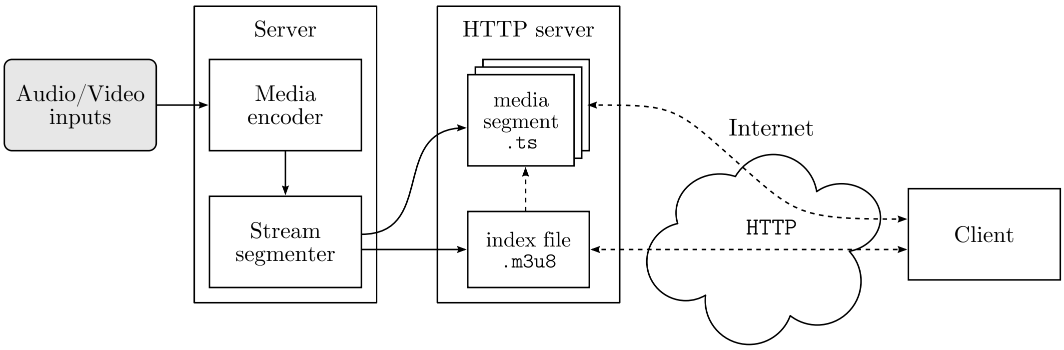


Figure 2.7: HTTP Live streaming architecture. Adapted from [11].

2.4.3 Microsoft's Live Smooth Streaming (LSS)

In 2009, Microsoft Corporation released its approach [53, 55, 74] for adaptive streaming over HTTP. Microsoft's Live Smooth Streaming (LSS[2]) format specification is based on the ISO Base Media File Format and standardized as the Protected Interoperable File Format (PIFF) [19], whereas the manifest file is based on the Extensible Markup Language (XML) [18] (a simplified example is shown in listing 2.3).

Microsoft provides a Smooth Streaming demo[3] which requires the Silverlight plug-in [54] to be installed. In this online application, the available bandwidth can be easily adjusted within a very simple user interface. A network usage graph is dynamically displayed as well as the adapted video output.

[2] Although Microsoft has not adopted an official acronym for Live Smooth Streaming, it will be referred as LSS in this master's thesis.

[3] Experience Smooth Streaming. http://www.iis.net/media/experiencesmoothstreaming.

Listing 2.3: Microsoft-LSS manifest sample.

```xml
<?xml version="1.0" encoding="UTF-8"?>
<SmoothStreamingMedia MajorVersion="2" MinorVersion="0" Duration="2300000000"
   TimeScale="10000000">
   <Protection>
      <ProtectionHeader SystemID="{9A04F079-9840-4286-AB92E65BE0885F95}">
         <!-- Base 64 Encoded data omitted for clarity -->
      </ProtectionHeader>
   </Protection>
   <StreamIndex Type = "video" Chunks = "115" QualityLevels = "2" MaxWidth = "720"
      MaxHeight = "480" TimeScale="10000000" Name="video"
      Url ="QualityLevels({bitrate},{CustomAttributes})/Fragments(video={start_time})">
      <QualityLevel Index="0" Bitrate="1536000" FourCC="WVC1"
         MaxWidth="720" MaxHeight="480" CodecPrivateData = "...">
         <CustomAttributes>
            <Attribute Name="Compatibility" Value="Desktop" />
         </CustomAttributes>
      </QualityLevel>
      <QualityLevel Index="5" Bitrate="307200" FourCC="WVC1"
         MaxWidth="720" MaxHeight="480" CodecPrivateData="...">
         <CustomAttributes>
            <Attribute Name="Compatibility" Value="Handheld" />
         </CustomAttributes>
      </QualityLevel>
      <c t ="0" d="19680000" />
      <c n ="1" t="19680000" d="8980000" />
   </StreamIndex>
</SmoothStreamingMedia>
```

2.4.4 Adobe's HTTP Dynamic Streaming

Adobe's HTTP dynamic streaming (HDS) approach enables on-demand and live streaming and it supports HTTP and Real Time Messaging Protocol (RTMP) [4]. It uses different format specifications for media files (Flash Video or F4V, based on the standard MPEG-4 Part 12) and manifests (Flash Media Manifest or F4M). In order to deploy Adobe's solution it is necessary to set up a Flash Media Streaming Server [37] which is a proprietary and commercial product. Additionally, users need to install Adobe's Flash Player.

2.4.5 MPEG Dynamic Adaptive Streaming over HTTP (MPEG-DASH)

MPEG Dynamic Adaptive Streaming over HTTP (MPEG-DASH) is a protocol presented by a joint working group [69] of Third Generation Partnership Project (3GPP) and MPEG. This protocol has recently been considered to become an ISO standard [1, 2]. MPEG-DASH defines a structure similar to Microsoft-LSS for adaptive streaming supporting on-demand, live, and time-shifting[4] viewing, but it proposes changes in the file formats, defining a XML-based **manifest** file.

MPEG-DASH introduced the concept of *media presentation*. A media presentation is a collection of structured video/audio content:

- A media presentation consists of a sequence of one or more *periods* which are consecutive and do not overlap.
- Each period consists of one or more *representations* of the same media content. Periods have an assigned start time which is relative to start of the media presentation.

[4] Time-shifting involves recording content to a storage medium to be watched at a later time that is more suitable for the user.

- Each representation[5] specifies a video quality *profile* consisting of several parameters such as bandwidth, encoding, and resolution. Representations contain one or more segments, represented by Universal Resource Locators (URLs).

- Segments contain fragments of the actual video content.

A Media Presentation Description (MPD) schema is an XML-based file which contains the whole structure of a media presentation introduced above. A simplified version is depicted in figure 2.8, and listing 2.4 provides a concrete example.

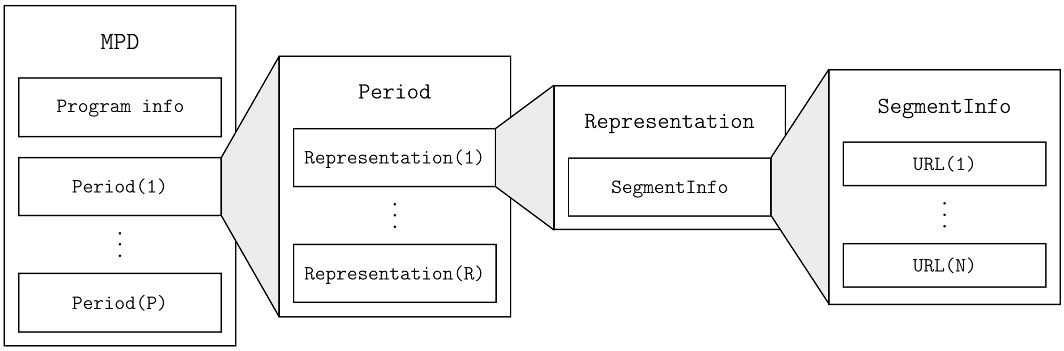


Figure 2.8: MPD simplified structure. Adapted from [69, figure 4].

Listing 2.4: MPD example. Optional elements and attributes are omitted for simplicity.

```xml
<?xml version="1.0" encoding="UTF-8"?>
<MPD minBufferTime="PT10S">
   <Period start="PT0S">
      <Representation mimeType="video/3gpp; codecs=263, samr" bandwidth="256000" id="256">
         <SegmentInfo duration="PT10S" baseURL="rep1/">
            <InitialisationSegmentURL sourceURL="seg-init.3gp"/>
            <Url sourceURL="seg-1.3gp"/>
            <Url sourceURL="seg-2.3gp"/>
            <Url sourceURL="seg-3.3gp"/>
         </SegmentInfo>
      </Representation>
      <Representation mimeType="video/3gpp; codecs=mp4v.20.9" bandwidth="128000" id="128">
         <SegmentInfo duration="PT10S" baseURL="rep2/">
            <InitialisationSegmentURL sourceURL="seg-init.3gp"/>
            <Url sourceURL="seg-1.3gp"/>
            <Url sourceURL="seg-2.3gp"/>
            <Url sourceURL="seg-3.3gp"/>
         </SegmentInfo>
      </Representation>
   </Period>
</MPD>
```

The MPEG-DASH protocol specifies the syntax and semantics of the MPD, the format of segments, and the delivery protocol (HTTP). Fortunately, it permits flexible configurations to implement different types of streaming services. The following parameters can be selected flexibly: (1) the size and duration of the segments (these can be selected individually for each representation), (2) the number of representations and (3) the profile of each representation (bit-rate, CODECs, container format, etc).

Regarding the client's behaviour, it can flexibly: (1) decide when and how to download segments, (2) select appropriate representation, (3) switch representations and, (4) select the

[5]Note that representation is a synonym of *quality level* or *bit-rate level* in the context of this master's thesis project. The three terms will be used indistinctly.

transport of the MPD file, which could also be retrieved by other means, rather than only through HTTP.

Figure 2.9 on page 16 exemplifies the communication between server and client in a MPEG-DASH streaming service. First the client retrieves the MPD file and afterwards it sequentially requests the media segments. In every period a representation level is selected, based on the fetching times and other parameters determined by the client.

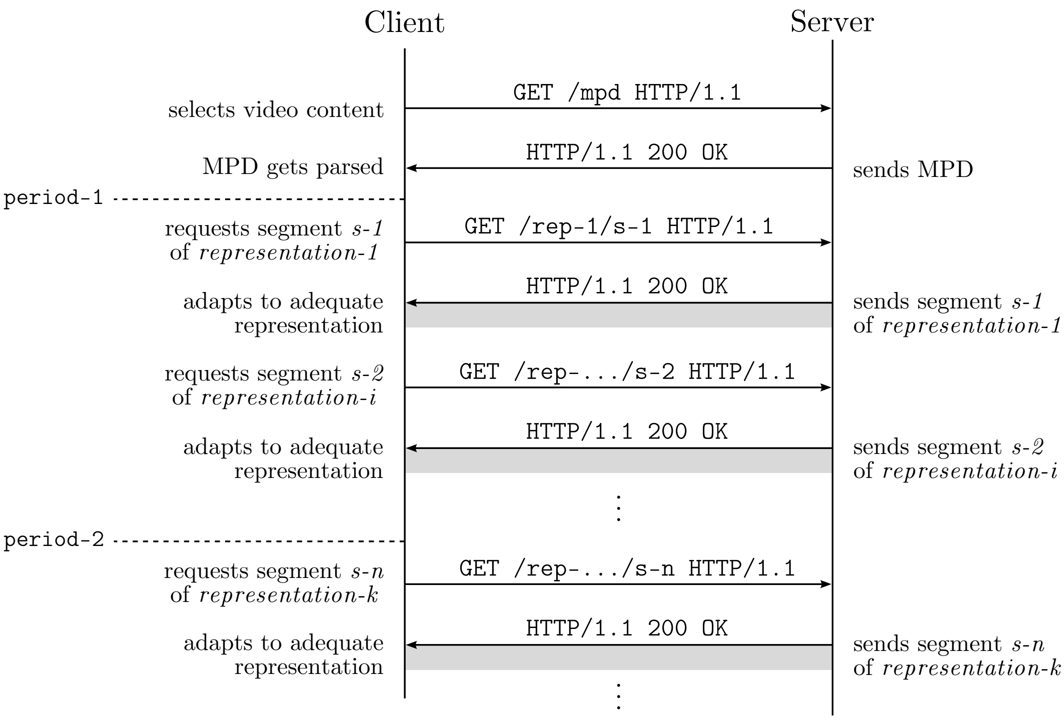


Figure 2.9: Client contract adaptation example in MPEG-DASH. Network delays are omitted for simplicity.

2.5 Video CODECs

This section describes a number of aspects of video coders and decoders that are relevant to a reader of this thesis.

2.5.1 Video frames

Compressed video standards only encode full frame data for certain frames, known as *key frames*, *intra-frames* or simply *I-frames*. The frames which follow a key frame, *predicted* frames or *P-frames*, are encoded considering only the differences with the preceding frame, resulting in less data being needed to encode these subsequent frames. Videos whose frame information changes rapidly require more key frames than a slowly changing visual scene. An example of the relationship between several frames is shown in figure 2.10.

Bidirectional encoding is also possible by means of Bi-predictive frames (*B-frames*). B-frames consider both previous and subsequent frame differences to achieve better compression.

A Group of Pictures (GOP) consists of one I-frame followed by several P-frames and optionally, B-frames. Lowering the GOP size (key frame interval) can provide benefit: using more frequent key frames helps to reduce distortion when streaming in a lossy environment. However, a low GOP size increases the media file size since key frames contain more bits than predictive frames.

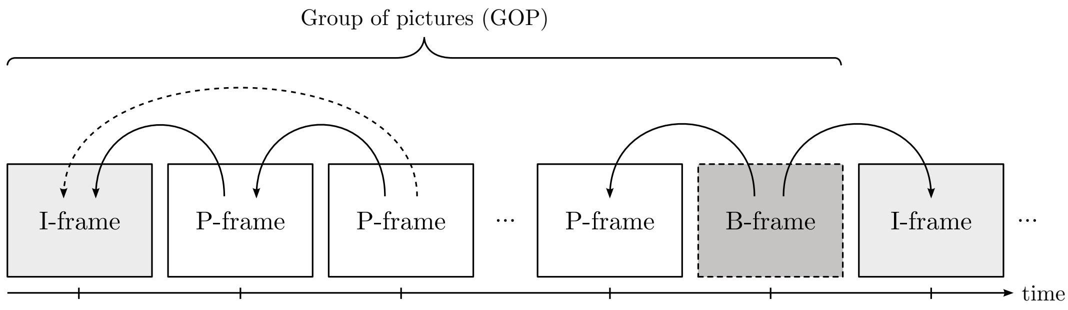


Figure 2.10: Distribution of frames in a video stream. Every group of pictures is constituted by one I-frame and several P-frames, and optionally B-frames.

2.5.2 Decoding and presentation time-stamps

Decoding Time-stamp (DTS) is used to synchronize streams and control the rate at which frames are decoded. It is not essential to include a DTS in all frames, since it can be interpolated by the decoder. In contrast, the Presentation Time-stamp (PTS) indicates the exact moment when a video frame has to be presented at the decoder's output. PTS and DTS only differ when bidirectional coding is used (i.e., when B-frames are used).

2.5.3 H.263

H.263 [44] is a low-bit-rate video compression standard designed for videoconferencing, although is widely used in many other applications. It was developed by the ITU-T Video Coding Experts Group (VCEG) in 1996. H.263 has been supported in Flash video applications and widely used by Internet on-demand services such as YouTube or Vimeo.

H.263 bit-rates range from 24 kb/s to 64 kb/s. Video can be encoded and decoded to this format with the free LGPL-licensed `libavcodec` library (part of the FFmpeg project [13]).

2.5.4 H.264/MPEG-4 AVC

H.264/MPEG-4 Part 10 [42] or Advanced Video Coding (AVC) is the successor of H.263 and other standards such as MPEG-2 and MPEG-4 Part 2. H.264 is one of the most commonly used formats for recording, compression, and distribution of high definition video. H.264 is one of the CODECs supported for Blu-ray discs. H.264 was developed by the ITU-T Video Coding Experts Group together with ISO/IEC MPEG in 2003. It is supported in Adobe's Flash Player and Microsoft's Silverlight. Therefore, multiple streaming Internet sources such as Vimeo, YouTube, and the Apple iTunes Store follow the H.264 standard.

H.264 specifies seventeen *profiles* which are oriented to multiple types of applications. The Constrained Baseline Profile (CBP) is the most basic one, followed by the Baseline Profile (BP) and the Main Profile (MP) in increasing order of complexity. CBP and BP are broadly used in mobile applications and videoconferencing. Additionally, these are the only H.264 profiles supported by Android's native media framework. Table 2.1 summarizes the major differences among these three profiles.

One of the most recent features added to the H.264 standard is Scalable Video Coding (SVC) [42, Annex G]. SVC enables the construction of bitstreams which contain sub-bitstreams, all conforming the standard. In addition, Multiview Video Coding (MVC) [42, Annex H] offers an even more complex composition of bitstreams, allowing more than one view point for a video scene[6].

[6] Mainly used for stereoscopic 3D video encoding.

Table 2.1: Major differences among H.264 Constrained Baseline Profile (CBP), Baseline Profile (BP) and Main Profile (MP).

Feature	CBP	BP	MP
Android support	Yes	Yes	No
Flexible macro-block ordering (FMO)	No	Yes	No
Arbitrary slice ordering (ASO)	No	Yes	No
Redundant slices (RS)	No	Yes	No
B-frames	No	No	Yes
CABAC entropy coding	No	No	Yes

2.5.5 VP8

VP8 is a video compression format originally created by On2, but eventually released by Google in 2010 after they purchased On2. VP8 was published with a BSD-license, therefore it is considered to be an open alternative to H.264.

VP8 encoding and decoding can be performed by the `libvpx` library [70]. Moreover, the FFmpeg team released a `ffvp8` decoder on July, 2010.

2.6 Audio CODECs

This section describes a number of aspects of audio coders and decoders that are relevant to a reader of this thesis.

2.6.1 MP3

MP3 [17, 36, 39] (published as MPEG-1 and MPEG-2 Audio Layer III) has undoubtedly become in the last decade the *de facto* audio CODEC due to its use in multiple media services and digital audio players. MP3 is a patented digital audio encoding format which reduces the amount of data required since it discards the less audible components to human hearing, i.e., it implements a lossy compression algorithm.

2.6.2 Advanced Audio Coding (AAC)

Advanced Audio Coding (AAC) [17] is an ISO/IEC standardized audio compression format which provides lossy compression encoding. It is supported in a extensive variety of devices. AAC is part of the MPEG-2 [40] and MPEG-4 [41] specifications. AAC was designed to be the successor of the MP3 format. A later extension defines the High-Efficiency Advanced Audio Coding (HE-AAC).

Three default profiles are defined [40]: Low Complexity (LC), Main Profile (MP), and Scalable Sample Rate (SSR). In conjunction with the Perceptual Noise Substitution and 45 Audio Object Types [41], new profiles are defined, such as the High Efficiency AAC Profile (HE-AAC and HE-AAC v2) and the Scalable Audio Profile. The latter utilizes Long Term Prediction (LTP).

2.6.3 Vorbis

Vorbis [72] is a free and open audio CODEC meant to replace patented and restricted formats such as MP3 (section 2.6.1). Vorbis provides a lossy compression encoding over a wide range of bit-rates. It has been shown to perform similar to MP3 [22].

2.7 Container formats

A container is a meta-format which wraps any kind of media data, resulting in a single file. Containers are used to interleave different data types, for instance video streams, subtitles, and even meta-data information. A vast variety of container formats has been developed, presenting different features and limitations. The most important multimedia containers are briefly introduced below.

- **MP4** (`.mp4`) is a popular container format defined in the MPEG-4 Part 14 standard. It supports almost any kind of media data. Typically an MP4 container contains video and audio streams encoded with H.264 and AAC, respectively.
- **3GP** (`.3gp`) [2] is widely utilized on 3G mobile phones. 3GP is defined as an extension of MPEG-4 Part 12. 3GP normally stores video streams encoded with either MPEG-4 Part 2, H.263, or H.264 and audio streams with either AAC (LC profile) or HE-AAC.
- **MPEG Transport Stream** (`.ts`) is defined in the MPEG-2 Part 1 standard and generally used in digital television broadcast systems.
- **Ogg** (`.ogg`) is an open container format developed by the Xiph.Org Foundation. Ogg generally combines the Theora and Vorbis CODECs.
- **WebM** [70] (`.webm`) is a recently released open, royalty-free container format based on the Matroska container. WebM has gained noteworthy popularity, since it has been adopted as one of the most suitable formats for the Web content (although HTML 5, the predominant markup language for web pages, is defined as being CODEC-agnostic [38, section 4.8.6]) due to its patent-free and open nature. Consequently, only open-source CODECs are recommended: video streams are encoded with VP8 and audio streams with Vorbis (these introduced in section 2.5.5 and section 2.6.3, respectively).

Table 2.2 provides a comparison between the container formats explained above, in terms of supported audio and video CODECs.

Table 2.2: Video and audio CODECs supported in several container formats (2011, August). Information collected from [2, 43, 70, 72].

Format	H.263	H.264	MPEG-4	VP8	MP3	AAC	HE-AAC	Vorbis
3GP	Yes	Yes	Yes	No	No	Yes	Yes	No
MP4	Yes	Yes	Yes	No	Yes	Yes	Yes	Yes
MPEG-TS	No	Yes	Yes	No	Yes	Yes	Yes	No
Ogg	No	No	No	No	No	No	No	Yes
WebM	No	No	No	Yes	No	No	No	Yes

2.8 Quality video levels

Video and audio content can be offered at multiple representations (*quality levels*) to adequate to different types of end-users. It is a well-known fact that end-users are affected by a wide variety of restrictions in terms of network capabilities, screen resolutions, and media formats supported, among other limitations. The more representations provided on the server's side, the better *granularity* characterizes the system, since a wider variety of alternative versions of media content is served. Nevertheless, the creation of multiple quality levels incurs a higher cost in terms of processing time, storage requirements, and CPU consumption. The following encoding parameters are especially relevant when defining a representation level:

Video bit-rate (kb/s) rate of information in the video stream.

Frame rate (fps) frequency of *frame* presentation, measured as frames per second.

Audio bit-rate (kb/s) rate of information in the audio stream.

Audio channels stereo (2) or mono (1).

Sampling rate (Hz) number of samples per second taken from a signal.

GOP size number of frames which follow a key-frame.

Resolution (pixels) size of a video image (*frame*).

Except for the GOP size, increasing any of these parameters leads to a higher quality audio or video output, consequently incurring a larger file size (requiring more bits to be transmitted).

2.9 Video on-demand and live streaming

There are two different ways to use streaming techniques. In the first one, video on-demand, users request media files which have been previously recorded and compressed and are stored on a server. Today this technique has become very popular, with YouTube being the most popular website offering on-demand streaming. The alternative is live streaming which enables an unbounded transmission where media is generated, compressed, and delivered on the fly. In the case of live streaming there may or not may be a concurrent recording (which could be transmitted later on-demand).

Both streaming techniques may offer the user basic video control functions such as pause, stop, and rewind. Additionally, for on-demand streaming there may be the possibility of issuing a fast-forward command. Note that fast forward is only possible when the media files are stored, thus the future content is known. Of course it is also possible for the system to implement the possibility of a fast-forward command if the user has paused the playback, but this will be limited to moving forward to the recently generated portion of the content.

2.10 Google's Android operating system

Android is an operating system specially designed for mobile devices. It is mainly developed and supported by Google Inc., although other members of the Open Handset Alliance (OHA) have collaborated in its development and release. Table 2.3 reviews Android's version history.

Android is based on a modified version of the Linux kernel and its applications are normally developed in the Java programming language[7]. However, Android has not adopted the official Java Virtual Machine (JVM), meaning that Java Byte code can not be directly executed. Instead, applications run on the Dalvik Virtual Machine (DVM), a JVM-based virtual machine specifically designed for Android. DVM is optimized for mobile devices, which generally have CPU performance and memory limitations. In addition, DVM makes more efficient use of battery power.

Applications are usually released via the Android Market, Google's official online store. Nevertheless, publication of the applications is not restricted, allowing installation from any other source. Figure 2.11 shows the current distribution of Android versions based on the operating system of the devices that have recently accessed the Android Market. As shown, Android's newer versions (the 3.x branch) are only slowly being adopted, for example *Honeycomb* still represents less than 1% of the overall of Android devices, while *Froyo* the predominate version (running on almost 60% of the devices that access the Android Market).

[7] The Android Development Kit (SDK) is freely available from the developers site: `http://developer.android.com/sdk`.

Table 2.3: Google's Android version history.

Version	Codename	Release date	Linux kernel version
1.0	None	23 September 2008	Unknown
1.1	None	9 February 2009	Unknown
1.5	Cupcake	30 April 2009	2.6.27
1.6	Donut	15 September 2009	2.6.29
2.0/2.1	Eclair	26 October 2009	2.6.29
2.2	Froyo	20 May 2010	2.6.32
2.3	Gingerbread	6 December 2010	2.6.35
2.4	Ice Cream Sandwich	Not released	Unknown
3.0	Honeycomb	22 February 2011	2.6.36
3.2	Honeycomb	15 July 2011	2.6.36

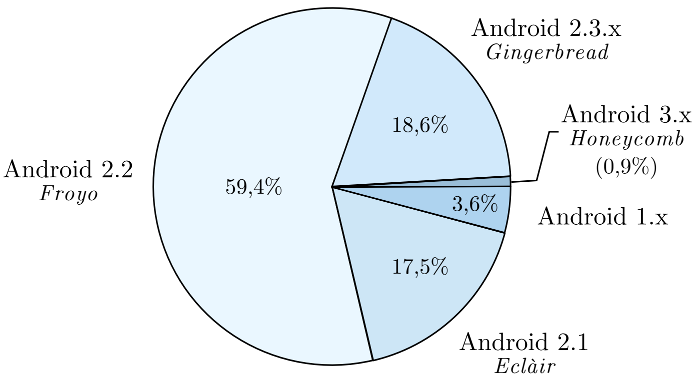


Figure 2.11: Distribution of Android platform versions (as of July 2011). Extracted from [7].

2.10.1 Media formats supported on Android

Android supports several multimedia formats and CODECs [28, p 195-250], including H.263 and H.264. Table 2.4 and Table 2.5 summarize respectively the video and audio CODECs and container formats that are supported. For media playback, only the decoding capabilities are relevant (encoding is typically used for recording purposes).

Table 2.4: Android supported video CODECs and container formats. Extracted from [6].

CODEC	Encoding	Decoding	Container format
H.263	Yes	Yes	3GPP (`.3gp`) and MPEG-4 (`.mp4`)
H.264	No (supported from 3.0 onwards)	Yes	3GPP (`.3gp`) and MPEG-4 (`.mp4`) Only Baseline Profile (BP)
MPEG-4	No	Yes	3GPP (`.3gp`)
VP8	No	No (supported from 2.3.3 onwards)	WebM (`.webm`)

Table 2.5: Android supported audio CODECs and container formats. Extracted from [6].

CODEC	Encoding	Decoding	Container format
AAC LC/LTP	Yes	Yes	3GPP (`.3gp`) and MPEG-4 (`.mp4`, `.m4a`)
HE-AAC v1	No	Yes	3GPP (`.3gp`) and MPEG-4 (`.mp4`, `.m4a`)
HE-AAC v2	No	Yes	3GPP (`.3gp`) and MPEG-4 (`.mp4`, `.m4a`)
MP3	No	Yes	MP3 (`.mp3`) Mono and stereo 8-320 kb/s constant (CBR) or variable bit-rate (VBR)
Vorbis	No	Yes	Ogg (`.ogg`)
PCM/WAVE	No	Yes	WAVE (`.wav`)

2.10.2 Adaptive protocols over HTTP supported on Android

Android's media framework natively supports streaming over RTP and RTSP. Unfortunately, the majority of Android versions do not support any of the adaptive protocols over HTTP mentioned earlier. Only Honeycomb features Apple-HLS natively. During the development of this master's thesis project there was no media player for Android supporting the recent MPEG-DASH standard. This section explores the existing compatibilities with regard to Apple-HLS, Microsoft-LSS, and Adobe-HDS.

2.10.2.1 Apple-HLS support

At the moment there are a few implementations of Apple-HLS for Android:

- **NexPlayer™** was released in September 2010 by Nextreaming Corp. They claim to support Apple's adaptive streaming approach. Unfortunately, neither the application nor detailed features are publicly available at their website.
- **VPlayer** is a commercial video player available from the Android Market[8]. Unfortunately, VPlayer is not compatible with all Android devices.
- **Daroon Player** is a free video player developed by Daroonsoft and offered from the Android market[9]. It supports a wide variety of media formats and streaming protocols, including RTSP and Apple-HLS.

2.10.2.2 Microsoft-LSS support

Microsoft's adaptive streaming approach for Android is not available yet officially, although Microsoft has indicated that they soon plan to support it through a Silverlight[10] browser plug-in soon. However, the open-source implementation of Silverlight for Unix-based operating systems (*Moonlight*), has been experimentally ported to Android[11].

[8] A free trial of VPlayer 0.9.9 can be downloaded from `https://market.android.com/details?id=me.abitno.vplayer.t`.

[9] Daroon Player 1.0.1 is available from `https://market.android.com/details?id=com.daroonsoft.player`.

[10] Silverlight is a Microsoft's application framework for creating rich Internet applications, with features similar to Adobe's Flash.

[11] More information can be found at `http://jeffreystedfast.blogspot.com/2011/04/moonlight-on-android.html`.

2.10.2.3 Adobe-HDS support

The Adobe Flash 10.1 plug-in for browsers is available[12] for Android 2.2, although it is only compatible with a limited variety of Android devices[13]. The plug-in supports RTP and RTSP streaming, HTML progressive download, Adobe's Flash Streaming, and Adobe-HDS.

2.11 Comparison among the different HTTP-based adaptive solutions

Table 2.6 summarizes the main features of the most relevant HTTP-based adaptive streaming solutions: Microsoft-LSS, Apple-HLS, and MPEG-DASH.

Table 2.6: Comparison among Microsoft-LSS, Apple-HLS, and MPEG-DASH. Extracted from [1, 55, 59]

Feature	Microsoft-LSS	Apple-HLS	MPEG-DASH
Specification	Proprietary	Proprietary	Standard
Video on demand	Yes	Yes	Yes
Live	Yes	Yes	Yes
Delivery protocol	HTTP	HTTP	HTTP
Origin server	MS IIS	HTTP	HTTP
Media container	MP4	MPEG-TS	3GP or MP4
Supported video CODECs	Agnostic	H.264	Agnostic
Recommended segment duration (s)	2	10	flexible
End-to-end latency (s) (variable, depending on the size of segments)	> 1.5	30	> 2
File type on server	Contiguous	Fragmented	Both

In order to implement a functional live streaming service for Android, all the limitations of the operating system must be considered, as well as the possibility of deploying a compatible server. We explicitly considered the following:

- Adobe's and Microsoft's solutions are proprietary and both require specialized servers. Such approaches increase cost and decrease the openness of the resulting service.
- Apple-HLS intends to be a IETF standard, but its specification (regarding CODEC and container format of segments) is strict enough to consider a straightforward implementation for Android. Although H.264 is a fully supported CODEC on Android, MPEG-TS as a container format is only included in Android version 3.0 (codenamed Honeycomb) and onwards. Therefore, a file conversion is required to support Apple-HLS.
- MPEG-DASH is an emerging adaptive HTTP streaming standard which is flexible enough to be implemented in devices with Android built-in.

[12] Detailed features and requirements can be read at `http://kb2.adobe.com/cps/860/cpsid_86018.html`.
[13] Certified devices are listed at Adobe's official site `http://www.adobe.com/flashplatform/certified_devices`.

Chapter 3

Related work

"If you wish to make an apple pie from scratch, you must first invent the universe."

– Carl Sagan

Extensive work has been carried out in the area of adaptive streaming over HTTP (i.e., using HTTP as a delivery protocol). Multiple rate adaptation mechanisms have been proposed and experiments have been performed under different network conditions. An extensive evaluation of adaptive streaming, including live sources under heterogeneous network rates, has been carried out in [26], although using RTP and RTSP as delivery protocols.

In [62] the media segmentation procedure has been utilized to provide a HTTP streaming server with dynamic advertisement splicing. Unfortunately, the evaluation only included experiments under homogeneous bit-rate conditions, therefore no rate-adaptation was performed in either server or client.

The fundamental capabilities of the 3GPP's MPEG-DASH standard have been demonstrated in [69], pointing out the most significant properties of the media presentation descriptor (MPD or simply *manifest* file). Long-session experiments for both on-demand and live video content were performed, featuring advertisement insertion. An experimental comparison between Apple's HLS and MPEG-DASH over an HSPA network has been carried out in [67], although only on-demand content was considered.

The benefits of the Scalable Video Coding (SVC) (an extension of H.264/MPEG-4 AVC [42, Annex G]) in a MPEG-DASH environment are demonstrated in [64]. Media content is divided into SVC layers and time intervals. By means of this H.264 extension, storage requirements and congestion at the origin server are claimed to be reduced. SVC in conjunction with Multiple Descriptor Coding (MDC) were tested over a peer-to-peer (P2P) video on-demand system in [3]. An initial adaptation algorithm is suggested, based on the client's display resolution, bandwidth, and processing power. During playback, a progressive quality adaptation is carried out, monitoring the buffer state and analyzing the change of download throughput during the buffering process.

In [57] a MPEG-DASH prototype is presented as a plug-in for the VideoLan player 1.2.0 (VLC). A novel rate adaptation algorithm for MPEG-DASH was proposed in [49], using a smoothed throughput measurement (based on the segment fetch time) as the fundamental metric. Therefore, the algorithm can be implemented at the application layer since it does not consider TCP's round-trip time (RTT). Upon detecting that the media bit-rate does not match the current end-to-end network capacity, an mechanism for conservative up-switching and aggressive down-switching of representations is invoked.

A pre-fetching approach for user-generated content video is presented in [45]. It predicts a set of videos which are likely to be watched in the near future and downloads them before they are requested. The benefits of the pre-fetching scheme are compared with a traditional caching scheme are demonstrated in a number of different network scenarios.

An intensive experiment on rate-adaptation mechanisms of adaptive streaming is presented in [5]. Three different players (OSMF, Microsoft Smooth Streaming, and Netflix) are evaluated in a broad variety of scenarios (both on-demand and live) with both persistent and short-term changes in the network's available bandwidth and shared bottleneck links. J. Yao, et al. [73] carried out an empirical evaluation of HTTP adaptive streaming under vehicular mobility.

An experimental analysis of HTTP-based request-response streams compared to classical TCP streaming is presented in [46]. It is claimed that the HTTP streams are able to scale with the available bandwidth by increasing the chunk size or the number of concurrent streams.

A Quality Adaptation Controller (QAC) for live adaptive video streaming which employs feedback control theory is proposed in [23]. Experiments with greedy TCP connections are performed over the Akamai High Definition Video Server (AHDVS), considering bandwidth variations and different streams which share a network bottleneck.

Evensen, et. al. [27] present a client scheduler that distributes requests for video over multiple heterogeneous interfaces simultaneously. Segments are divided into smaller sub-segments. They experimented with on-demand and quasi-live streaming. Evaluations have been performed over three different types of streaming: on-demand (assuming infinite buffer, only limited by network bandwidth), live streaming with buffering (the whole video is not available when streaming starts), and live streaming without buffering. The last scenario considers *liveness* as the most important metric, thus segments are skipped if the stream lags too far behind the broadcast.

An elaborated comparison between Apple's HLS on iPhone and RTP on Android 1.6 is presented in [61]. In particular, the impact of packet delay and packet loss are evaluated with respect to the start-up delay and playback, as well as TCP traffic fairness.

From previous work in the area of adaptive streaming we can deduce that there is still a lack of evaluation on mobile devices, especially those using the most recent standards (such as MPEG-DASH, introduced in section 2.4.5) for the particular case of live content sources. This master's thesis aims to fill the gap by deploying a full service for mobile devices, providing an extensive evaluation (over a set of heterogeneous network scenarios similar to the experiments carried out in [5]) with different adaptation mechanisms (also described as feedback controllers). These mechanisms are substantially based on the algorithms proposed in [23, 49, 67], although some enhancements have been made, specifically: (1) a mechanism to discard segments upon abrupt reduction of the network's available bit-rate, and (2) a procedure to lower the selected media quality on the client's side in case of a buffer underflow.

Chapter 4

Design and implementation

”Simplicity is the prerequisite for reliability”

– Edsger W. Dijkstra

This chapter explains each of the elements of the overall system (depicted in figure 4.1). The most important entities are the server and the client which are explained in section 4.3 and section 4.4, respectively. Communication between these two entities flows over HTTP. The advantages of using HTTP were described in chapter 2. Synchronization of the client and server are described in section 4.2.

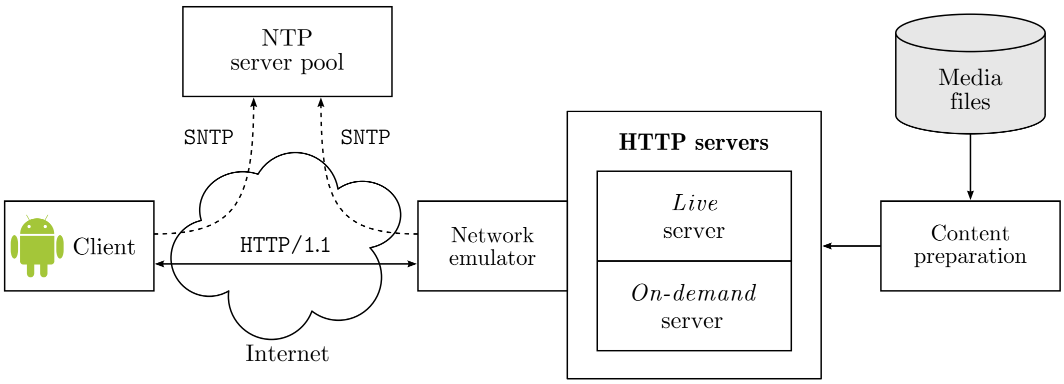


Figure 4.1: System architecture.

Two types of servers have been deployed depending on the nature of content: one of the servers provides video *on-demand* (section 4.3.1) and the other offers *live* video (section 4.3.2). As, explained in chapter 2, the media content needs to be encoded and segmented to satisfy the specifications of the MPEG-DASH standard and Apple-HLS (see section 2.4.5 and section 2.4.2 respectively). This procedure is represented by the *content preparation* module, which is characterized in section 4.1.

In reality, network traffic conditions are susceptible to change. A *network emulator* is described in section 4.5. This network emulator enables controlled experiments to be performed with different bit-rates, various delays, and different packet loss rates.

4.1 Content preparation

Figure 4.2 depicts the modules which multiplex the input media content into different quality streams followed by a segmentation procedure. The transcoder part and the selection of the R representations are explained in section 4.1.1, followed by the segmentation, combiner, and indexing parts in sections 4.1.2 and 4.1.3, respectively. The overall output will be pushed to the HTTP servers as is explained in sections 4.3.1 and 4.3.2.

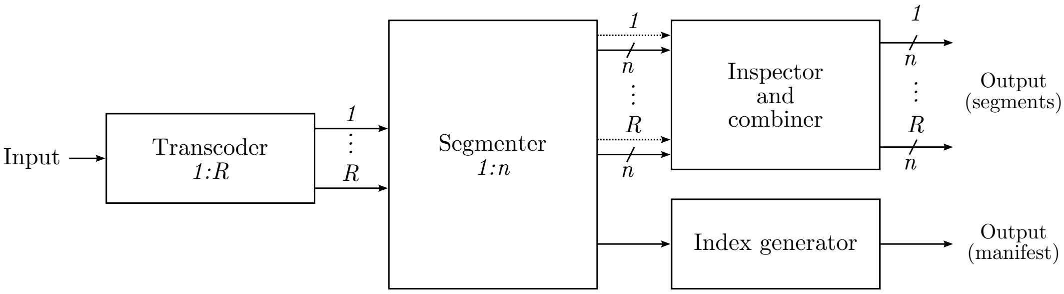


Figure 4.2: Modules for content preparation. A media file is indicated as input. R different representations are generated, producing n segments for each original segment. An index file is also produced as output.

4.1.1 Transcoder module

The transcoder module is responsible for generating different quality levels, as described in section 2.8. This module receives a media file as input (containing a video and audio stream, at least one of them is required to be present), then produces from the audio/video stream several files encoded at different bit-rates. Audio and video are combined using the MP4 container format. This module is implemented as a BASH script and relies on the FFmpeg [13] and x264 [71] libraries[1]. Listing 4.1 and listing 4.2 illustrate the use of the `ffmpeg` command and the x264 parameters applied, in order to satisfy the H.264 Baseline Profile (introduced in the CODECs section on page 16).

Listing 4.1: FFmpeg command line used in the transcoder module. Note the fixed parameters defined in the first line: frame rate, resolution, aspect ratio, and GOP size.

```
ffmpeg -i $INPUT -y -r 25 -s 480x320 -aspect 3:2 -g 25 \
      -acodec libfaac -ab $ABITRATE -ac $CHANNELS -ar $SAMPLE_RATE \
      -vcodec libx264 $X264_PARAMS -b $VBITRATE -bufsize $VBITRATE -maxrate $VBITRATE \
     -async 10 -threads 0 -f $FILE_FORMAT -t $CLIP_DURATION $OUTPUT
```

Listing 4.2: Parameters used for H.264 encoding at Baseline Profile. Note that the *coder* attribute is set to 0 to restrict to H.264 Baseline Profile.

```
X264_PARAMS=-coder 0 -flags +loop+mv4 -cmp 256 -subq 7 -trellis 1 -refs 5 -bf 0 -wpredp 0
   -partitions +parti4x4+parti8x8+partp4x4+partp8x8+partb8x8 -flags2 -wpred-dct8x8
   -me_range 16 -g 25 -keyint_min 25 -sc_threshold 40 -i_qfactor 0.71 -qmin 10
   -qmax 51 -qdiff 4
```

4.1.2 Segmenter and combiner modules

The segmenter module receives a set of media files encoded at different bit-rates and splits them into several segments (with similar features to those described in [51]). In addition, an *initialization segment* is also generated as an output. The initialization segment provides the meta-data[2] which describes the media content, without including any media data. Furthermore, it supplies the timing information (specifically the DTS and PTS, as defined in section 2.5.2) of every segment.

This module reads the different parts or *boxes* of the container format and separates the file into several pieces of approximately the same duration (this duration is passed in as an input parameter). It attempts GOP alignment between all the input files, that is, *segments* always start with a key-frame[3] and the *breaking point* is the same for all representations.

[1] The FFmpeg capabilities can be found in the appendix B.
[2] In 3GPP's terminology, the *ftyp* box, *moov* box and optionally the *pdin* box.
[3] This is an assumption for this prototype. MPEG-DASH supports segments which do not start with a key-frame.

Several tools can be used to analyze the structure of a media file. In particular, `MP4box`[4] is able to list all the elements of a container format in a NHML file (an XML-based type for multiplexing purposes), indicating which samples are a Random Access Points (RAPs) and which are not. Listing 4.3 shows a sample NHML output from `MP4box`.

Listing 4.3: NHML example file produced by MP4box.

```xml
<?xml version="1.0" encoding="UTF-8" ?>
<NHNTStream version="1.0" timeScale="25" streamType="4" objectTypeIndication="33"
specificInfoFile="..." width="480" height="320" trackID="1"
baseMediaFile="..." >
   <NHNTSample DTS="0" dataLength="1082" isRAP="yes" />
   <NHNTSample DTS="1" dataLength="11" />
   ...
   <NHNTSample DTS="25" dataLength="413" isRAP="yes" />
   ...
   <NHNTSample DTS="4644" dataLength="413" isRAP="yes" />
   ...
</NHNTStream>
```

Sequentially, the combiner module produces segments that can be played on Stagefright. To enable this it transforms all the chunks into self-contained files taking the header information from the initialization segment. Since the live stream will consist of several self-contained segments, there is no need to modify the DTS or PTS.

4.1.3 Indexing module

An index of all the segments generated in the previous steps must be pushed to the HTTP servers. This module inspects the segments that have been produced and generates an ordered list (MPD) which satisfies the MPEG-DASH standard guidelines. Two different types of MPDs are created:

Standard form provides a full file, containing all the representation levels and a list of available segments for each one (i.e., all URIs).

Template form provides a shorter file. It uses the `Urltemplate` tag, indicating the index bounds (`startIndex` and `endIndex`). This type of manifest is especially useful in the case of live content, since fewer modifications must be done for every MPD update.

4.2 Synchronization between server and client

In the particular case of live video content, it is useful if both server and client have the same sense of time. *Synchronization* in this context means that provider and consumer sides communicate with an external time server to set their clocks to the same accurate time base. Compared to a non-synchronized scheme, clients do not need to make so many queries to the server (*HTTP requests*) since the clients knows in advance when new content will be available.

Synchronization is achieved by means of the Simple Network Time Protocol (SNTP) [56], which is based on the Network Time Protocol (NTP). Fortunately, many NTP public servers are freely available on the Internet. The NTP Pool project[5] has been selected for this purpose because it provides a pool of free NTP servers operating on a reasonable-use basis. Indeed, the implementation of this prototype follows the recommendations of [56, sec. 10] to perform a fair use of the time servers, thus periodic requests are never performed more frequently than every 30 seconds. Figure 4.3 (a) depicts the relation between client, server, and the NTP pool.

[4] MP4box. Available from: `http://gpac.wp.institut-telecom.fr/mp4box`

[5] The pool.ntp.org project is a virtual cluster of timeservers which provides reliable NTP service. Available from `http://www.pool.ntp.org`.

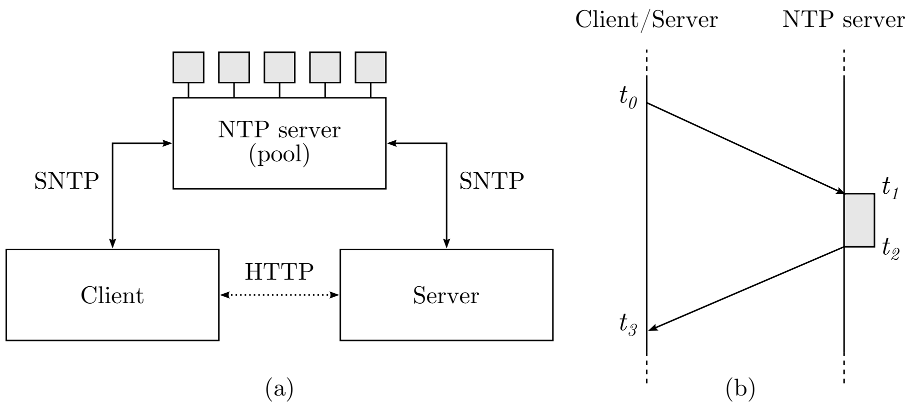


Figure 4.3: (a) Communication among client, server and NTP server pool. (b) A simple SNTP request.

Figure 4.4 depicts the header of a NTP packet. There are three fields needed for the simplest synchronization: time-stamp of client's request (`Originate Timestamp` field), time-stamp of the client's request arrival at the time server (`Receive Timestamp` field), and a time-stamp of when the server's response was transmitted (`Transmit Timestamp`). The rest of header fields (such as `Poll`, `Stratum`, and `Precision`) and are not considered here for simplicity (for further details see [56]).

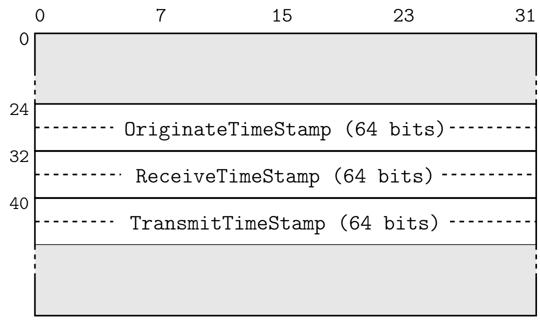


Figure 4.4: NTP packet header. Relevant fields for the synchronization procedure are shown highlighted.

The synchronization procedure (shown in figure 4.3 (b)) is performed in this prototype as follows:

1. The NTP server receives the request and adds the time when the request was received (t_1) to the `Receive Timestamp` header field.
2. A response is sent with a new time-stamp (t_2) indicated in the `Transmit Timestamp` header field.
3. The client receives the response and computes the offset (T_{offset}) that should be applied to its local time.

The offset (T_{offset}) (between the machine's local time and the NTP time) and round-trip-time (RTT) (path delay between the client and NTP server) can be determined as:

$$T_{offset} = \frac{(t_1 - t_0) + (t_2 - t_3)}{2} \tag{4.1}$$

$$RTT = (t_3 - t_0) - (t_2 - t_1) \tag{4.2}$$

Both HTTP client and HTTP server will add their respective offset to their local time (note that offset can be a negative quantity). In particular, the HTTP client's request time becomes $t_0 + T_{offset}$. For the HTTP client's operations that need an absolute time reference, the offset is simply added to the result of a Java method `System.currentTimeMillis()` invocation.

4.3 HTTP Servers

Two types of HTTP servers have been deployed in this architecture. The first server is suitable only for video on-demand, whereas the second server serves content from live sources. Each of these servers is briefly explained in the following sections.

4.3.1 On-demand server

An Apache [47] HTTP server acts as video on-demand server. Apache has been selected because it is robust and easy to deploy on Gnu/Linux machines. The purpose of this server in our architecture is simple: this HTTP server provides a list of *manifest* files which contain URLs for the segments generated for every representation. Table 4.1 lists the Multipurpose Internet Mail Extensions (MIME) [32] types that needed to be added to the Apache configuration. These types are added to the Apache's configuration file (`httpd.conf`).

Table 4.1: Additional MIME types needed for the Apache HTTP server.

Type	MIME type	File extension
DASH manifest	video/vnd.3gpp.mpd	`.3gm`
DASH video segment	audio/vnd.3gpp.segment	`.3gs`
Apple-HLS playlist	application/x-mpegURL	`.m3u8`
Apple-HLS video segment	video/MP2T	`.ts`

4.3.2 Live server

We have decided to use Twisted [30] is an event-driven networking engine written in Python [35], licensed under the MIT license[6], because it supports a wide variety of protocols and it contains multiple resources to deploy a simple web server. The live server is based on a content-loop server developed previously at Ericsson GmbH. The content-loop server has been modified to satisfy the requirements of the system architecture proposed in this chapter.

HTTP responses are generated by an abstract class which extends Twisted's `Resource` type. A simplification of this class is shown in listing 4.4.

Listing 4.4: DashResource abstract class.

```
class DashResource(resource.Resource):

   def returnSuccess(self, request, content, contentType):
      ... // Set HTTP headers
      request.setHeader('Content-Length', "%d" % len(content))
      request.setResponseCode(http.OK)
      request.write(content)
      request.finish()

   def returnFailed(self, request, error):
      request.setResponseCode(404)
      request.write(error)
      request.finish()
```

[6] The conditions of a MIT license can be found in `http://www.opensource.org/licenses/mit-license.php`.

The live server receives all media segments and manifest files that have previously been generated (as explained in section 4.1) and prepares a live source. In our prototype, live content is provided by looping several clips and numbering all segments by means of the mathematical *modulo* function to produce an unbounded stream of content. Segments requested with a index greater than the available segments are automatically pointed to an existing segments modulo the total number of segments, thus providing an infinite loop of video content. Segments are numbered with an arbitrary length integer. The behaviour of the server is summarized as follows:

1. Server starts. The server inspects all the representations and segments in order to generate the first *manifest* file. This MPD has the attribute `type` set to `Live`, indicating that the availability of segments is limited (in time - i.e., that they have to be requested within a bounded period of time) and susceptible to change (i.e., asking for a given segment number at a later time might result in a different segment of media content).

2. A set of segments is offered according to the *available shifting time* or *window*, indicated in the server configuration. The server calculates when the next update will occur, this primarily depends on the segments' duration. When the duration of one segment has elapsed, a new update is executed.

3. If a HTTP request is received from a client, the *overloaded* Twisted method `render(self, request)` is invoked. Different situations will arise depending on the client's request and server's state:

    a) If the client is requesting the **MPD file**, then the server simply responses with the last updated XML content in this file.

    b) If the client is requesting a **media segment** numbered with an index i (as depicted in figure 4.5). The server checks whether the segment i belongs to the set of available segments:

        - If this segment is within the available shifting time, then the server performs the modulo operation and replies with a satisfactory HTTP response (`code 200`), starting the transmission of the segment.
        - Otherwise, the server replies with a HTTP unsatisfactory response (`code 404`) and a simple message. There are two possible situations in which situation may occur: the client requests the segment too soon (segment not available yet) or the segment was requested too late (segment no longer available).

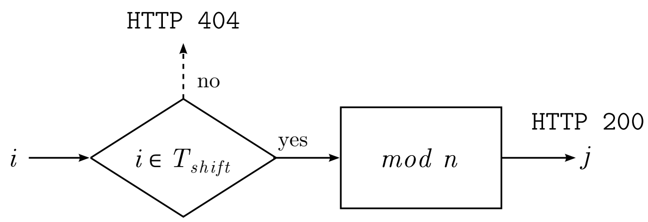


Figure 4.5: Characterization of the Live server. T_{shift} represents the indexes of segments within the available shifting time, n the number of segments for one representation.

4.4 Client

In this architecture, an Android application acts as the client. Android cellphones have sufficient capabilities to provide video playback and perform communication over HTTP.

4.4.1 Features

The client developed in this master's thesis project has the following features:

- Media adaptation by means of several bit-rate algorithms (section 4.4.2). The optimal representation level is selected depending on the network's restrictions/performance.
- Supports the MPEG-DASH protocol, as summarized in section 2.4.5. In particular, this client is compatible with the *manifest* files (MPD) that the server produces.
- Minimal support for Apple-HLS (section 2.4.2), more specifically it is compatible with Apple's extended playlists (`.m3u8`) and conversion of the MPEG-TS container format to the 3GP/MP4 formats supported on Android (a complete list of Android supported CODECs and media formats were listed in table 2.4 on page 21).
- Handling of HTTP connections occurs as background tasks, thus preventing interruptions of the media player.
- Database management of content sources, as indicated by the URL of MPDs or extended M3U playlists.
- Automated search of manifest files.
- A rich Graphical User Interface (GUI) is provided. During playback, the client displays dynamic plots which provide accurate information about the actual bandwidth utilization or when segments are downloaded. A full-screen mode is automatically launched when the cellphone is turned to a landscape orientation.

4.4.2 Adaptation mechanisms

The adaptation mechanisms proposed in this master's thesis project follow three requirements:

1. Playback shall not be stopped (i.e. buffer underflow should be avoided).
2. Optimal use of network resources, selecting the highest possible bit-rate level while meeting requirement 1.
3. Switching to the appropriate quality level should be performed as rapidly as possible.

If any of these requirements is not satisfied, this indicate an erroneous use of the available bandwidth. If the first requirement is not met, this indicates that the choice of representation level selected was overestimated. As a result, the time it will take to download the next segment will be longer than the segment's own duration, leading to playback interruptions if the representation level is not reduced. Not fulfilling the second requirement indicates that the representation level has been underestimated. In this case, the user of this client will not experience the best possible quality - however, they will be able to watch/listen to the content at less than the highest possible quality. The third requirement involves a design choice: when switching events may occur. In our implementation, the adaptation mechanism is always invoked right after segments are downloaded (buffered). Therefore, all the proposed mechanisms are equally fast, but provide different criteria for the appropriate quality level.

Three adaptation mechanisms has been proposed: *aggressive* adaptation, *conservative* adaptation, and *mean* adaptation. Details of these three mechanisms are explained in the following subsections.

4.4.2.1 Aggressive adaptive mechanism

The aggressive mechanism is defined in algorithm 1. This mechanism has the following characteristics:

- The starting quality level is the level with the lowest bandwidth requirement.
- Selection of the lowest representation level occurs when the playlist (*buffer*) is empty.
- Multiple step switching, i.e., the selected quality level can be adjusted drastically up and down upon termination of the algorithm, since the switching operations are contained in a `while` loop.

The aggressive mechanism determines the optimal quality level considering *only* the last throughput measurement τ. The selected quality level is increased when the last throughput measurement is greater than the current representation's bit-rate. Otherwise, the quality level is decreased.

Algorithm 1: Aggressive adaptive algorithm.

Data: Last throughput measurement τ, playlist P, ordered array of representations r with size $|r| = R$, current representation index r_{curr}

begin
- **if** $|P| = 0$ **then**
  - *Switch-down to minimum:* $r_{curr} \longleftarrow 0$
- **else**
  - **if** $\tau > r[r_{curr}]$ **then**
    - **while** $r[r_{curr}+1] < \tau$ *and* $r_{curr} < R-1$ **do**
      - *Switch-up one level:* $r_{curr} \longleftarrow r_{curr}+1$
  - **else**
    - **while** $r[r_{curr}-1] > \tau$ *and* $r_{curr} > 0$ **do**
      - *Switch-down one level:* $r_{curr} \longleftarrow r_{curr}-1$

This mechanism is referred to as aggressive because it provides a rapid change in behaviour in its response to bandwidth fluctuations. Nonetheless, problems may arise with this algorithm due to short-term bit-rate peaks. Since it will try to switch level, the increased bandwidth might not be available for all of the next segment's download, hence the client will not have time to download the new segment. The expected advantages and disadvantages of the aggressive mechanism are:

Advantage best utilization of available bandwidth.

Disadvantage high sensitivity to changes in available bandwidth. This could lead to selection of too high quality level (due to bit-rate peaks).

4.4.2.2 Conservative adaptive mechanism

The conservative mechanism (specified in algorithm 2) is based on the aggressive mechanism (described in section 4.4.2.1). However, it enhances the selection of the quality level by adding a *sensitivity* parameter which is applied to the last measured throughput τ. Consequently, the client becomes *less* sensitive to the available network bit-rate, resulting in a more *conservative* selection of the representation level. In this mechanisms, a sensitivity of the 70% is applied (note that a sensitivity of 100% will produce the same behaviour as the aggressive mechanism). As a result, the expected advantages and disadvantages of the conservative mechanism are:

Advantage this algorithm avoids pauses while providing a continuous playback experience, since the selected quality level is systematically slightly below the optimal.

Disadvantage by underestimating the available network bit-rate, this algorithm leads to a non-optimal bandwidth utilization and less than the highest possible quality.

Algorithm 2: Conservative adaptive algorithm.

Data: Last throughput measurement τ, playlist P, ordered array of representations r with size $|r| = R$, current representation index r_{curr}

```
begin
    Sensitivity ⟵ 0.7
    if |P| = 0 then
        Switch-down to minimum: r_curr ⟵ 0
    else
        τ' ⟵ τ × Sensitivity
        if τ' > r[r_curr] then
            while r[r_curr + 1] < τ' and r_curr < R − 1 do
                Switch-up one level: r_curr ⟵ r_curr + 1
        else
            while r[r_curr − 1] > τ' and r_curr > 0 do
                Switch-down one level: r_curr ⟵ r_curr − 1
```

4.4.2.3 Mean adaptive mechanism

The mean mechanism is built upon the aggressive mechanism (described in section 4.4.2.1). Using this mechanism the optimal quality level decision is based upon the arithmetic mean[7] of the last three throughput measurements (see algorithm 3). The overall behaviour is similar to the adaptive mechanism proposed in [67] where the last five measurements were considered. In the mean mechanism the throughput average is calculated based on the last **three** measures (τ_1, τ_2, and τ_3) In addition, a high sensitivity parameter is applied to the throughput average. The expected advantages and disadvantages of the mean mechanism are:

Advantage better utilization of bandwidth compared to the aggressive and the conservative mechanism in the long-term. Reduced sensitivity to bit-rate peaks and troughs.

Disadvantage longer reaction time when there is a large bit-rate variation. Selection of the appropriate quality level might be performed in several switching steps.

[7] Another statistical operation such as the median could have been used. However, the median is *less* sensitive than the mean to extreme fluctuations. Consider the following example: the last three throughput measurements are 100 kb/s, 200 kb/s and 300 kb/s. The mean and median of the ordered list {100, 200, 300} are both 200 kb/s. If the next measurement is, for instance, 600 kb/s, the ordered list is updated to {100, 200, 600}, where the mean is increased to 300 kb/s and the median has not changed.

Algorithm 3: Mean adaptive algorithm

Data: Last 3 throughput measurements τ_1, τ_2, and τ_3, playlist P, ordered array of representations r with size $|r| = R$, current representation index r_{curr}

begin

  $Sensitivity \longleftarrow 0.95$

  $\tau_{mean} \longleftarrow \frac{\tau_1+\tau_2+\tau_3}{3}$

  if $|P| = 0$ **then**

    Switch-down to minimum: $r_{curr} \longleftarrow 0$

  else

    $\tau'_{mean} \longleftarrow \tau_{mean} \times Sensitivity$

    if $\tau'_{mean} > r[r_{curr}]$ **then**

      while $r[r_{curr}+1] < \tau'_{mean}$ *and* $r_{curr} < R-1$ **do**

        Switch-up one level: $r_{curr} \longleftarrow r_{curr}+1$

    else

      while $r[r_{curr}-1] > \tau'_{mean}$ *and* $r_{curr} > 0$ **do**

        Switch-down one level: $r_{curr} \longleftarrow r_{curr}-1$

4.4.3 Module characterization

Figure 4.6 illustrates the modules which constitute the client's application. A dashed line separates the prototype from the cellphone's external resources, such as the available memory (*external storage*) and the *user interface*. The user interface represents the user's interaction with the device's buttons and (where available) touch-screen.

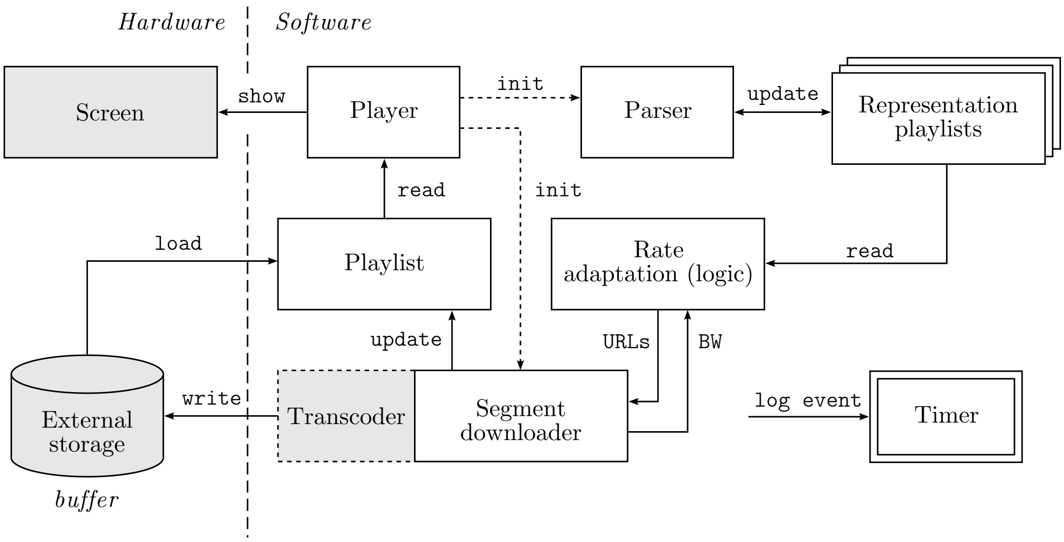


Figure 4.6: Overview of the client's application modules. The dashed line separates the device's *hardware* resources from the client's application (*software*).

The client's functionality can be summarized as follows. The *player module* (described in section 4.4.4) starts the application and manages the video controller and graphical resources, in particular, the Android surface where the video is displayed. The *parser module* (described in section 4.4.5) is launched and it transforms the index file or *manifest file* into several *representation playlists*, each one corresponding to a determined quality level. If the parsing procedure is successfull, this module periodically checks for manifest updates as a background task.

Next the *segmeter-downloader module* (described in section 4.4.6) starts to request the media segments over HTTP using persistent connections. A query is sent to the *rate-adaptation module* (described in section 4.4.7) after each download. The rate-adaptation module is responsible for selecting the most appropriate quality level depending on the network conditions. Consequently, the *transcoder module* performs a media conversion when necessary, as was explained in section 4.1.1.

Segments successfully downloaded into the buffer are added to a primary *playlist* (described in section 4.4.6), which enumerates the received pieces of content. Changes in the playlist will be constantly monitored by the player module.

The timer module (described in section 4.4.9) calculates the timing of all the events which take place in the system. This information is essential for the evaluation described in the next chapter.

4.4.3.1 Activities

In Android terminology, an *activity* is an application component that provides a graphical interface, listening to the user's interaction. Activities are analogous to *windows* in typical computer applications as they provide graphical components (such as text or buttons) and can be opened or closed in a specific order.

Activities are controlled by several listeners: `onCreate()` is the most important method, as this method is invoked at the beginning of the activity. The remaining listeners (`onResume()`, `onStop()`, `onPause()`, `onRestart()`, and `onDestroy()`) have been adapted to satisfy the desired behaviour of the application[8]. In particular these methods:

- Stop the background tasks (*threads*) when the user exits the application.
- Handle the device's *orientation* changes, i.e., when the user turns the cellphone more than 90 degrees. Two orientations are defined in Android: *landscape* (horizontal) and *portrait* (vertical), as illustrated in figure 4.7.

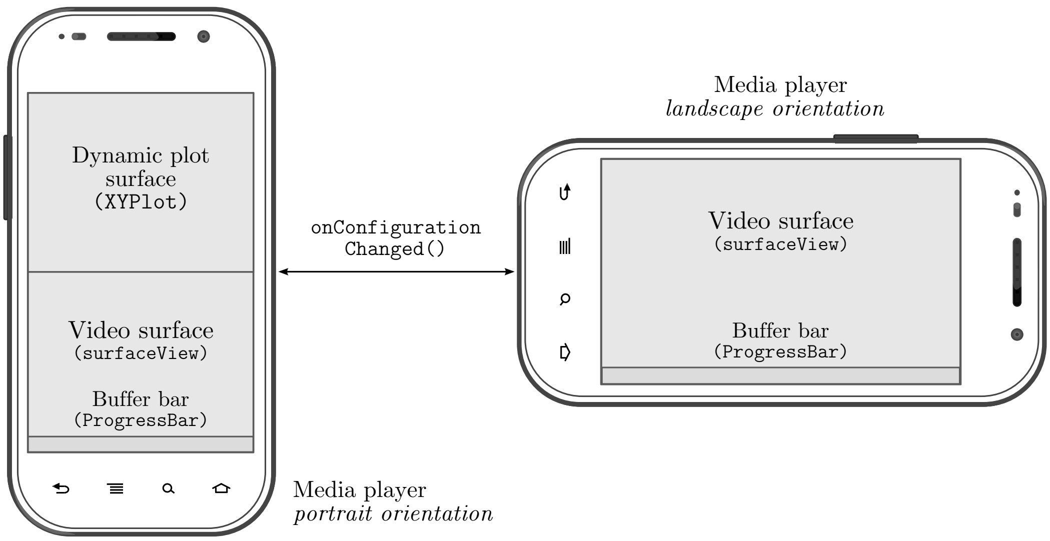


Figure 4.7: Activity orientation.

Three activities have been designed in this prototype, see figure 4.8. The first activity, depicted on the left of the figure, displays a list of manifest files. The user can easily add, modify and

[8]Detailed information about these Java methods and the states of an activity can be found at http://developer.android.com/reference/android/app/Activity.html.

remove entries using the GUI components (Android's `contextMenu`). When an element of the list is selected, the second activity is started. This step is only used for our evaluation, since it selects the proposed adaptive algorithms (these algorithms will be introduced in section 4.4.7). The last activity handles the actual media playback, displaying both the video and dynamic plots on the screen. A demonstration of the GUI can be found in the appendix A.

All activities have to be described in the `AndroidManifest.xml` file, as shown in the simplified in listing 4.5. Lines 5-8 indicate the first activity to be launched when the application is started (`ContentSelection` activity). In addition, the Android OS permissions required for the application need to be specified. In our case these permissions are:

- `WRITE_EXTERNAL_STORAGE`: to write into the MicroSD card (the location of the *buffer* in this prototype).
- `INTERNET`: to open a HTTP communication with the servers, described in section 4.3.

Listing 4.5: Simplified version of the Android manifest XML-file.

```
1  <?xml version="1.0" encoding="utf-8"?>
2  <manifest xmlns:android="http://schemas.android.com/apk/res/android">
3     <application>
4          <activity android:name="ContentSelection">
5              <intent-filter>
6                 <action android:name="android.intent.action.MAIN" />
7               <category android:name="android.intent.category.LAUNCHER" />
8             </intent-filter>
9          </activity>
10         <activity android:name="RateAlgorithmSelection"/>
11         <activity android:name="Player"/>
12      </application>
13       <uses-permission android:name="android.permission.WRITE_EXTERNAL_STORAGE" />
14       <uses-permission android:name="android.permission.INTERNET" />
15 </manifest>
```

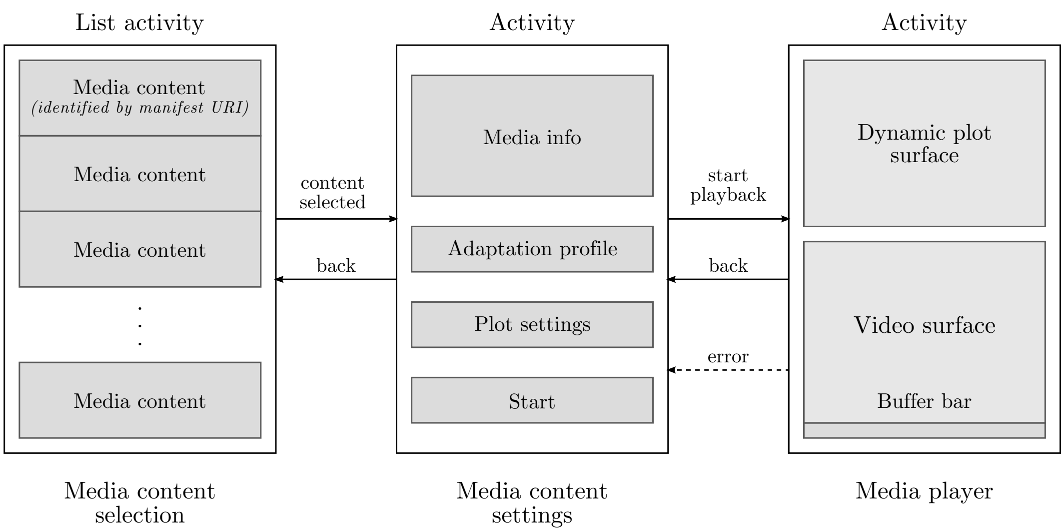


Figure 4.8: Activities of the client application.

4.4.4 Player module

The player module is the essential and main component of the application. It manages the playback of the media segments, displaying the video stream on the screen. Figure 4.9 depicts the set of actions performed in this module. Initially the module creates two background tasks with the following purposes: (1) periodically parse the *manifest* file (to be performed by the parsing module, explained in detail in section 4.4.5) and (2) download the media fragments (performed by the segment-downloader module, described in section 4.4.6).

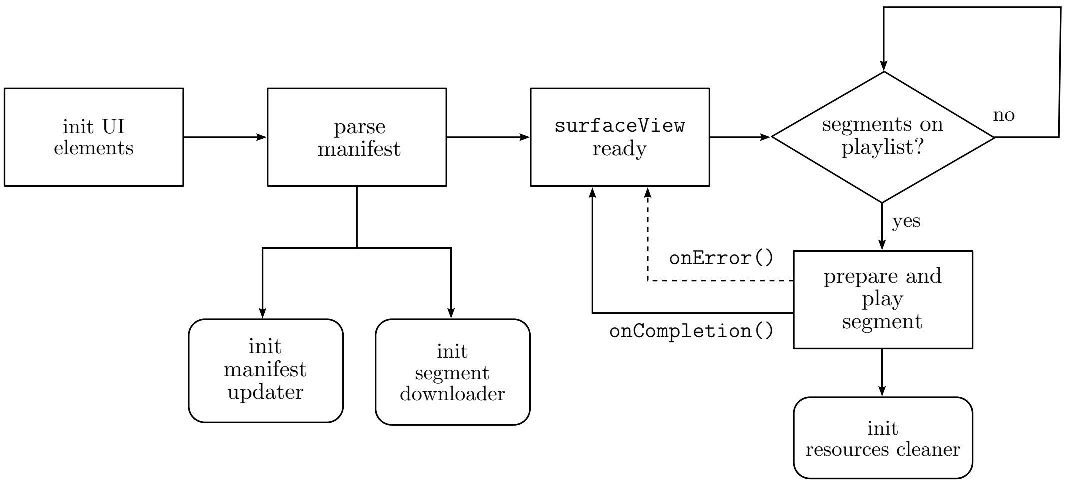


Figure 4.9: Sequence of events produced in the player module.

The player module examines a generated *playlist* of buffered segments (which is regularly updated by the segment-downloader module). The player continues this processing until the main activity is closed.

4.4.4.1 Video surface management

A video surface is the main element of a video player. This surface is represented as a `SurfaceView` element in the Android framework. The media files will be played (i.e., displayed) on this surface. The process of binding a video surface with to instance of a Android `MediaPlayer` object takes place in four steps (as depicted in figure 4.10):

1. The surface is created in an Activity and its surface holder is created. The type of the surfaceHolder must be `SURFACE_HOLDER_PUSH_BUFFERS` in order to play audio and video, otherwise video will not be shown on the screen.

2. This holder starts three listeners:

    a) `surfaceCreated`: triggered when the surface is ready.

    b) `surfaceChanged`: detects surface changes, for instance a change in its size (which could result in a change in video resolution).

    c) `surfaceDestroy`: launched when the activity which holds the surface ends.

3. When `surfaceCreated()` is invoked, the holder can be bound to an instance of a `MediaPlayer` object. At this point it is possible to start loading a media file. Listing 4.6 shows how the first segment will be prepared in a new thread.

4. Binding between the surface and the Android media player is done.

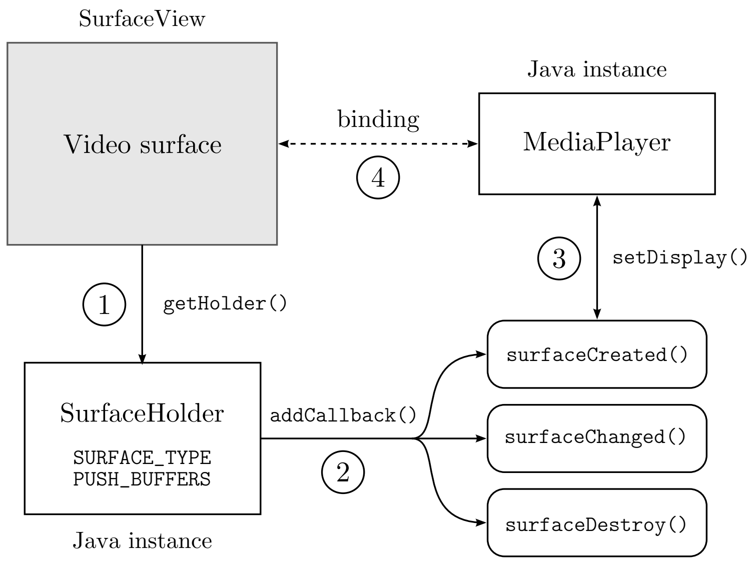


Figure 4.10: Surface view binding.

Listing 4.6: Listener launched when video surface is ready.

```
public void surfaceCreated(SurfaceHolder holder) {
   mediaPlayer.setDisplay(holder); // SurfaceHolder binding
   new Thread(new Runnable() { // Start segment handling as background task
      @Override
      public void run() {
         Looper.prepare();
         playHandler = new Handler();
         playHandler.post(nextSegment);
         Looper.loop();
      }
   }).start();
}
```

Different techniques were studied in order to load different video segments concurrently. By means of creating more than one instance of the `MediaPlayer` class, it may be possible to prepare several video segments at the same time. However, this approach is not suitable because of the *unique binding* condition, i.e., only one instance of `MediaPlayer` can be attached to a `surfaceHolder` (as depicted in figure 4.11). The Java method `setDisplay()` can only

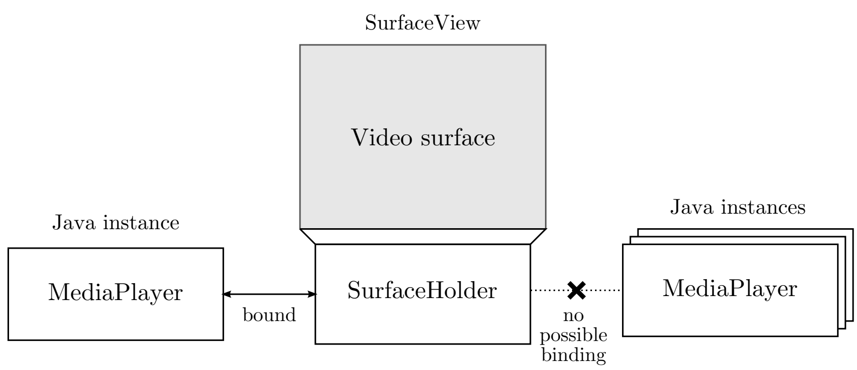


Figure 4.11: Binding problem.

be invoked once, and further calls are ignored. This makes it necessary to utilize another surface video for every `MediaPlayer`. Unfortunately, since `SurfaceView` is a heavy object and it consumes a significant amount of resources, this approach is not efficient. Therefore, in our implementation several instances of `MediaPlayer` are created but only one instance is attached to a `SurfaceView`.

4.4.4.2 Implementation

The listeners of the *Player activity* and the `MediaPlayer` class constitute the essential elements of this implementation. Listing 4.7 shows the most significant lines of the `onCreate()` Java method, here the `surfaceHolder` and `MediaPlayer` objects are instantiated.

Listing 4.7: Fragment of the activity's initialization method `onCreate()`.

```
surfaceHolder = surfaceView.getHolder(); // Create surfaceHolder and set listeners
surfaceHolder.addCallback(this);
surfaceHolder.setType(SurfaceHolder.SURFACE_TYPE_PUSH_BUFFERS);

mediaPlayer = new MediaPlayer(); // Create media player and set listeners
mediaPlayer.setOnPreparedListener(this);
mediaPlayer.setOnCompletionListener(this);
mediaPlayer.setOnErrorListener(this);
mediaPlayer.setScreenOnWhilePlaying(true);
```

Listing 4.8 shows the `handleNextSegment()` method. It is launched as a background task once the `surfaceView` resource is ready. This task manages the playback of media segments, by checking whether there are entries in the playlist. When there are new entries, the next appropriate media segment is load asynchronously, as shown in listing 4.9. There is one restriction imposed by the Android specification (as depicted in the state diagram of figure 4.12): the `MediaPlayer` object used by the activity must be restarted for every segment, since `setDataSource()` can only be invoked after `reset()`.

Listing 4.8: Procedure to handle the next media segment to be played.

```
private void handleNextSegment() {
   while (!playList.isReadyToPlay()) {} // Waiting for segments on playlist
   ... // Update UI elements (loading wheel)
   ... //Update segment pointers
   lastSegmentPath = currentSegmentPath;
   currentSegmentPath = playList.getNext(); // Read next entry

   if (currentSegmentPath != null) {
      setNextDataSource(currentSegmentPath); // Start segment load procedure
   } else {
      if (buffer.get404Errors() > MAX_404ERRORS)
         closeMedia("Many segments missing");
   }
}
```

Listing 4.9: Setting the next data source of the `MediaPlayer` instance.

```
private void setNextDataSource(String nextFilePath) {
   try {
      mediaPlayer.reset();
      mediaPlayer.setDataSource(nextFilePath);
      mediaPlayer.prepareAsync(); // Prepare segment asynchronously
   } catch (Exception e) {
      playHandler.post(nextSegment); // If errors: continue with next segment
   }
}
```

Once segments are successfully loaded, the `onPrepared()` listener is triggered, as illustrated in listing 4.10. The player proceeds to play the segment as soon as possible, subsequently launching (according to the `DELETION_TIMEOUT` parameter[9]) a background cleaning task. This task removes old entries in the representation lists and deletes already played segments.

[9] This timeout is set to avoid unnecessary consumption of CPU cycles within the segment's load interval. Any arbitrary value longer that a segment's load average loading time (500 ms) would be acceptable.

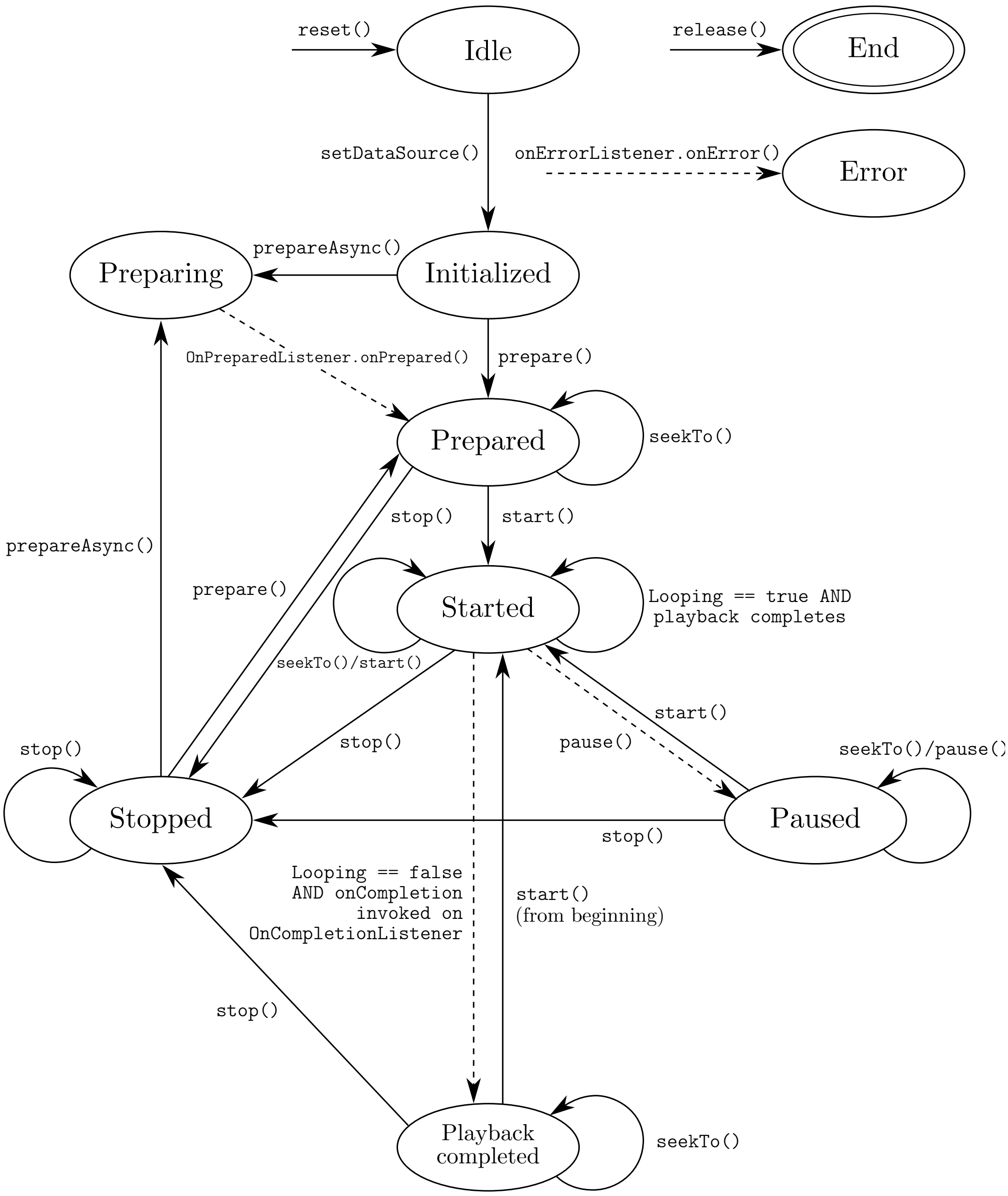


Figure 4.12: State diagram of Android's media player. Note that `setDataSource()` can only be called from the *Idle* status. Figure adapted from [8].

Listing 4.10: Listener triggered when the media segment is loaded.

```
public void onPrepared(MediaPlayer mp) {
   if (playedSegments == 0) Timer.startPlayback(); // Timing info
   mp.start();
   new Thread(new Runnable() {   // Update dynamic plots on screen in a new thread
      @Override
      public void run() {
         cleaningHandler.postDelayed(new Runnable() {
            @Override
            public void run() {
               ... // Perform RepresentationLists cleaning
               ... // Remove played segments from external storage
            }
         }, DELETION_TIMEOUT);
      }
   }).start();
}
```

A completion listener (`onCompletion()`) is invoked when segments have reached their end. Consequently, the next segment is immediately prepared in order to minimize interruptions in the media during playback. The code to do this is shown in listing 4.11.

Listing 4.11: Listener triggered after playback completion.

```
public void onCompletion(MediaPlayer mp) {
   ... // Logging
   playedSegments++;
   playHandler.post(nextSegment); // Launch background task
   ... // Update UI's buffer bar
}
```

Upon termination of the activity, background tasks and resources are released (using the method `closeMedia()` as shown in the simplified listing 4.12).

Listing 4.12: Termination of background tasks.

```
public void closeMedia() {
   if (mediaPlayer != null) mediaPlayer.release();
   if (buffer != null) buffer.stop(); // Stop buffering background task
   if (parser != null){ // Stop manifest updater task
      ...
      if (isLive())
         if (mpdHandler != null) mpdHandler.getLooper().quit();
   }
   finish(); // Terminate activity
}
```

4.4.5 Parser module

This module parses a file which follows either the MPEG-DASH standard or Apple's `m3u8` extended playlists. After completion, a list of available segments is generated for every representation, ordered by bandwidth (a basic example was illustrated in table 4.2). In addition, two parsing *modes* are defined:

Strict mode the parser checks that the manifest file follows all specifications, in particular mandatory attributes.

Non-strict mode the parser operates on a best effort basis. If there are some attributes missing, it utilizes with default values.

The player is responsible for calling the parser module at the start of the player's execution. If the manifest file is available and it satisfies the supported standards implemented in this prototype, an initial set of parameters is defined: number of representations, number of segments, type of content (on-demand or live) and segment duration, among other parameters (a full list of the supported parameters is given in the following section).

Table 4.2: Sample set of representation lists, assuming three quality levels (denoted by *bw1*, *bw2*, and *bw3*) and 10s-long segments.

Time	Segment URL
0.00	.../bw1/1.mp4
10.00	.../bw1/2.mp4
20.00	.../bw1/3.mp4
...	...

Time	Segment URL
0.00	.../bw2/1.mp4
10.00	.../bw2/2.mp4
20.00	.../bw2/3.mp4
...	...

Time	Segment URL
0.00	.../bw3/1.mp4
10.00	.../bw3/2.mp4
20.00	.../bw3/3.mp4
...	...

In the case of *live* content, this module is executed quasi-periodically (assuming that the segments are the same length). The manifest file is parsed again and the lists of segments are updated. The parsing procedure is aborted if the manifest file has not changed[10].

4.4.5.1 DOM and SAX

Java's Simple API for XML (SAX) and the Document Object Model (DOM) are two widely used parsing methodologies. In this design, SAX has been selected as the parsing technology for this module[11]. SAX demonstrates better capabilities for the type of files handled in the overall application. Advantages of SAX compared to DOM are:

- SAX parsing can be stopped at any line, at any time.
- SAX is better for large files because it consumes less memory than DOM.
- SAX takes less time than DOM to read the document.
- SAX is read only, whereas DOM can produce changes in the file.
- SAX parses sequentially, DOM can go backwards to the parent nodes.

In summary, SAX provides a faster method to parse XML files than DOM. Since these files basically consist of a set of parameters and list of segments, it can be read sequentially and there is no need for the client to modify any field. In fact, the server is responsible for providing new updates of the manifest file and is the *only* entity that updates this file.

4.4.5.2 Implementation

Listing 4.13 shows the Java constructor, which makes use of the `SAXParser`, `SAXParserFactory`, and `XMLReader` classes included in the Android framework. The main method is `parse()` (see listing 4.14) which will be invoked just once for on-demand services and periodically for live content. A `XMLHandler` private class (listing 4.15) reads all XML-tags and their attributes to generate Java objects and lists of segments for different representations. Tables 4.3 to 4.8 present the attributes of the MPD [1] supported in this implementation.

[10] Modifications of the manifest file are detecting by reading attributes which are meant to change, such as the time-stamps (available start time) or the last segment index.

[11] Specific parsers such as the Piccolo XML parser for Java (see `http://piccolo.sourceforge.net`) could have been chosen for this prototype. However, Android does not run a standard Java virtual machine, therefore customized parsers might not offer the same performance benefits when running on the Android virtual machine (Dalvik). Thus, using the DOM or SAX APIs ensures a better compatibility with any version of Android since they are officially supported.

Listing 4.13: DASH parser constructor.

```
public DASHParser(String manifestURL, Mode mode) throws IOException,
        UnvalidManifestException {
   ... // Init SAX variables
   try {
      SAXParser = factory.newSAXParser();
      XMLReader = SAXParser.getXMLReader();
      userXMLHandler = new UserXMLHandler();
      XMLReader.setContentHandler(userXMLHandler);
   } catch (...) {} // Exception handling
   initTempDirectory(); // Create temporal directory to save segments
}
```

Listing 4.14: Java method to parse an MPD file.

```
public void parse() throws IOException, SAXException,
        UnvalidManifestException {
   XMLReader.parse(parsingUrl);
   if (!isValidManifest())
      throw new UnvalidManifestException("Manifest is not valid");
   sortSegmentLists();
   this.segmentLists = getSegmentLists();
}
```

Listing 4.15: XMLHandler private class, it overrides SAX methods to parse supported MPD tags. It parses attributes and transforms them into Java objects and lists of segments.

```
private class UserXMLHandler extends DefaultHandler {
   ... // Override methods: startDocument(), endDocument() and endElement()
   @Override
   public void startElement(String uri, String localName, String qName,
         Attributes attributes) throws SAXException {
      /* Detect all supported tags and transform to Java objects */
      createMPD(attributes); // MPD tag
      createProgramInformation(attributes); //ProgramInformation tag
      createPeriod(attributes); // Period tag
      createSegmentInfoDefault(attributes); // SegmentInfo tag
      createRepresentation(attributes); // Representation tag
      createSegmentInfo(attributes); // SegmentInfo tag
      createURL(attributes); // URL tag
      createUrlTemplate(attributes); // URLTemplate tag
   }
}
```

Table 4.3: Supported attributes for the `MPD` tag in MPEG-DASH.

Attribute	Definition
Type	*Optional, on-demand by default.* Type of the media presentation. On-demand and live types are defined
Base URL	*Optional.* Base URL on MPD level
Minimum update period	*Mandatory.* Minimum period the MPD is updated on the server.
Minimum buffering time	*Mandatory.* Minimum amount of initially buffered media that is needed to ensure smooth playback.
Media presentation duration	*Optional.* Duration of the entire media presentation.
Availability start time	*Mandatory for live, optional for on-demand.* Start time of the first period of the media presentation in UTC format.
Available shifting time	*Optional.* Duration of the time shifting buffer that is available for a Live presentation. If it is present for on-demand services, the client should ignore this attribute.

Table 4.4: Supported attributes for the `Period` tag in MPEG-DASH.

Attribute	Definition
Start	*Optional.* Accurate start time of the period relative to the *availability start time* of the media Presentation.
Identifier	*Optional.* Unique identifier for this period within the media Presentation.
Default segment information	*Optional.* Default Segment information about Segment durations and, optionally, URL construction.

Table 4.5: Supported attributes for the `Representation` tag in MPEG-DASH.

Attribute	Definition
Identifier	*Mandatory.* Unique identifier for this representation within the period.
Bandwidth	*Mandatory.* Minimum bandwidth of a hypothetical constant bit-rate channel in bits per second over which the representation can be delivered such that a client, after buffering for exactly the *minimum buffering time* can be assured of having enough data for continuous playback.
MIME type	*Mandatory.* MIME type of the initialization segment, if present. If not, it provides the MIME type of the first media segment. This MIME type includes the CODEC parameters for all media types, including profile and level information where applicable.

Table 4.6: Supported attributes for the `SegmentInfo` tag in MPEG-DASH.

Attribute	Definition
Base URL	*Optional.* Base URL on representation level.
Segment duration	*Mandatory if duration is not specified on period level.* Constant approximate segment duration. All segments within this *segment information* element have the same duration unless it is the last segment within the period, which could be significantly shorter. If this attribute is not present, the value of this attribute is derived to be equal to the value of the duration attribute on period level, if present.
Start index	*Optional, 1 by default.* Index of the first accessible media segment in this representation.
URL template	*Optional.* The presence of this element specifies that a template construction process for media segments is applied. The element must include attributes to generate a segment list for the representation associated with this element.

Table 4.7: Supported attributes for the `URL` tag in MPEG-DASH.

Attribute	Definition
Source URL	*Optional.* URL of the media segment. If not present, then any *base URL* is mapped to the *sourceURL* attribute and the *range* attribute should be present.

Table 4.8: Supported attributes for the `URLTemplate` tag in MPEG-DASH.

Attribute	Definition
Source URL	*Optional.* The source string providing the template. If the template is not present, the *id* attribute on representation level provides the necessary information to construct the template.
End index	*Optional.* Index of the last accessible media segment in this representation.

The lexical representation of all `duration` attributes follows the W3C ISO 8601 Date and Time Formats syntax [15, section 3.2.6] *"P nY nM nD T nHnM nS"*, where *nY* represents the number of years, *nM* the number of months, *nD* the number of days, *T* is the date/time separator, nH *the number of hours*, *nM* the number of minutes and *nS* the number of seconds (decimal digits supported).

Additionally, extended M3U playlists can be parsed (see listing 4.16). Table 4.9 and table 4.10 show the tags of the Apple's extended M3U playlists [59] which are supported in the prototype.

Table 4.9: Supported tags for extended M3U playlists.

Tag	Definition
`#EXTM3U`	*Mandatory.* All playlists files must start with this tag. If not, the client must not attempt to use the playlist.
`#EXT-X-VERSION`	*Optional.* It specifies the protocol version. The client checks if it supports the version. if not, it must not attempt to use the playlist file.
`#EXT-X-TARGETDURATION`	*Mandatory.* It specifies the maximum media file duration. The `EXTINF` duration of each media file in the playlist file must be less than or equal to the target duration.
`#EXT-X-MEDIA-SEQUENCE`	*Optional.* It indicates the sequence number of the first URI that appears in a playlist file. This tag can only appear once.
`#EXTINF`	*Conditionally mandatory.* It is a marker which describes the media file identified by the URI that follows it. Each media file URI must be preceded by this tag.
`#EXT-X-STREAM-INF`	*Optional.* It indicates that the next URI in the playlist file identifies another playlist file (sub-playlist).

Table 4.10: Supported attributes for the `EXT-X-STREAM-INF` tag.

Attribute	Definition
`BANDWIDTH`	*Mandatory.* Decimal integer of bits per second. It must be an upper bound of the overall bit-rate of each media file, including container overhead.
`PROGRAM-ID`	*Optional.* Decimal integer that uniquely identifies a particular presentation within the scope of the playlist file. A playlist file may contain multiple `EXT-X-STREAM-INF` tags with the same `PROGRAM-ID` to identify different encodings of the same presentation. These variant playlists could contain additional `EXT-X- STREAM-INF` tags.
`CODECS`	*Optional.* Quoted string containing a comma-separated list of formats, where each format specifies a media sample type that is present in a media file in the playlist file. Valid format identifiers are those in the ISO File Format Name Space [34].
`RESOLUTION`	*Optional.* Decimal value describing the approximate encoded horizontal and vertical resolution of video within the stream.

Listing 4.16: Java method to parse an extended M3U playlist (`.m3u8`).

```java
public void parse() throws MalformedURLException, IOException,
        InvalidPlaylistException, UnmodifiedMPDException {

    String filename = downloadManifest(new URL(playlist));
    FileReader reader = new FileReader(filename);
    Scanner scanner = new Scanner(reader);
    ...
    while (scanner.hasNextLine()) {
        ... // Analysis of each extended tag
    }
    sortSegmentLists();
}
```

4.4.6 Segment-downloader module

The segment-downloader module is responsible for opening HTTP connections to fetch the available segments. It sends HTTP requests and waits for the corresponding reply, checking the HTTP headers and response codes in the reply. Once the connection is opened, the received byte stream is transferred to a *buffer* to store the media files (in this prototype, the cellphone's MicroSD card - a so-called external storage device - acts as the buffer[12]). Although the Android system natively supports HTTP progressive streaming, it is necessary to pre-download the segments in order to avoid pauses once playback has started. Immediately after a new segment is stored in the buffer, this module performs two actions:

1. It converts the media file into a supported format, if necessary. This occurs when Apple-HLS segments are contained in the MPEG-TS (`.ts`) format, as they cannot be directly played by Stagefright. This task will be performed by the *transcoder module*, further explained in section 4.1.1 on page 28.

2. It tests the media file using a background (*fake*) player, which is not bound to any `SurfaceView`. This fake player simply prepares the segment and triggers the `onPrepared()` and `onError()` listeners. If there is an error preparing a particular segment, then the module will try to download this segment one more time. If an error persists, the software will skip this segment. If the buffer is not empty, the next segment is played. Otherwise, a *buffering animated wheel* is shown to the user to indicate that the playback has paused.

Video fragments that are successfully downloaded are added to a *playlist*, as proposed in [69]. This list is represented as a table with two columns: estimated (relative) playback time and filename (*path*) to the media segments. A simple example is shown in table 4.11. The timing column is calculated based upon the information provided by the parsing module.

This module also provides a segment *re-download mechanism*. Segments whose download time increases greatly, must be immediately discarded. This is one of the indicators of an inadequate representation level and the *rate-adaptation module* must be notified, (the information that will be shared between the segment-downloader and the rate-adaptation modules is depicted in figure 4.13). Equation 4.3 indicates the maximum download time or *timeout* ($T_{timeout}$) where S_d represents the duration of the segment. In this prototype, the

[12] The reason to choose the external storage device as a buffer is rather simple: internal memory in Android cellphones is accessible and writable, but only plain-text files can be saved directly. Other file types such as video files can only be saved into the internal storage if they are transferred from the external storage, by means of the `openFileInput` and `openFileOutput` Java methods, therefore segments have to be saved into the external storage in any case. More information about data storage on Android devices can be found at `http://developer.android.com/guide/topics/data/data-storage.html`.

timeout value is fixed at 80% of the segment duration, which provides enough tolerance to enable a segment to be downloaded and potentially be downloaded a second time at a lower bit-rate.

$$T_{timeout} = 80\% \cdot S_d \tag{4.3}$$

Table 4.11: Playlist example.

Time	Path to segment
0.00	/media/sdcard/tmp/...segment1.mp4
10.00	/media/sdcard/tmp/...segment2.mp4
20.00	/media/sdcard/tmp/...segment3.mp4
...	...

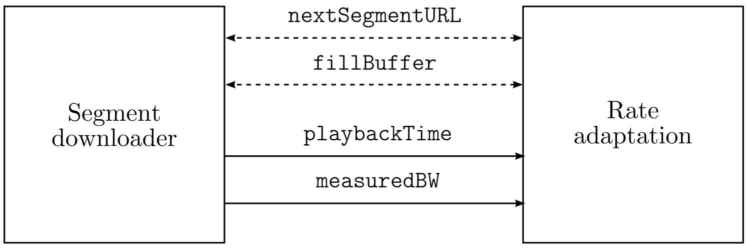


Figure 4.13: Communication between the segment-downloader and the rate-adaptation modules. Dashed lines represent requests whereas normal lines represent notifications.

A client's *throughput* metric (τ, measured in bits per second) has been defined to inform the rate-adaptation module right after every segment is downloaded of the recently experienced HTTP connection throughput. Throughput at the client's side is calculated as:

$$\tau = \frac{S_i}{T_i} \tag{4.4}$$

Where S_i is the size in bits of the segment i and T_i is the time it took to download it, measured in seconds.

4.4.6.1 Implementation

The segment-downloader module is implemented as a background task (i.e., a secondary thread executing as an infinite loop), as shown in the listing 4.17. First, a call to `fillBuffer()` in the `rateLogic` (rate adaptation module) determines whether a new segment should be downloaded to the buffer. If so, the destination URL will be requested by means of the `getNextSegment()` method. Two situations could arise: (1) there is already a new segment available to download or (2) there is not, in which case the loop continues in case of *live* content or ends in case of video *on-demand*. Note that this module only downloads segments given a URL. The decisions about the appropriate representation level are managed in the *rate-adaptation module*.

Listing 4.17: Main method of the segment-downloader module.

```
public void run() {
   while (running) {
      try {
         if (rateLogic.fillBuffer()) {
            URL url = rateLogic.getNextSegment();
            if (url != null) {
               try {
```

```
                  downloadSegment(url); // Fetch next segment
                  rateLogic.addEstimatedTime(); // Notify rate-adaptation module
                  storedSegments++;
                  playList.isReadyToPlay(); // Update playlist status
                  bufferBarUpdate();
               } catch (DownloadAgainException e) { ...
               } finally {
                  rateLogic.adapt(); // Perform adaptation in the rate-adaptation module
               }
            } else {
               if (playList.isLive()) continue; // Infinite loop for live content
               else break;
            }
         }
      } catch (...) {} // Catch exceptions
   }
   playList.setEnd();
}
```

If there is a new segment available, then this module calls the `downloadSegment()` method. The first step is the initialization of the HTTP connection, which is shown in listing 4.18. The code must check that the HTTP responses have correct headers and a valid response code. If there is an error, then the code attempts for a limited number of times (modeled as `MAX_ATTEMPS` and set to five in this implementation) to download the segment again - until a valid HTTP code is returned.

Listing 4.18: Opening a valid HTTP connection.

```
while (attempt < MAX_ATTEMPTS) {
   connection = (HttpURLConnection) url.openConnection();
   connection.setConnectTimeout(TIMEOUT);
   connection.setDoInput(true);
   connection.connect();

   int responseCode = connection.getResponseCode(); // Obtain HTTP response code
   if (responseCode == HttpURLConnection.HTTP_OK)
      break;
   if (responseCode == HttpURLConnection.HTTP_NOT_FOUND)
      throw new IOException("HTTP 404 not found");
   attempt++;
   connection.disconnect();
}
if (connection.getResponseCode() != HttpURLConnection.HTTP_OK)
   return; // Exit if HTTP response code is not HTTP-200
... // Read the Content-Length header field
connection = (HttpURLConnection) url.openConnection();
```

Listing 4.19 illustrates the management of the flow of bytes. The byte stream obtained from the HTTP connection (produced by an instance of `BufferedInputStream`) is redirected to the external storage (instance of `FileOutputStream`). In order to provide maximum timing accuracy, timing measurements are performed at the beginning of the byte transfers. A *timeout* (as explained above) is defined in this implementation as `MAX_ALLOWED_TIME` and is used as the criteria of when to **discard** and re-download segments that could not be retrieved in time.

Listing 4.19: Procedure to download a new media segment.

```
private void downloadSegment(URL url) throws ... {
   String filename = tempPath + getFileName(url.toString());
   if (!new File(filename).exists()) {
      ... // Init HTTP connection
      bis = new BufferedInputStream(connection.getInputStream(), BYTE_BLOCK);
      fos = new FileOutputStream(filename);
      ... // Start timer
      while ((count = bis.read(data, 0, data.length)) != -1) {
         ... // Stop downloading if player is closed
         if (Timer.getByteFlowTime() > MAX_ALLOWED_TIME) {
```

```
            if (rateLogic.shouldDownloadAgain()) {
               ... // Calculate availBW and notify Logic module
               throw new DownloadAgainException(...);
            }
         }
         fos.write(data, 0, count);
         bytes += count;
      }
      ... // Close streams
      try {
         availBW = (int) ((totalBytes * 8 * 1000) / Timer.endByteFlow());
         rateLogic.setAvailBandwidth(availBW);
      } catch (ArithmeticException e){}

      if (Player.isM3U8) // Perform MPEG-TS conversion if necessary
         filename = Transcoder.convertToMp4(filename);

      try {
         new FakePlayer(filename); // Test downloaded segment in FakePlayer
      } catch (DownloadAgainException e) {
         rateLogic.retryOnError();
         throw new DownloadAgainException(e.getMessage());
      }
      playList.add(filename); // Add segment to the playlist
   }
}
```

Once the media file has been correctly downloaded, the segment-downloader module provides a new bandwidth measurement (equation 5.5) to the rate-adaptation module. Notification is also performed when discarding segments, which forces the selection of the appropriate quality level before fetching another piece of content. Finally, the segment's path is added to the playlist, which is implemented as a Java `List<MediaSegment>`, where `MediaSegment` objects consist of a estimated playback time-stamp and a file path.

4.4.7 Rate adaptation module

This module can be considered the core of the adaptation logic. It receives information about the estimated playback time and the measured bandwidth from the segment-downloader module (as explained in section 4.4.6). In addition, the rate-adaptation module has access to the different playlists generated by the parsing module (see section 4.4.5). Based upon all of this information the rate-adaptation module runs the adaptation algorithm to decide which representation level is the optimal one. In this module, the aggressive, conservative, and mean algorithm (proposed in section 4.4.2) have been implemented.

4.4.7.1 Implementation

In terms of Java programming, the rate-adaptation module is a submodule of the segment-downloader, since they execute in the same thread. The algorithms proposed in the previous sections rely on actions that increase or decrease the representation level. Hence, four Java methods (listing 4.20) have been developed to provide this functionality. `switchUp()` and `switchDown()` receive a parameter indicating the number of level switching steps. The implementation of the switch-up method is shown in listing 4.21.

Listing 4.20: Java methods used to switch the representation level.

```
private void switchUp(int levels);
private void switchDown(int levels);
private void switchMaxUp();
private void switchMinDown();
```

Listing 4.21: Switch-up method. Increase the selected representation level. The number of steps to increase is passed as a parameter.

```
private void switchUp(int levels) {
   while (levels > 0 && !isMaxBandwidth) {
      try {
         /* Try to access the next representation list */
         segmentLists.get(currentId + levels);
         /* Success, update the pointers */
         currentId += levels;
         currentList = segmentLists.get(currentId);
         setBandwidths();
         break;
      } catch (IndexOutOfBoundsException e) {}
      levels--;
   }
}
```

The process of adaptation is performed after the download of every segment. The segment-downloader module calls the `adapt()` method within the rate-adaptation module. Listing 4.22 shows a simplified version of the Java code, only the *adaptive (aggressive)* algorithm is shown in this example. The remainder of the implementation follows the algorithms.

Listing 4.22: Adaptation Java method. It selects the appropriate representation level for each algorithm.

```
public void adapt() {
   switch (logic) { // Apply rate algorithm according to adaptive profile
   case PROGRESSIVE:
      ... // Null adaptation, testing purposes
   case ADAPTIVE_CONSERVATIVE:
      ... // Implementation of the conservative adaptive algorithm
   case ADAPTIVE_CONSERVATIVE:
      if (playList.isEmpty()) {
         switchMinDown();
         } else {
            int throughput = (int) (measuredThroughput * SENSITIVITY);
            if (throughput > currentBW) {
               while (nextBW < throughput && !isMaxBandwidth)
                  switchUp(1);
            } else {
               do {
                  switchDown(1);
               } while (prevBW > throughput && !isMinBandwidth);
            }
         }
         break;
   case ADAPTIVE_MEAN:
      ... // Implementation of the mean adaptive algorithm
   }
   ... // Prevent re-download a segment at the same quality twice
}
```

4.4.8 Transcoder module

Media segments are fetched and allocated in the Android device's external storage, as was previously explained in section 4.4.6. In case of the MPEG-DASH standard, media formats produced by the HTTP servers are fully supported in Stagefright, making the segments suitable for playback. However, the Apple-HLS specification contemplated only MPEG-TS as a media container format. Unfortunately, this container format is not natively supported in Stagefright, hence a transcoder is utilized to perform the necessary conversion.

The transcoder module solves the compatibility problem by providing additional processing for Apple's HLS media content. Fortunately, most Apple-HLS sources contains streams encoded with H.264 (baseline profile) and AAC, thus the transcoder only needs to change the MPEG-TS container into one of the Android's supported formats (listed in table 2.4 on page 21). Figure 4.14 illustrates the necessary transformation for compatibility. Since this procedure consumes a considerable amount of CPU cycles[13], it **must** be performed within a bounded period of time to avoid interrupting the playback (see experiments in section 5.7). As an absolute maximum upperbound, the *overall execution* of the operations performed in the buffering and transcoder modules may never take longer that the previous segment's duration, as expressed in equation 4.5.

$$T(i)_{download} + T(i)_{conversion} < T(i-1)_{duration} \tag{4.5}$$

Assuming that all segments are of the same length, equation 4.5 can be expressed as:

$$T_{download} + T_{conversion} < T_{duration} \tag{4.6}$$

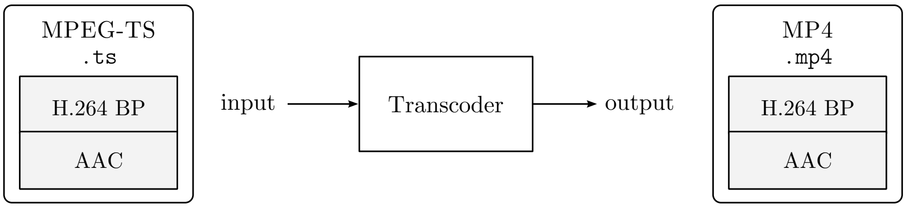


Figure 4.14: Media container conversion performed in the transcoder module. It provides compatibility with Apple-HLS.

4.4.8.1 Implementation

This module makes use of the FFmpeg audio and video libraries. Since they are written in the C and C++ programming languages, a binding is needed to invoke the proper functions from Android's Java standard code. This binding is achieved by means of the Java Native Interface (JNI). Fortunately a Native Development Kit (NDK)[14] is offered at the official Android developers site, which provides several tools to link Java code to pieces of native code. The basic Android application model does not change, since the NDK works in combination with the Android's SDK (introduced in section 2.10). The integration of the FFmpeg libraries can be found in the appendix C.

Listing 4.23 describes how exceptions can be thrown from the native code to the Java activities to notify them of an exception in the native code.

Listing 4.23: C function which throws a Java exception via JNI.

```c
int jniThrowException(JNIEnv* env, const char* className, const char* msg) {
    jclass exceptionClass = env->FindClass(className);
    if (exceptionClass == NULL)
        return -1;
    env->ThrowNew(exceptionClass, msg)
    return 0;
}
```

A set of native functions have been defined in a Java FFmpeg class: the most important of these are shown in listing 4.24. These methods provide a basic interface to prepare the conversion. The first two functions, `native_avcodec_register_all()` and `native_av_register_all()`, initialize all the media formats and CODECs that have been enabled when FFmpeg was

[13] External libraries (FFmpeg) perform the conversion of container format and require a significant amount of time smartphones' CPUs.

[14] Android NDK. Available from `http://developer.android.com/sdk/ndk`.

compiled. Input and output file names are set with `native_av_setInputFile()` and `native_av_setOutputFile()`, respectively. The output CODECs are set by calling the `native_av_setVideoCodec()` and the `native_av_setAudioCodec()` functions. In the FFmpeg notation, media streams can remain unaltered if copy is indicated as the video and audio CODECs, in this case changes will be made to the container format, which is the main purpose of this module in our current prototype.

Listing 4.24: A particular set of native functions defined in the `FFmpeg` class.

```
/* Codec initialization */
private native void native_avcodec_register_all();
private native void native_av_register_all();
private native void native_av_init()
      throws RuntimeException;
/* Input parameters */
private native FFmpegAVFormatContext native_av_setInputFile(String filePath)
      throws IOException;
private native FFmpegAVFormatContext native_av_setOutputFile(String filePath)
      throws IOException;
private native void native_av_setVideoCodec(String codec);
private native void native_av_setAudioCodec(String codec);
private native void native_av_parse_options(String[] args)
      throws RuntimeException;
/* Main conversion method */
private native void native_av_convert()
      throws RuntimeException;
/* Release resources */
private native int native_av_release(int code);
```

The initialization procedure is specified in listing 4.25, where the input options are allocated. Input parameters are parsed in the `FFmpeg_parseOptions()` native function, which has been simplified in listing 4.26.

Listing 4.25: Initialization native function.

```
static void FFmpeg_init(JNIEnv *env, jobject obj) {
    sObject = (*env)->NewGlobalRef(env, obj);
    jclass clazz = (*env)->GetObjectClass(env, obj);
    int i=0;
    for(i=0; i<AVMEDIA_TYPE_NB; i++){
        avcodec_opts[i]= avcodec_alloc_context2(i);
    }
    avformat_opts = avformat_alloc_context();
    sws_opts = sws_getContext(16, 16, 0, 16, 16, 0, sws_flags, NULL, NULL, NULL);
}
```

Listing 4.26: Parsing native function to obtain conversion parameters.

```
static void FFmpeg_parseOptions(JNIEnv *env, jobject obj, jobjectArray args) {
   ... // Init variables
   if (args != NULL) {
      argc = (*env)->GetArrayLength(env, args);
      argv = (char **) malloc(sizeof(char *) * argc);
      for(i=0;i<argc;i++) {
         jstring str = (jstring)(*env)->GetObjectArrayElement(env, args, i);
         argv[i] = (char *)(*env)->GetStringUTFChars(env, str, NULL);
      }
   }
   parse_options(argc, argv, options, opt_output_file); // Parse options
   /* Check input and output files */
   if(nb_output_files <= 0 && nb_input_files == 0)
      jniThrowException(env, ...);
   if (nb_output_files <= 0)
      jniThrowException(env, ...);
   if (nb_input_files == 0)
      jniThrowException(env, ...);
}
```

Finally, listing 4.27 illustrates `FFmpeg_transcode()`, where the fundamental FFmpeg function `av_transcode()` is invoked. Calling this function implies that all input parameters have been properly set without throwing any `JniException`.

Listing 4.27: Convert native function.

```
static void FFmpeg_transcode(JNIEnv *env, jobject obj, jlong outputFile,
      jlong inputFile) {
    ...
    if (av_transcode(output_files, nb_output_files, input_files, nb_input_files,
                     stream_maps, nb_stream_maps) < 0)
        jniThrowException(env, ...);
}
```

The Java method `convertTo()` (simplified in listing 4.28) wraps all the native functions previously explained, providing a simple *shortcut* to utilize the FFmpeg libraries from any Android activity. This procedure is depicted in figure 4.15. The whole operation can be simplified into three simple steps:

Listing 4.28: Java conversion method.

```
private static String convertTo(String inputFile, String extension)
         throws FFmpegException, RuntimeException, IOException {

    ffmpeg = new FFmpeg(); // Prepare conversion parameters
    ffmpeg.setVideoCodec("copy"); // Only conversion of the container format
    ffmpeg.setAudioCodec("copy");
    ffmpeg.init(inputFile, outputFile);
    ffmpeg.convert(); // Start format conversion
    return outputFile;
}
```

1. The Java `convertTo()` method is invoked after downloading a segment in the segment-downloader module (as was explained in section 4.4.6). The function `convertTo()` internally calls the native function `native_av_convert()`.
2. JNI launches the corresponding C function (`FFmpeg_transcode()`, listing 4.27) which makes use of the FFmpeg libraries.
3. The FFmpeg `av_transcode()` function is started. There are two possible outcomes:
    - Conversion was successfully performed, thus a new file has been created in the Android device's external storage and this file should be properly added to the *playlist*.
    - A failure occurred in some point in the conversion procedure (`av_transcode()` returned -1). In this case a `JniException` is thrown up to the Java layer, which is viewed as a `RuntimeException`.

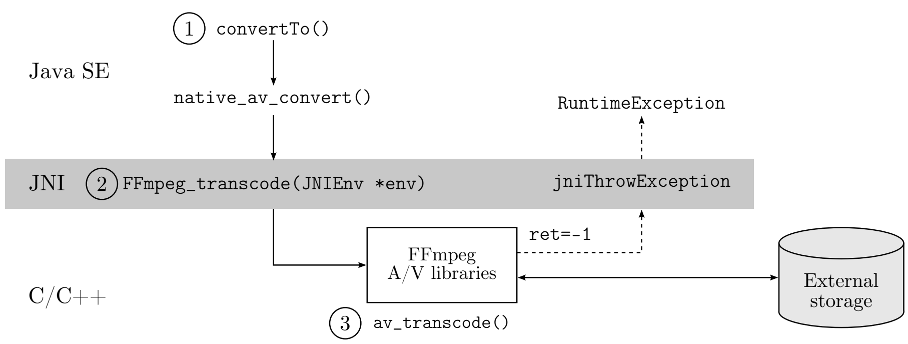


Figure 4.15: FFmpeg conversion interface.

4.4.9 Timer module

The timer module provides accurate timing information about multiple events that take place in the client application. The following measurements are essential for our evaluation of the system:

Start-up delay time it takes to start the playback.

Download delay time required to download a single media segment.

Absolute playback time starting playback time of a single segment, expressed in absolute time (based upon being synchronized with a NTP server).

Load time it takes to prepare the visualization of a single segment to be played until it is shown on the screen.

Parsing delay time it takes to receive and parse the whole playlist since it is requested.

Conversion delay time spend in transcoding or conversion operations (currently this is only applicable for Apple-HLS, as described in section 4.1.1).

Inactivity interval of time when the client is inactive, i.e., when all available segments have been fetched.

Pause time the player spends stopped during playback because the buffer is empty.

Session lifetime total duration of the session, i.e., the time since the system was started until it is shutdown.

The timer module is synchronized with a pool of NTP servers, as introduced in section 4.2. Table 4.12 summarizes the most significant SNTP parameters used in this implementation.

Table 4.12: NTP settings.

NTP server	NTP port	Version	Mode	NTP timeout (s)	NTP update (s)
`europe.pool.ntp.org`	123	3	3	15	30

4.4.9.1 Implementation

A `sntpClient` class has been created to request timing information from an NTP server. A NTP *datagram* is sent to the server and port specified in table 4.12, indicating the current system time in the `OriginateTimeStamp` header field. Calculation of the *offset* time between the cellphone's internal clock and the NTP server's clock is shown in listing 4.29.

Listing 4.29: Request NTP time.

```
private static boolean requestNTPTime() {
   if (ntpClient.requestTime(NTP_SERVER, NTP_TIMEOUT)) {
      /* Calculate offset */
      offsetNTP = ntpClient.getNtpTime() + SystemClock.elapsedRealtime()
            - ntpClient.getNtpTimeReference()
            - System.currentTimeMillis();
      return true;
   }
   return false;
}
```

The first NTP request is sent upon initialization of this module. Consequently, a background task is started to make future requests to the NTP servers (see listing 4.30). This module offers two types of timing metrics:

Absolute time calculated as the sum of the system's internal time and the NTP received offset.

Time intervals based strictly on the system's internal clock, computed as the difference between time-stamps.

Listing 4.30: NTP initialization.

```
public static void init() {
   time = System.currentTimeMillis(); // System's local clock
   ... // Request NTP time and calculate offset
   ... // Set all counters to zero
   new Thread(new Runnable() { // Init NTP update in a background task
      @Override
      public void run() {
         Looper.prepare();
         ntpHandler = new Handler();
         ntpHandler.postDelayed(NTPUpdate, NTP_UPDATE);
         Looper.loop();
      }
   }).start();
}
```

4.5 Network emulator

A prototype of the media player presented in section 4.4 has been developed to provide network adaptation. Rate adaptation implies that the client *reacts* in a reasonable time to changes in the network conditions.

4.5.1 Emulator requisites

In order to evaluate the capabilities of the application (as will be done in chapter 5), a network simulator was interposed between the client and the server to control the characteristics of the HTTP traffic. This emulator must do at least the following: (1) perform bandwidth limitation (bidirectional, and optionally, unidirectional), (2) produce packet loss with a given probability or error rate, and (3) induce appropriate packet delays.

Dummynet [20, 21] has been selected as the evaluation tool since it satisfies the above requirements and it is easily integrated in the system[15]. Dummynet's is briefly explained in the following section, along with the functions that will be used during the evaluation (see the *network scenarios* defined in section 5.1.7). There are multiple alternative emulation tools available, but they present some integration difficulties:

- **WANem**[16] is a Wide Area Network (WAN) emulator, which provides several tools to perform experiments in a Local Area Network (LAN) environment. WANem can be used to emulate different WAN conditions such as packet loss, packet re-ordering, or jitter. However, WANem has two disadvantages for integration in our test environment: (a) WANem is intended to be deployed with *ethernet* interfaces, but the communication between client and server is performed over a wireless interface) and (b) unfortunately, WANem is only offered as a Gnu/Linux (Knoppix) live distribution, which makes scheduled scripting more difficult, since it requires a separate machine running the emulator.
- **Trickle**[17] is a Gnu/Linux bandwidth shaper which limits traffic over a socket. The `trickle` command simply precedes any other command in order to limit bandwidth in

[15] A second alternative would have been **tc**. Tc is a Linux kernel tool which can be used to configure network traffic with similar functionality as Dummynet. A comparison study [58] determined that both tc and Dummynet equally emulated bit-rates up to 500 Mb/s.

[16] WANem. Available from: `http://wanem.sourceforge.net`.

[17] Trickle. Available from: `http://monkey.org/~marius/pages/?page=trickle`.

this application. An example of this would be: `trickle [parameters] application`. Since both on-demand and live server (described in sections 4.3.1 and section 4.3.2) have been deployed as Gnu/Linux *daemons*, `trickle` cannot be easily invoked several times during the servers' execution[18]. Trickle does not support modification of the emulation parameters on-the-fly, therefore servers will have to be restarted for every variation in the emulation scenario. Since the live server requires a considerable[19] time to inspect all segments during its initialization, it should not be restarted for every variation in the intended emulation.

4.5.2 Dummynet

Dummynet is the network emulator deployed for our evaluation. For simplicity, Dummynet was installed on the same Gnu/Linux machine as the HTTP server. Dummynet intercepts the packets within the protocol stack, as shown in figure 4.16. These packets are selected according to different rules, which have been previously specified by means of the `ipfw` shell command.

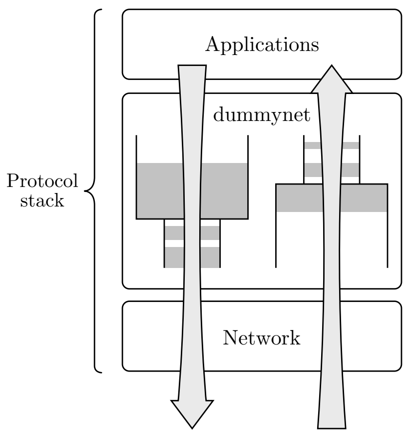


Figure 4.16: Dummynet introduces one or more pipes and queues in the protocol stack. Packets are intercepted and delayed according to the set of network rules. Adapted from [21, figure 3].

The intercepted packets are passed through one or more queues and pipes (as depicted in figure 4.17) which emulate propagation delays, limitations of available bandwidth, and packet loss. Pipes behave as fixed-bandwidth channels and queues can be weight-assigned.

Network rules can be easily defined. These rules establish a *networking profile*. Several profiles were proposed for the experiments and measurements described in the next chapter. For example, listing 4.31 shows how to define a simple pipe rule that restricts the available bandwidth to 2 Mb/s and introduces a packet delay of 1000 ms. Listing 4.32 shows a more complex example, where both *outcoming* and *incoming* traffic have different bit-rates, packet delay, and packet loss specifications.

Listing 4.31: A simple pipe created in Dummynet. The pipe bidirectionally limits the network traffic to 2 Mb/s and induces a packet delay of 1000 ms.

```
# Define bandwidth and delay of the emulated link
ipfw pipe 1 config bw 2Mbit/s delay 1000ms
# Pass all traffic through the emulator
ipfw add pipe 1 ip from any to any
```

[18] Gnu/Linux daemons are intended to be as user-independent as possible. Daemons are executed in background and typically provide a reduced set of commands such as `start`, `stop`, `status`, and `restart`.

[19] The live servers analyzes all the media segments encoded at different bit-rates (for all the media clips), in order to detect missing segments (i.e., removed from the system for any reason) and generate manifest files accordingly. The manifest file is not available until this procedure has finished.

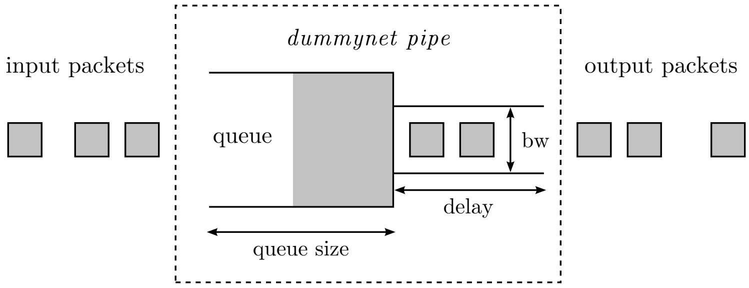


Figure 4.17: Pipes and queues, the basic elements of Dummynet. Adapted from [20, figure 3].

Listing 4.32: A more complete example of pipes in Dummynet. Incoming network traffic is limited to 512 kb/s with a delay of 32 ms and 1% probability of error; whereas the outcoming traffic is limited to 256 kb/s, 100 ms of packet delay, and 5% of packet loss.

```
# Define pipes for incoming and outcoming network traffic
ipfw add pipe 3 out
ipfw add pipe 4 in
# Pipes configuration
ipfw pipe 3 config bw 512Kbit/s delay 32ms plr 0.01
ipfw pipe 4 config bw 256Kbit/s delay 100ms plr 0.05
```

Chapter 5

Evaluation

"Premature optimization is the root of all evil."
– Donald Knuth

In this chapter a number of potential experiments are proposed to evaluate the performance of the adaptive mechanisms presented in section 4.4.7.

The evaluation environment is introduced in section 5.1, including the segmentation schemas (defined in section 5.1.3), the input/output characterization (section 5.1.5), the metrics proposed (section 5.1.6), and the network scenarios emulated (section 5.1.7). Results of the experiments are described in section 5.2 to section 5.7.

5.1 Experimental environment

Figure 5.1 illustrates the evaluation environment. Three components of the architecture have been provided with input parameters:

1. The client's application provides several rate mechanisms: the *aggressive*, *conservative*, and *mean* algorithms (as introduced in section 4.4.7).
2. A set of scenarios (explained in detail in section 5.1.7) are emulated in the underlying network between server and client.
3. The preparation of the content follows different *segmentation schemas* (section 5.1.3). In particular, schemas might differ in media formats, CODECs, and duration of the segments.

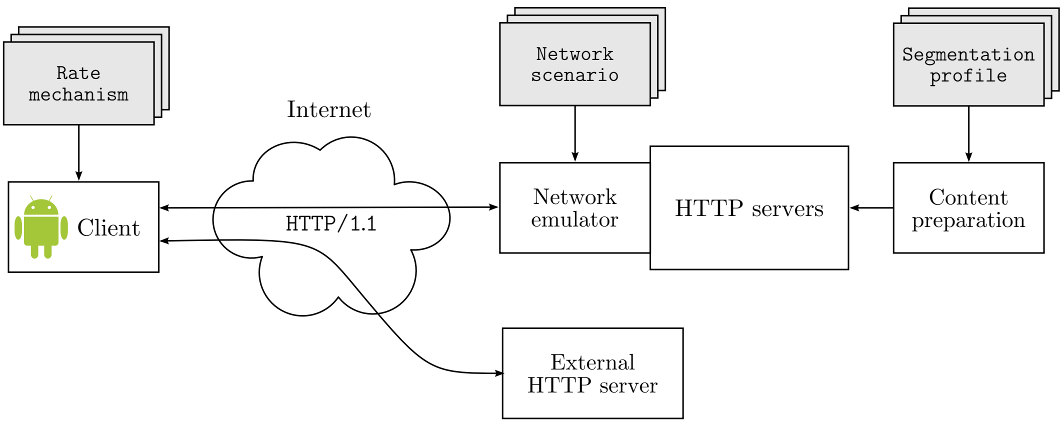


Figure 5.1: Evaluation environment with three different parametrized components (shown in gray).

Additionally, a real-world scenario has been included in this evaluation. This scenario exploits the fact that several TV channels following the Apple-HLS standard are freely available on the Internet (see section 5.7).

5.1.1 Experimental devices

Table 5.1 summarizes the specifications of the devices utilized on the experiments. The same machine (a standalone netbook) acts as HTTP server and prepares the content (as explained in section 4.1 and section 4.3). The client is a mid-range smartphone running Android 2.2.1.

Table 5.1: Specifications of devices employed in the experiments. Extracted from [63].

	Client	Server
Name	Samsung Galaxy Ace S5830	Acer Aspire One D150
CPU	800 MHz ARM 11 processor	1.60 GHz Intel Atom N270
Memory	RAM: 256 MB Internal: 158 MB External: 2GB microSD	RAM: 1 GB DDR3 @ 667 MHz Storage: 160 GB SATA HDD
OS	Android v2.2.1 (Froyo)	Kubuntu Gnu/Linux 2.6.36
Network	2G: GSM 850/900/1800/1900 3G: HSDPA 900/2100 WLAN: Wi-Fi 802.11 b/g/n	WLAN: Wi-Fi 802.11 b/g/n
Display	TFT HVGA touchscreen. 480 × 320 px	-

5.1.2 Content source

The 3D animation film *Sintel* [16] has been selected for this evaluation, since is free-distributable under the Creative Commons Attribution (CC-A) license. Sintel is a 15-minute movie produced by the Blender Foundation and created entirely with open source software. It was released at the Netherlands Film Festival in September 2010.

5.1.3 Segmentation schemas

The media content which is pushed to the HTTP servers can be offered in multiple ways. Our evaluation focus on two essential aspects: the number of quality levels (R) and the duration of the segments (S_d). Table 5.2 defines the different segmentation schemas used in our evaluation.

Each schema provides a different media segment duration S_d, from 5 s to 15 s. The *reference time* has been set to 10 s, as this duration was recommended by Zambelli [74] (see also table 2.6 on page 23). For simplicity, a homogeneous but sufficient [49, 50] number of representations has been provided in all schemas. Selection of the different levels of quality is explained in section 5.1.4.

Table 5.2: Segmentation schemas.

	Schema 1	Schema 2	Schema 3	Schema 4
S_d (s)	5	8	10	15
R	10	10	10	10

5.1.4 Selection of media quality levels

Table 5.3 illustrates the Android's official encoding recommendations for low and high quality media content. For the sake of simplicity, in this evaluation the *frame rate*, *audio channels*, *sampling rate*, *GOP size*[1], and *resolution* are set to typical values (as shown in table 5.4), whereas video and audio bit-rates are part of our design choices. Both audio and video bit-rates are increased in every media representation that we have conducted experiments with.

Table 5.3: Official Android's encoding recommendations for low and high quality video. Extracted from the Android's developer site [6].

Parameter	Low quality video	High quality video
Video CODEC	H.264 Baseline Profile	H.264 Baseline Profile
Video resolution (px)	176 × 144	480 × 360
Frame rate (fps)	12	30
Video bit-rate (kb/s)	56	500
Audio CODEC	AAC-LC	AAC-LC
Audio channels	1 (mono)	2 (stereo)
Audio bit-rate (kb/s)	24	128

Table 5.4: Set of fixed parameters used in all representations.

Container format	CODECs	Frame rate (fps)	GOP size	Resolution (px)
MP4 (`.mp4`)	H.264 BP + AAC	25	25	480 × 320

Table 5.5 defines the quality levels which will be provided to the HTTP servers, with increasing bit-rates from 80 kb/s to 2 Mb/s. In particular, these representations have been selected to supply a higher density of levels for lower bit-rates, since this prototype is intended for mobile networks that may have more limited bandwidth. Therefore, the difference in rates between levels is not uniform. This selection of media bit-rates leads to potentially large changes in quality levels by the rate adaptation algorithms.

Table 5.5: Set of media representation levels generated on the streaming server.

Level (r_i)	1	2	3	4	5	6	7	8	9	10
Video bit-rate (kb/s)	56	70	100	150	200	400	700	900	1400	1900
Audio bit-rate (kb/s)	24	24	44	64	94	94	94	94	94	94
Audio channels	1	1	1	1	2	2	2	2	2	2
Sampling rate (kHz)	22.05	22.05	22.05	22.05	44.1	44.1	44.1	44.1	44.1	44.1
Avg. bit-rate (kb/s)	80	100	150	200	300	500	800	1000	1500	2000

As an example of the video streams, figure A.6 shows a set of frames from the same clip encoded at different video/audio bit-rates. For simplicity, only four quality levels are represented: 80 kb/s (first representation), 150 kb/s, 800 kb/s, and 2000 kb/s (last representation). Higher rates assign more bits per image, resulting in a better quality.

[1] A fixed GOP setting in the encoder simplifies the stream switching between video levels.

(a) 80 kb/s **(b)** 150 kb/s

(c) 800 kb/s **(d)** 2000 kb/s

Figure 5.2: Frame samples from *Sintel* encoded at different bit-rates.

5.1.5 Input and output characterization

Table 5.6 and table 5.7 summarize, respectively, the input and output parameters that have been considered in our evaluation.

Table 5.6: Input parameters.

Notation	Unit	Definition
T	s	Total session time
T_{shift}	s	Available shifting time on server
$T_{min-buf}$	s	Minimum buffering time, specified on server
S_d	s	Segment duration
S	segments	Total number of segments in session. Calculated as $S = T/S_d$
$T_{timeout}$	s	Segment's downloading timeout. $T_{timeout} = S_d \cdot 80\%$
R	levels	Total number of representations offered on the server
r_i	b/s	Ordered representation levels by bit-rates, where $1 < i < R$. The last representation, r_R represents the highest quality level offered by the server
$bw_{avail}(t)$	b/s	Available bandwidth emulated during the session.
$\widetilde{bw}_{avail}(t)$	b/s	Maximum available bandwidth
p_e	-	Probability of error

The **available bandwidth** ($bw_{avail}(t)$) represents the bit-rate offered at the server's side over the session time as emulated by Dummynet. The available bandwidth is a function defined from $t = 0$ s to $t = T$.

Table 5.7: Output parameters.

Notation	Unit	Definition
T_{buf}	s	Total buffering time, i.e., total time the playback is interrupted due to buffer underrun during the session
$T_{start-up}$	s	Start-up time
$T_{inactive}$	s	Total inactive time, i.e., the client is not downloading segments
S_{missed}	segments	Total number of missed segments due to HTTP-404 responses
S_{retry}	segments	Total number of retried segments, i.e., segments which are discarded to be re-downloaded at another quality level
$\tau(t)$	b/s	Throughput, i.e., measured link bit-rate on client's side
$b(t)$	b/s	Media bit-rate selected on client's side
$s_{client}(t)$	s	Segments' actual playback on client (absolute time-stamps synchronized with NTP)
$s_{server}(t)$	s	Server's segment availability (absolute time-stamps synchronized with NTP)
$c_{state}(t)$	-	Client's state function

The **maximum available bandwidth** ($\widetilde{bw}_{avail}(t)$) is defined as the available bandwidth at the highest representation bit-rate offered by the server (r_R). In this evaluation, truncation is produced at $r_R = 2$ Mb/s (corresponding to representation number 10 in table 5.5). The maximum available bandwidth is a function defined from $t = 0$ s to $t = T$. Figure 5.3 depicts an example of the $bw_{avail}(t)$ and the $\widetilde{bw}_{avail}(t)$ functions.

$$\widetilde{bw}_{avail}(t) = bw_{avail}(t) \quad 0 < \widetilde{bw}_{avail}(t) < r_R \tag{5.1}$$

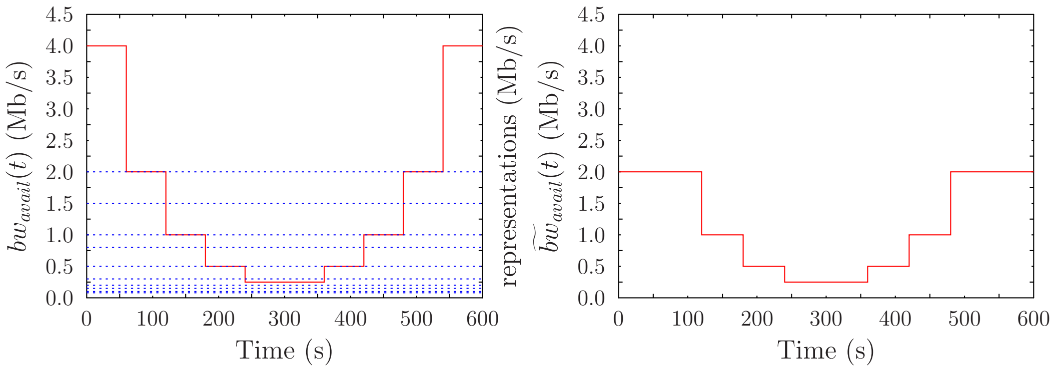


Figure 5.3: Example of a available bandwidth function ($bw_{avail}(t)$) and the representation levels depicted in the left hand figure. The maximum available bandwidth function ($\widetilde{bw}_{avail}(t)$) is depicted on the right hand figure, truncated at the maximum representation level ($r_R = 2$ Mb/s)

The **throughput** function ($\tau(t)$) represents the measured link bit-rate on the client's side. The throughput is a stepwise function computed over the individual throughput of every segment (τ_i). This is calculated as:

$$\tau_i = \frac{S_i}{T_i}$$

Where S_i is the size in bits of the segment i and T_i the time the client spends downloading the segment i. The throughput function is constructed as follows:

$$\tau(t) = \begin{cases} 0 & 0 < t < t_{b1} \\ \tau_1 = \dfrac{s_1}{t_{b1} - t_{a1}} & t_{b1} < t < t_{b2} \\ \tau_2 = \dfrac{s_2}{t_{b2} - t_{a2}} & t_{b2} < t < t_{b3} \\ \ldots & \ldots \end{cases}$$

Where $s_1, s_2, \ldots$ are the sizes of the downloaded segments and $t_{a1}, t_{a2}, \ldots$ and $t_{b1}, t_{b2} \ldots$ are respectively the starting and ending time-stamps which determine the downloading time for each segment.

The **selected media bit-rate** ($b(t)$) is a stepwise function which records the representation level selected by the client for the segments that have been *successfully* downloaded. Note that, the representation level of the segments that are discarded (to be re-downloaded at a lower quality level) are not considered in $b(t)$. The selected media bit-rate is the result of the decisions taken by the adaptation mechanisms (introduced in the previous chapter, section 4.4.7). These mechanisms use the throughput ($\tau(t)$) as the only metric to produce adaptation.

5.1.6 Metrics

Table 5.8 enumerates the metrics defined for our evaluation. All of these metrics are defined using a common reference (i.e., the session time, T), to ease comparison regardless of the experiments' session time. These metrics are calculated over all the data accumulated over the session. Thus, the disparity of the results obtained by running the same experiment several times is minimized. The following subsections explain in detail how these metrics are computed.

Table 5.8: The metrics that will be used for our evaluation.

Notation	Unit	Definition
$u(t)$	%	Bandwidth utilization
$e(t)$	s	End-to-end latency
ε_{bw}	%	Bandwidth utilization efficiency
ε_{buf}	%	Buffering efficiency
ε_{fetch}	%	Segment fetch efficiency
ε_{retry}	%	Segment-retry efficiency
ε_{active}	%	Active efficiency
$\varepsilon_{start-up}$	%	Start-up efficiency
ε_{up}	%	Reaction (switching-up) efficiency
ε_{down}	%	Reaction (switching-down) efficiency

5.1.6.1 Weighted functions

A pair of weighted functions (depicted in figure 5.4 have been proposed to define some of the metrics of this evaluation. In particular, these functions are used for the active efficiency (section 5.1.6.8), the start-up efficiency (section 5.1.6.9), and the reaction efficiency (section 5.1.6.10).

The first weighting function ($w_{long}(t)$) is a linear, monotonically decreasing function which assigns higher weights at the beginning than at the end of session T. The function $w_{long}(t)$ is defined to provide weight to metrics which involve measurements during the whole session time (T).

$$w_{long}(t) = 1 - \frac{t}{T} \quad 0 < t < T$$

$w_{short}(t)$ is a non-linear function which assigns higher weights given small time values. $w_{short}(t)$ decreases much faster than $w_{long}(t)$, such that a delay of 20 s (the maximum segment duration in the experiments) is assigned[2] a weight of 50% ($w_{short}(20) = 0.5$). As a result, $w_{short}(t)$ provides higher weight to metrics which measure short delays or intervals shorter than the session time (T), such as reaction times.

$$w_{short}(t) = \frac{1}{\frac{t}{20} + 1} \quad 0 < t < T$$

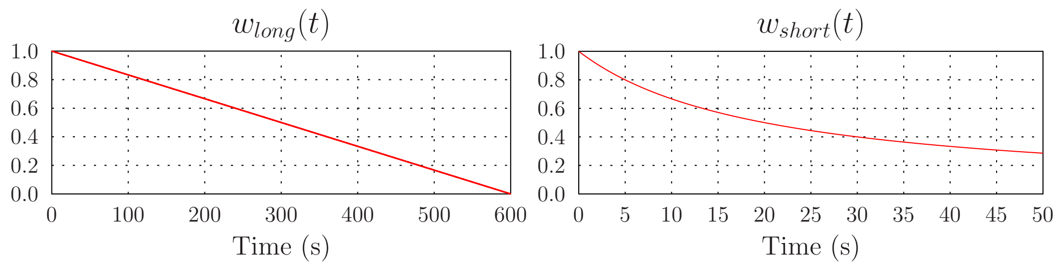


Figure 5.4: Weighted functions. $w_{long}(t)$ weights metrics over the whole session T, whereas $w_{short}(t)$ is more suitable to weight delays and short intervals.

5.1.6.2 Bandwidth utilization

The bandwidth utilization function ($u(t)$) is defined as the selected bandwidth of the client ($b(t)$) normalized by the maximum available bandwidth ($\widetilde{bw}_{avail}$). Thus, given a time t, this function compares the media bit-rate selected on the client's side compared to the maximum available bit-rate in the network. Values of $u(t)$ lower than 1 represent an *underestimation* of the maximum available bandwidth, whereas values greater than 1 denote an *overuse* of the available bandwidth.

$$u(t) = \frac{b(t)}{\widetilde{bw}_{avail}(t)} \tag{5.2}$$

5.1.6.3 Bandwidth efficiency

The bandwidth utilization efficiency (ε_{bw}) is defined as the integral of the bandwidth utilization ($u(t)$), normalized to the session time T. As a result ε_{bw} represents the efficacy of the bandwidth utilization throughout the whole session. This coefficient is calculated as follows:

First, the bandwidth utilization function ($u(t)$) is separated into the sum of two sub-functions $u_{under}(t)$ and $u_{over}(t)$:

$$u(t) = u_{under}(t) + u_{over}(t)$$

[2] Many other conditions could have been chosen arbitrarily. For our evaluation, the condition for $w_{short}(t)$ is defined based on the largest value of S_d considered in our experiments, i.e., 20 s.

Where $u_{under}(t)$ and $u_{over}(t)$ contain respectively the values of $u(t)$ that are lower and greater than 1:

$$u_{under}(t) = \begin{cases} u(t) & u(t) < 1 \\ 0 & u(t) > 1 \end{cases}$$

$$u_{over}(t) = \begin{cases} 0 & u(t) < 1 \\ u(t) & u(t) > 1 \end{cases}$$

Given these two functions, the bandwidth utilization efficiency is calculated as follows:

$$\varepsilon_{bw} = \frac{1}{T}\left(\int_T u_{under}(t)dt + \int_T \frac{1}{u_{over}(t)}dt\right) \tag{5.3}$$

Note that the second integral is calculated using the inverse of $u_over(t)$ since all values of $u_over(t)$ are all greater than 1. The inverse is utilized to correlate ε_{bw} with the underestimation and overestimation of the available bandwidth.

5.1.6.4 Buffering efficiency

The buffering efficiency (ε_{buf}) measures the total playback time over the session. Note that in our evaluation it is assumed that there is always playable content during the experiments, therefore the player can be only switched between two states: *buffering* or *playing*. ε_{buf} is calculated using the cumulative buffering time (T_{buf}), i.e., the time the playback has been interrupted due to *buffer underrun* as indicated in equation 5.4.

$$\varepsilon_{buf} = 1 - \frac{T_{buf}}{T} \tag{5.4}$$

5.1.6.5 Segment-fetch efficiency

The segment-fetch efficiency (ε_{fetch}) is the ratio of segments successfully downloaded ($S - S_{missed}$) over the total number of segments that should have been in the session (S). It provides a comparison between the number of segments that have been successfully fetched and the missed segments due to `HTTP-404` responses.

$$\varepsilon_{fetch} = 1 - \frac{S_{missed}}{S} \tag{5.5}$$

5.1.6.6 Segment-retry efficiency

The segment-retry efficiency (ε_{retry}) represents the proportion of the time consumed when the client re-downloads segments at different qualities, compared to the session time (T). The consumed time is calculated as the number of re-attempts (S_{retry}) multiplied by the maximum time to download a segment ($T_{timeout}$). Note that $T_{timeout}$ is defined as the 80% of the segment duration (S_d).

$$\varepsilon_{retry} = 1 - \frac{S_{retry} \cdot T_{timeout}}{T} \tag{5.6}$$

5.1.6.7 End-to-end latency

The end-to-end latency ($e(t)$) indicates the delay from when a new segment is available on the server's side until the client actually plays the segment (**not** when the segment is stored in buffer). $e(t)$ is an stepwise function calculated as follows:

$$e(t_i) = s_{client}(t_i) - s_{server}(t_i) \tag{5.7}$$

Where $e(t_i)$ is the end-to-end latency of the segment i, $s_{client}(t_i)$ represents the time-stamp of the actual playback of segment i on the client's side, and $s_{server}(t_i)$ is the time-stamp when segment i is produced on the server's side. Time-stamps are synchronized with a NTP server (as explained in section 4.2).

5.1.6.8 Active efficiency

The active efficiency (ε_{active}) provides a weighted sum of the total time the client is in the *active* state, normalized by the total active time during the session. ε_{active} is calculated as:

$$\varepsilon_{active} = 1 - \frac{1}{T_{inactive}} \int_T c_{state}(t) \cdot w_{long}(t) dt \tag{5.8}$$

Where $w_{long}(t)$ is a weighting function (defined in section 5.1.6.1) and $c_{state}(t)$ denotes the stepwise *state function*:

$$c_{state}(t) = \begin{cases} 1 & \text{inactive} \\ 0 & \text{active} \end{cases}$$

5.1.6.9 Start-up efficiency

The start-up efficiency ($\varepsilon_{start-up}$) is defined as the weighted start-up time, that is, the $w_{short}(t)$ weighted function is applied to the time it takes to start playback ($T_{start-up}$). $T_{start-up}$ is defined as the interval since the begin of the experiment ($t = 0$ s) until the first media segment starts to play.

$$\varepsilon_{start-up} = w_{short}(T_{start-up}) \tag{5.9}$$

5.1.6.10 Reaction efficiency

Figure 5.5 illustrates the criteria employed to calculate the reaction times in our evaluation. Two types of intervals are defined:

1. The switching-up reaction interval (T_{up}) is the elapsed time since a bit-rate increase occured in the network ($bw_{avail}(t)$) until the client *increments* the quality level ($b(t)$). Note that further switching operations might be performed; however, they are not considered in the T_{up} interval.
2. The switching-down reaction interval (T_{down}) is defined similarly to T_{up}, in case of bit-rate reductions in $bw_{avail}(t)$.

Thus, the reaction efficiencies for switching-up and switching-down (ε_{up}, ε_{down}) are defined as T_{up} and T_{down} weighted by the $w_{short}(t)$ function.

$$\varepsilon_{up} = w_{short}(T_{up}) \tag{5.10}$$

$$\varepsilon_{down} = w_{short}(T_{down}) \tag{5.11}$$

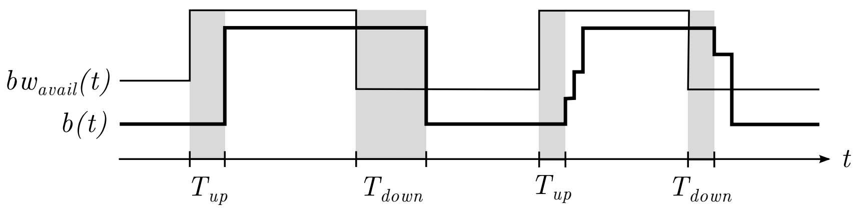


Figure 5.5: Reaction times.

5.1.7 Network scenarios

Figure 5.6 depicts the network scenarios considered in our evaluation. These scenarios provide persistent and non-persistent bandwidth fluctuations. The session time for all the experiments was defined to be sufficiently long enough to determine the long-term behaviour of the system: an experimental run of 10 minutes ($T = 600$ s) prevents bias of the results. Considering the longest segment duration evaluated on this scenarios ($S_d = 20$ s), the session duration is 600 ± 20 s ($\pm 3.33\%$), thus providing a confidence of 96.67%. For all scenarios, two families of experiments were performed:

1. Performance of the three adaptive mechanisms during the session. The duration of segments is fixed to 10 seconds ($S_d = 10$ s) in this set of experiments.
2. Performance of the conservative mechanism with different segment durations (5 s, 10 s, and 20 s).

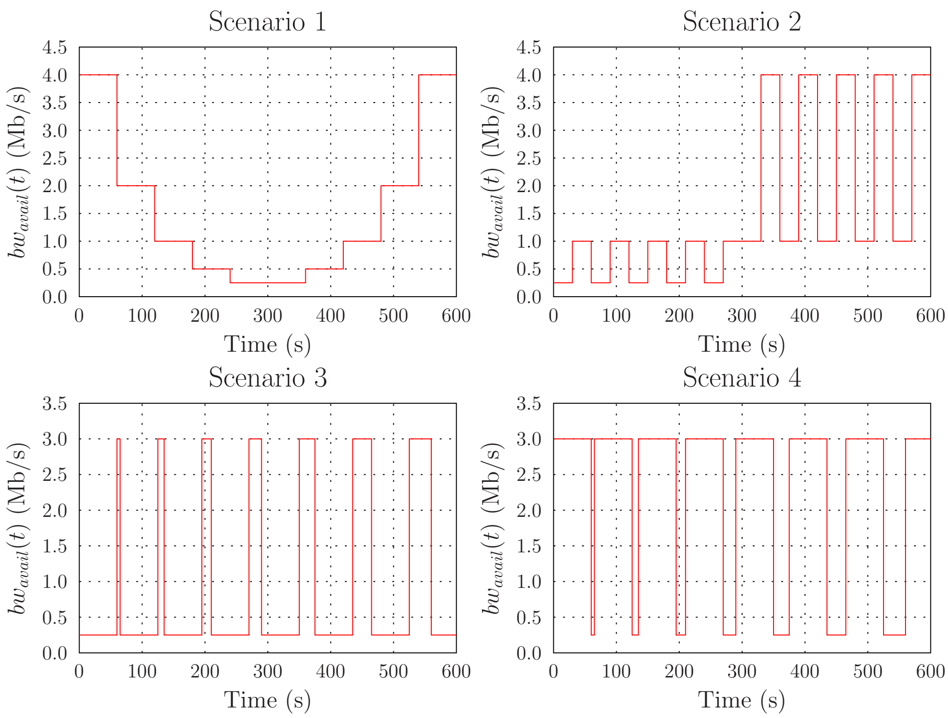


Figure 5.6: Network scenarios emulated during the evaluation. All of them produce variations in the available bandwidth.

5.2 Scenario 1: long-term variations of the available bandwidth

The first network scenario (depicted in figure 5.7) produces long-term variations of the available bandwidth. The scenario has two parts: from $t = 0$ s to $t = 300$ s, bit-rates are decreased at intervals of one minute from 4 Mb/s to 250 kb/s. In the second part, bit-rates are increased from 250 kb/s to 4 Mb/s at one minute intervals.

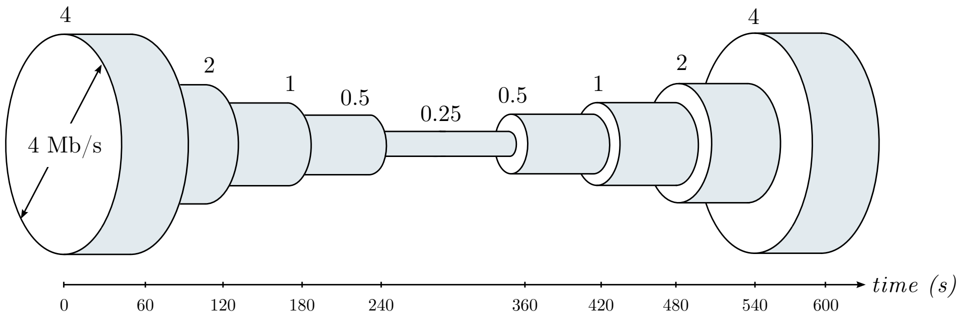


Figure 5.7: Series of *Dummynet* pipes for the first network scenario.

This network scenario evaluates the behaviour of the client to abrupt, but not frequent fluctuations in the available bandwidth. The first part forces the client to reduce the selected media level, whereas in the second part the client may gradually switch to the highest possible representation level.

5.2.1 Performance of the adaptation mechanisms

Figure 5.8, figure 5.9 and figure 5.10 represent respectively the performance of the aggressive, conservative, and mean mechanisms in terms of bandwidth utilization and buffer state. The analysis of the figures show that all algorithms started with the lowest quality level and successfully switched to the highest possible level within the first 12 s (aggressive: $t = 11$ s; conservative: $t = 8$ s; mean $t = 8$ s), subsequently achieving the 100% use of bandwidth ($u(t) = 1.0$).

The aggressive mechanism (figure 5.8) was significantly more sensitive to bandwidth fluctuations. Consequently, the selected media levels ($b(t)$) suffered unnecessary variations during the 250–350 s interval. This high sensitivity is illustrated at $t = 145$ s when there was a sudden diminution of bandwidth and the aggressive algorithm detected it appropriately.

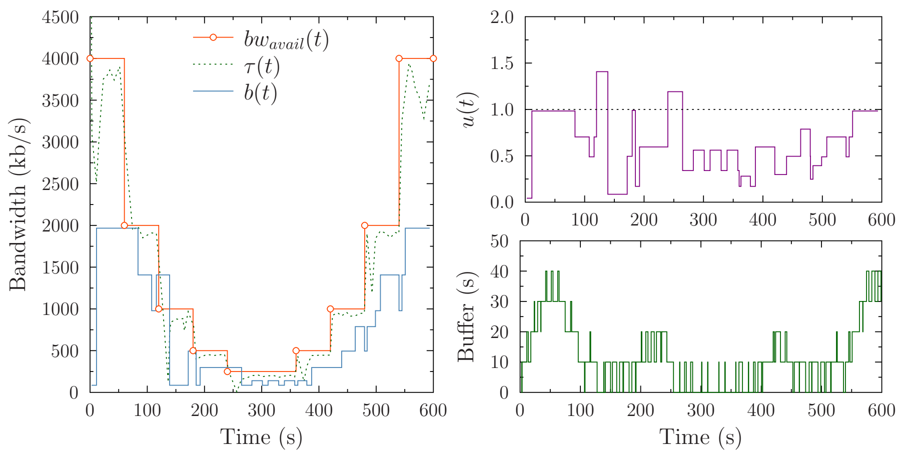


Figure 5.8: Performance of the aggressive mechanism over the scenario 1.

The conservative mechanism (figure 5.9) selected different quality levels with considerably fewer fluctuations than the aggressive method. During the whole session the aggressive mechanism performed 32 switching operations, whereas the conservative mechanism only switched 21 times (versus 25 switching operations in the case of the mean algorithm). However, in the 500–560 s interval, the conservative mechanism produced a notably lower efficiency (50%) than the other two mechanisms (about 70%).

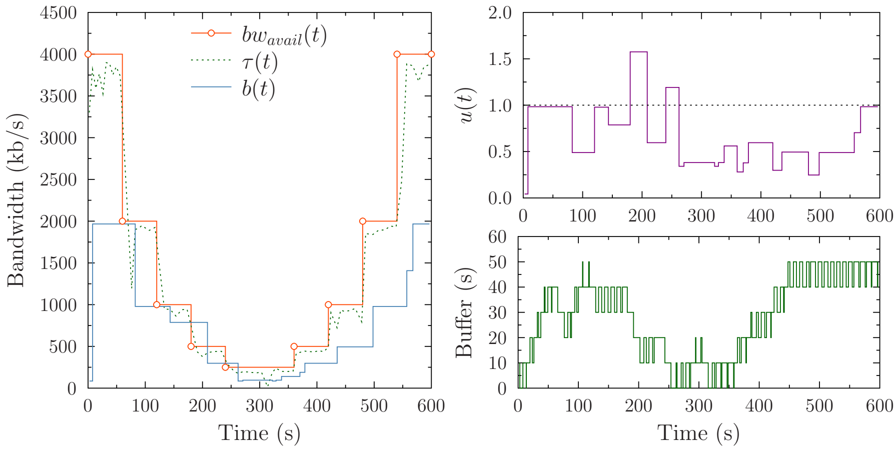


Figure 5.9: Performance of the conservative mechanism over the scenario 1.

The mean mechanism (figure 5.10) required two (or more) switching steps to select the appropriate bandwidth level. This behaviour is a consequence of the nature of the algorithm, which considers the last three throughput measurements (as defined in algorithm 3). These additional steps took place during the intervals 165–176 s and 360–385 s. Furthermore, this mechanism is the slowest one to adapt during the first part of the scenario (monotonically decreasing bit-rate). The bandwidth utilization remained at 200% ($u(t) = 2.0$) during 45.2 s, meaning that the quality of the segments was overestimated, i.e., segments' downloading times increased.

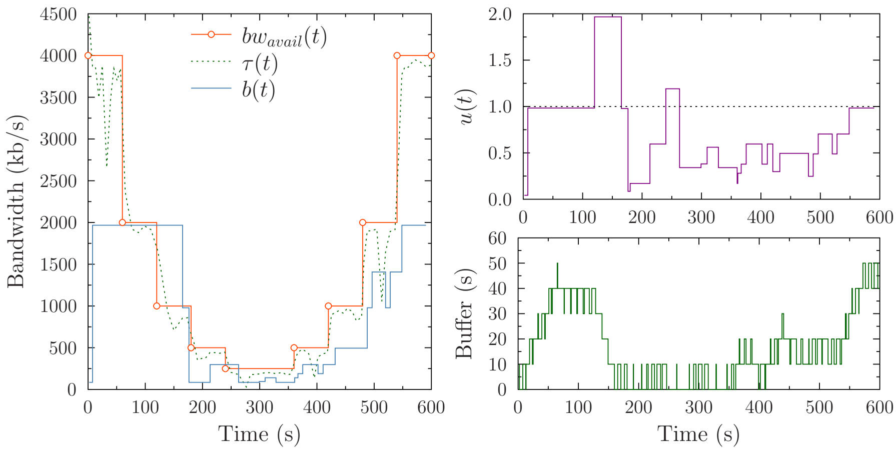


Figure 5.10: Performance of the mean mechanism over the scenario 1.

5.2.1.1 Impact on the metrics

Results of this experiment have been summarized in Table 5.9 and figure 5.11 according to the metrics defined previously in section 5.1.6. Figure 5.11 provides a graphical representation of table 5.9, where the differences among the metrics for each mechanism are easily discernible.

Figure 5.11: Graphical comparison under network scenario 1 for aggressive, conservative, and mean adaptive mechanisms.

Table 5.9: Computed metrics under network scenario 1 for aggressive, conservative, and mean adaptive mechanisms.

Mechanism	ε_{bw}	ε_{buf}	ε_{fetch}	ε_{retry}	ε_{active}	$\varepsilon_{start-up}$	ε_{up}	ε_{down}
Aggressive	60.22%	96.28%	91.6%	79.78%	49.99%	87.43%	69.70%	52.26%
Conservative	61.59%	100%	100%	92.59%	72.3%	87.51%	50.01%	43.22%
Mean	62.14%	94.71%	95%	84.26%	96.85%	90.08%	72.92%	37.14%

The bandwidth, buffering, and start-up efficiencies (ε_{bw}, ε_{buf}, $\varepsilon_{start-up}$) are not affected by the adaptive mechanism selected, whereas the active efficiency (ε_{active}) and the segment-retry efficiency (ε_{retry}) reveal a remarkable dependency.

The bandwidth efficiency (ε_{bw}) exceeds the 50% in all adaptive mechanisms (aggressive: 60.22%; conservative: 61.59%; and mean: 62.14%). With regard to the buffering efficiency (ε_{buf}), the conservative mechanism did not stop during the whole session (denoted by $\varepsilon_{buf} = 100\%$), whereas the aggressive and the mean mechanism produced interruptions in playback during 3.72% ($\varepsilon_{buf} = 96.28\%$) and 5.29% ($\varepsilon_{buf} = 94.71\%$) of the session time due to buffer underrun.

The segment-fetch efficiency (ε_{fetch}) is over 90% in all mechanisms, with the worst value in the aggressive one (91.6%). This means that less than 10% of the media content is not played. The aggressive mechanism re-downloads more segments due to more frequent switching operations (it presents the worst segment-retry efficiency, $\varepsilon_{retry} = 79.78\%$).

The active efficiency (ε_{active}) is significantly better for the mean mechanism (96.85%) than the aggressive (49.99%) and conservative mechanism (72.3%).

5.2.2 Performance with different duration segments

Figure 5.12 and figure 5.13 represent the behaviour of the conservative mechanism over time with a shorter and longer duration segments, respectively (for 10s-long segments, see figure 5.9). The fundamental difference between the three cases is the adaptability to the available bandwidth. It is notable that the reaction times are dependent of the duration of the segments, since the throughput is measured after segments have been transferred. Switching operations are produced earlier for 5s-long segments than 10s-long segments and consequently, even earlier than for 20s-long segments.

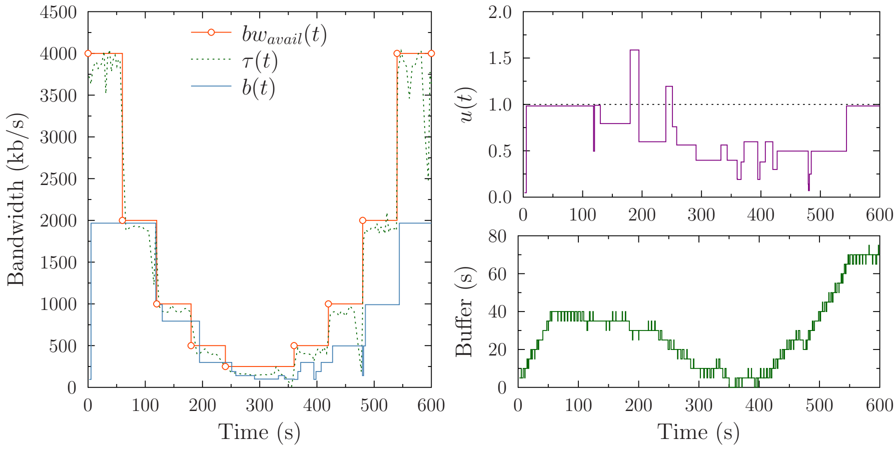


Figure 5.12: Performance of the conservative mechanism over the scenario 1 with a segment duration of 5 s.

Figure 5.13 shows that the selection of the adequate bit-rate level for 20s-long segments is performed inadequately during the first 300 s of the session. The conservative mechanism downloaded the first and second media segments at the lowest quality levels, therefore it takes more than 40 s (a pair of 20s-long segments) to switch to the highest possible quality level. Upon reduction of the available bandwidth, segments are more likely to be discarded therefore the buffer can be empty. In consequence, levels are unnecessarily lowered to the minimum quality level in four occasions ($t = 77$ s, $t = 146$ s, $t = 207$ s and $t = 283$ s) due to buffer underrun.

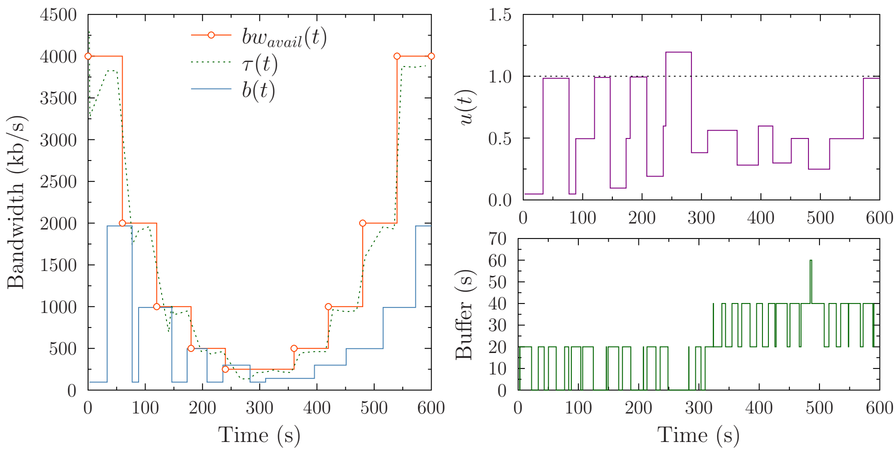


Figure 5.13: Performance of the conservative mechanism over the scenario 1 with a segment duration of 20 s.

5.2.2.1 Impact on the metrics

Table 5.10 and figure 5.14 summarize the results of this experiment. The buffering, segment-fetch, segment-retry, and start-up efficiencies (ε_{buf}, ε_{fetch}, ε_{retry}, and $\varepsilon_{start-up}$, respectively) show minor dependency on the duration of the segments. In contrast, the active efficiency (ε_{active}) and the reaction efficiencies (ε_{up} and ε_{down}) are improved for shorter duration of segments (best case for 5s-long, worse case for 20s-long).

In general, a shorter segment duration improves the efficiencies defined in our evaluation. In the case of 5s-long segments, only the start-up efficiency ($\varepsilon_{start-up}$) presents worse values than 20s-long segments. Since the required buffered time to start playback is set to 10 s

($T_{min-buf}$ = 10 s), two 5s-long segments need to be downloaded to start playback, whereas only one segment is enough if $S_d \geq 10$ s.

Table 5.10: Metrics comparison under network scenario 1 for 5s-long, 10s-long and 20s-long segments.

S_d (s)	ε_{bw}	ε_{buf}	ε_{fetch}	ε_{retry}	ε_{active}	$\varepsilon_{start-up}$	ε_{up}	ε_{down}
5	64.18%	96.7%	95.83%	96.77%	100%	80.55%	62.11%	63.82%
10	61.59%	100%	100%	92.59%	72.3%	87.51%	50.01%	43.22%
20	52.52%	97.23%	100%	90.36%	53.4%	88.69%	37.24%	41.12%

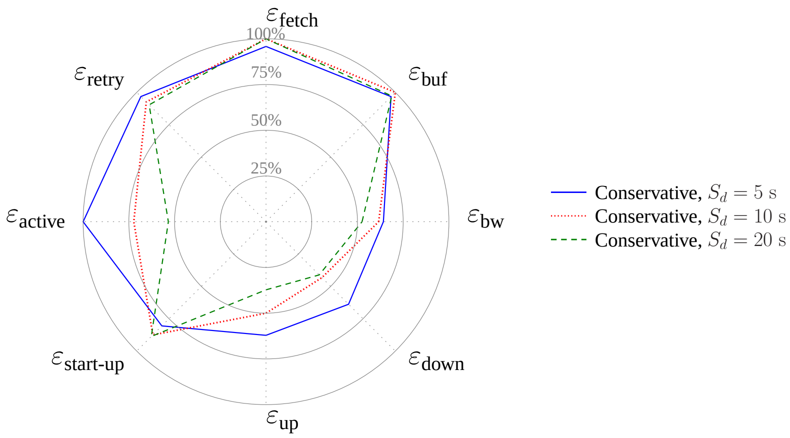


Figure 5.14: Graphical comparison under network scenario 1 for 5s-long, 10s-long, and 20s-long segments.

5.2.3 Analysis of the end-to-end latency

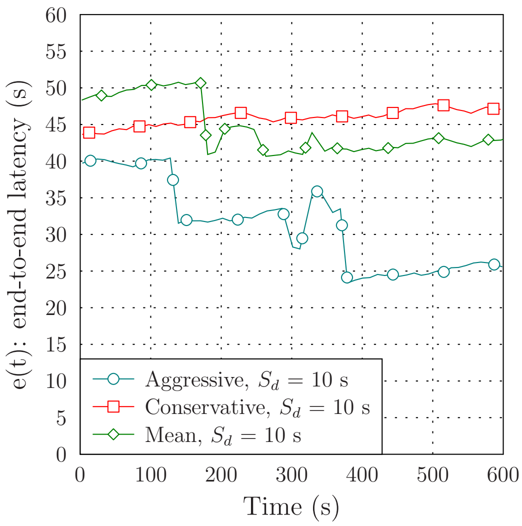
Figure 5.15 represents the end-to-end latency ($e(t)$) for the first scenario in two situations: (1) experiments with the three adaptation mechanisms and (2) experiments with different segment durations.

Note that in our evaluation, the end-to-end latency is **decreased** if segments are missed (the client receives `HTTP-404` responses), since the next segment in the playlist will be played earlier[3]. In contrast, the end-to-end latency is **increased** if the playback is interrupted due to an empty buffer, since the next segment will be player after the interruption in playback. Thus, the important aspects of the $e(t)$ function are related to the variations produced by these two occurrences (missed segments and playback interruptions).

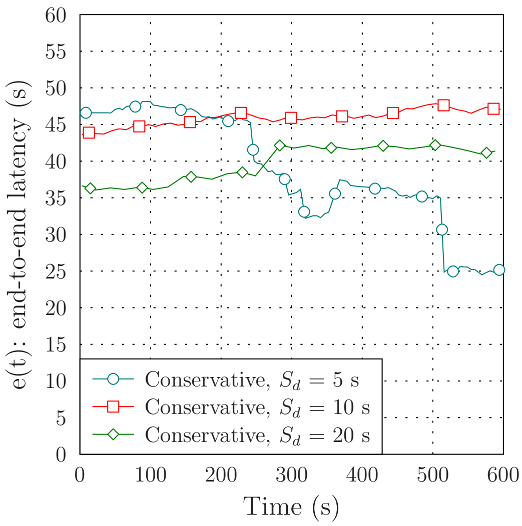
According to figure 5.15a, the aggressive mechanism presents the most significant variation in latency, with a delay of 40 s at the beginning of the session ($t = 0$ s) and 25 s at the end ($t = 600$ s). The effects of packet losses can be seen for the mean mechanism around $t = 190$ s and $t = 250$ s when the latency is reduced. The conservative mechanism is the only one whose latency function remains stable throughout the whole experiment, meaning that there were no segments lost and there were no playback interruptions.

The effects of segment durations on the end-to-end latency are shown in figure 5.15b. A shorter duration of the segments ($S_d = 5$ s) produced more variation of the end-to-end delay: 47 s at the beginning of the session and 25 s at the end. A longer duration of the segments ($S_d = 20$ s) only produced an increase of end-to-end delay at $t = 250$ s.

[3] In this master's thesis project, segments are played sequentially. If there is a missed segment, the next buffered segment is played directly without waiting.



(a) Different adaptation mechanisms



(b) Different duration of segments

Figure 5.15: End-to-end latency throughout the session for scenario 1.

5.2.4 Discussion

The first scenario was intended to test the behaviour of the adaptation algorithms under the reduction of the available bit-rate (performed in four steps until the middle of the session) and subsequent increase of the available bit-rate until the end of the experiment.

All of the adaptation mechanism were able to detect the bit-rate fluctuations although they exhibited major differences. The aggressive mechanism performed multiple switching operations due to small variations in the throughput measurements. The quality levels selected by the client oscillated more with the aggressive and mean mechanism, without any real improvement in the utilization of bandwidth.

The mean algorithm reacts later than the other mechanisms to the bit-rate fluctuations, requiring several switching steps to select the highest possible representation. The conservative mechanism performed fewer switching operations, thus seeming to be the most appropriate algorithm for this scenario, with the cost of underestimation of the available bandwidth at nearly the end of the session.

The effects of different segment durations take place in the first part of the scenario 1 (when bit-rates are decreased). Longer segment duration lead to a inadequate utilization of the available bandwidth, since throughput measurements are taken less frequently. Consequently, buffer underrun events are more probable, resulting in a abrupt reduction of the media bit-rate selected.

5.3 Scenario 2: short-term variations of the available bandwidth

The second network scenario (represented in figure 5.16) produces periodic short-term variations of the available bandwidth. In this context, short-term changes are fluctuations produced at intervals of a maximum of 30 s. The duration of these intervals is deliberately chosen to analyze their impact on the mean algorithm, which considers the last three throughput measurements.

Two parts are defined in this network scenario: in the first 5 minutes (300 s) the bit-rates oscillate between 250 kb/s and 1 Mb/s with a frequency of 1/30 Hz. In the second part, bit-rates switch between 1 Mb/s and 4 Mb/s. This scenario is intended to evaluate the behaviour of the client with more frequent bit-rate fluctuations than scenario 1.

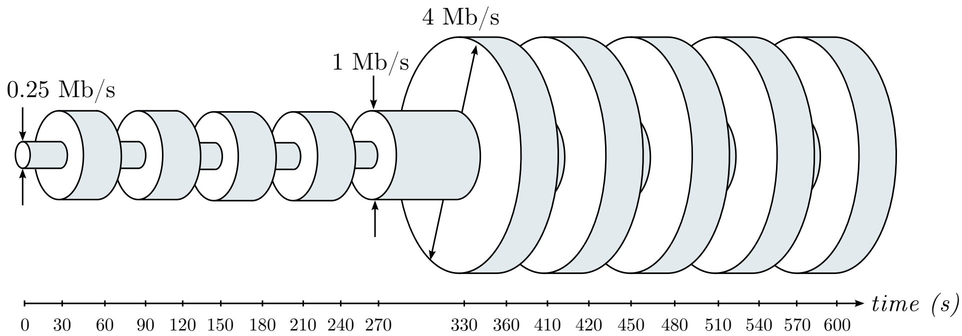


Figure 5.16: Series of Dummynet pipes for the second network scenario.

5.3.1 Performance of the adaptation mechanisms

Figure 5.17, figure 5.18, and figure 5.19 illustrate the selected bandwidth by the three different mechanisms over time. The conservative mechanism offers the most stable behaviour in comparison with the aggressive and mean algorithms.

The aggressive mechanism (figure 5.17) seems to produce better adaptation upon the bandwidth restrictions in the first part of the experiments (from 0 s to 300 s). In this part, it only overuses the offered network bit-rate during two short intervals (60–66 s and 180–199 s). However, it is noticeable that there is an excessive number of switching operations (i.e., 28 switches). In the second part (from 300 s to 600 s) the bandwidth is doubly overestimated on four occasions[4] (intervals 360–390 s, 420–450 s, 480–496 s and 540–565 s). Finally, in t = 500 s the mechanism needed to switch to the lowest quality level.

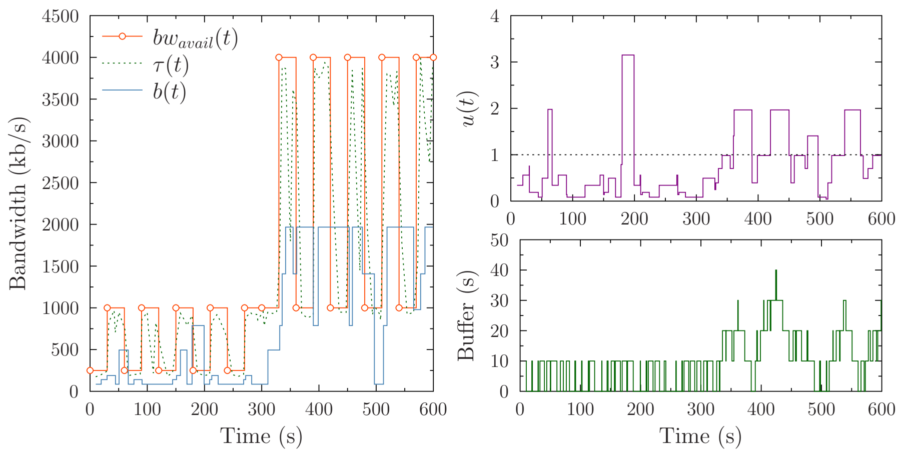


Figure 5.17: Performance of the aggressive mechanism over the scenario 2.

The conservative mechanism (figure 5.18) appropriately followed the fluctuations produced by this scenario. During the first part, the bandwidth utilization ($u(t)$) is better than the aggressive scenario (about 50%), although it is overestimated in more occasions (up to four). After $t = 300$ s, $u(t)$ oscillates between 100% an 200%, with short intervals where the utilization decreases to 50%. This mechanism did not need to switch down abruptly to the lowest quality level during the second part of the experiment.

[4]Note that the starting time of the intervals always corresponds to a bit-rate fluctuation in the network's emulated bandwidth.

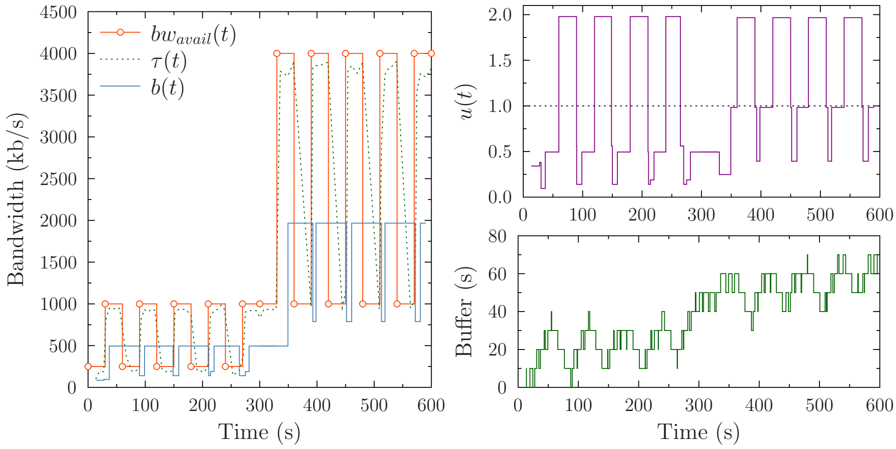


Figure 5.18: Performance of the conservative mechanism over the scenario 2.

The mean mechanism (figure 5.19) presented a mixed behaviour between the two previous mechanisms. In the first part of the experiment, 25 switching operations were performed. The available bandwidth was overestimated for four intervals, although the duration of those intervals differ (unlike the conservative mechanism). In the second part, the mean mechanism needed to switch to the lowest quality levels on two occasions, increasing to 100% of bandwidth utilization in several switching steps.

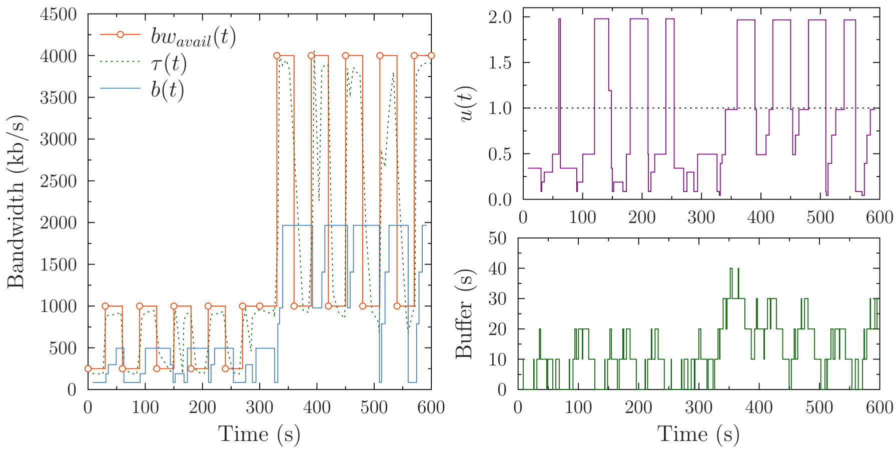


Figure 5.19: Performance of the mean mechanism over the scenario 2.

5.3.1.1 Impact on the metrics

Table 5.11 and figure 5.20 collect the performance results of the three adaptive mechanisms under the second scenario. The majority of coefficients have similar values. Only the segment-retry efficiency (ε_{retry}) turns out to be considerable better for the conservative mechanism (86.66%) than the aggressive (51.66%) and mean (48.33%) mechanisms.

Active efficiency (ε_{active}), start-up efficiency ($\varepsilon_{start-up}$), and reaction efficiencies (ε_{up} and ε_{down}) are slightly better for the mean mechanism and aggressive mechanisms.

Table 5.11: Metrics comparison under network scenario 2 for aggressive, conservative, and mean adaptive mechanisms.

Mechanism	ε_{bw}	ε_{buf}	ε_{fetch}	ε_{retry}	ε_{active}	$\varepsilon_{start-up}$	ε_{up}	ε_{down}
Aggressive	45.60%	98.34%	95%	74.25%	70.67%	66.17%	72.73%	47.65%
Conservative	52.95%	100%	98.33%	91.46%	70.3%	59.14%	70.09%	39.72%
Mean	48.87%	94.36%	91.66%	74.25%	83.07%	70.17%	79.91%	46.29%

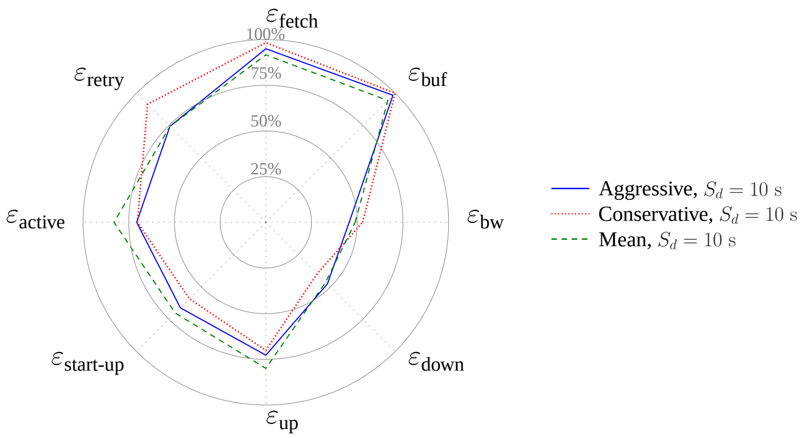


Figure 5.20: Graphical comparison under network scenario 2 for aggressive, conservative, and mean adaptive mechanisms.

5.3.2 Performance with different duration segments

Figure 5.21 represents the performance of the conservative mechanism with 5s-long segments. Reaction times are significantly improved although the overall adaptation is similar that achieved for segments of 10 s (figure 5.18 in page 78).

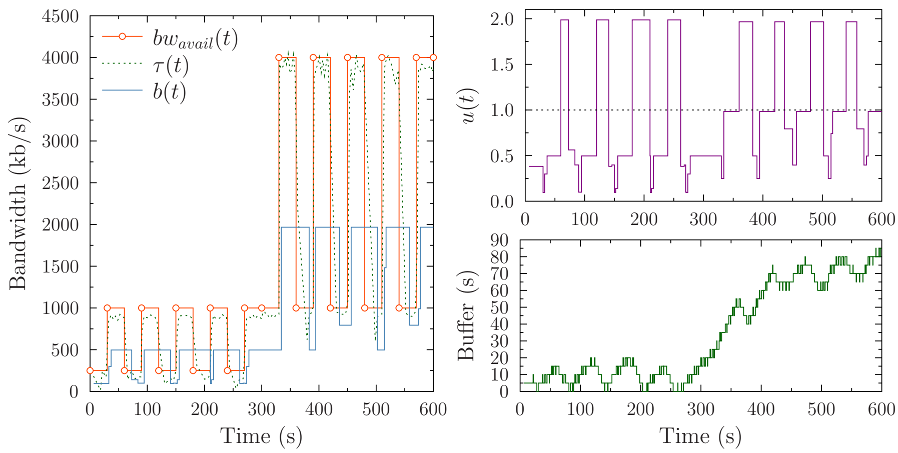


Figure 5.21: Performance of the conservative mechanism over the scenario 2 with a segment duration of 5 s.

Deficient reaction times are illustrated in figure 5.22, due to use of 20s-long segments. Under these conditions, the conservative algorithm is not able to follow rapid bit-rate variations, since the algorithm is only run after a whole segment has been downloaded. This disadvantage is present in the throughput curves ($\tau(t)$) of both figures. For segments of 5 s, $\tau(t)$ is contained within the limits of the available bandwidth ($bw_{avail}(t)$) whereas for 20 s, the throughput measurements are more inaccurate.

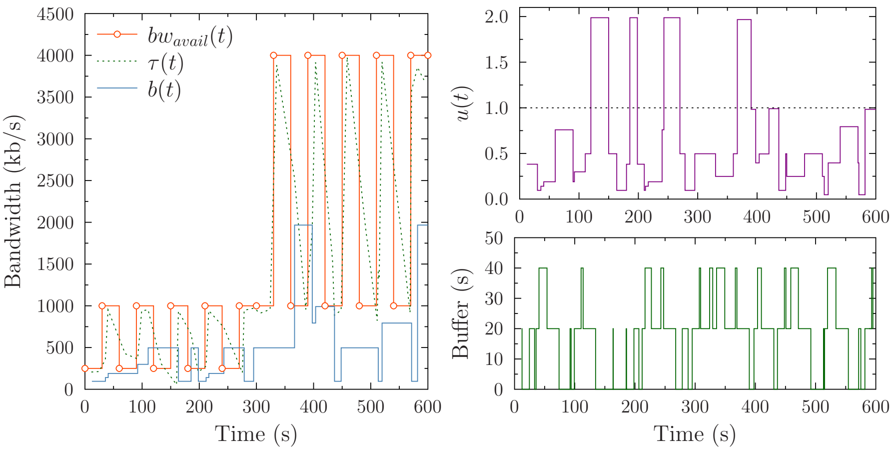


Figure 5.22: Performance of the conservative mechanism over the scenario 2 with a segment duration of 20 s.

5.3.2.1 Impact on the metrics

The metrics for the scenario 2 are collected in table 5.12 and figure 5.23. 5s-long segments provide a better performance for most of the parameters (ε_{bw}, ε_{buf}, ε_{retry}, ε_{active}, ε_{up}, and ε_{down}), and similar results for the segment-fetch efficiency.

The most influenced metrics in this scenario are the active efficiency (ε_{active}) and the reaction time for switching up operations (ε_{up}). The improvement of ε_{active} relates to the number of requests needed for shorter segments. In order to download 20 s of video content, four 5s-long segments have to be downloaded, whereas just one for 20s-long segments. ε_{up} is completely dependent of the segment length (5s-long: 84.11%, 10s-long: 39.06%, 20s-long: 17.25%).

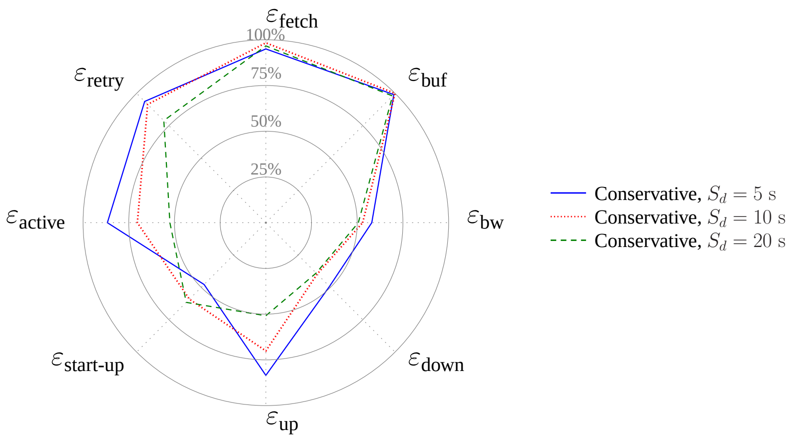


Figure 5.23: Graphical comparison under network scenario 2 for 5s-long, 10s-long and 20-seconds long segments.

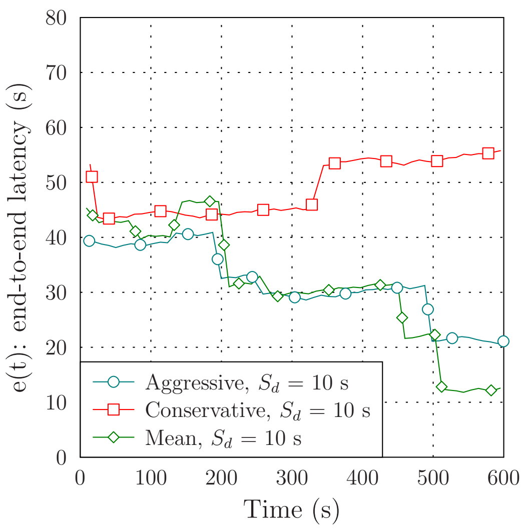
Table 5.12: Metrics comparison under network scenario 2 for 5s-long, 10s-long and 20s-long segments.

S_d (s)	ε_{bw}	ε_{buf}	ε_{fetch}	ε_{retry}	ε_{active}	$\varepsilon_{start-up}$	ε_{up}	ε_{down}
5	57.96%	99.03%	95%	93.75%	86.7%	47.78%	83.46%	49.26%
10	52.95%	100%	98.33%	91.46%	70.3%	59.14%	70.09%	39.72%
20	50.6%	98.01%	96.66%	78.94%	52.5%	61.59%	50.87%	38.93%

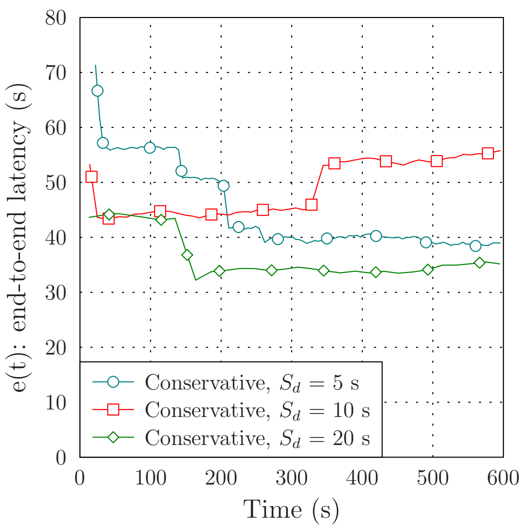
5.3.3 Analysis of the end-to-end latency

Figure 5.24 depicts the end-to-end latency during the session for the second scenario. As it can be seen in figure 5.24a, the mean mechanism has the strongest variation of end-to-end delay, starting with 45 s of delay at the beginning of the session, being reduced to 11 s at the end of the experiment ($t = 600$ s). This behaviour is caused by successive segments missed in $t = 200$ s, $t = 450$ s, and $t = 500$ s. The aggressive mechanism presents a similar behaviour to the mean mechanism, reducing the end-to-end delay from 40 s ($t = 0$ s) to 20 s ($t = 600$ s).

Considering different segment durations (see figure 5.24b), a shorter segment duration (5s-long) reduces the performance in the first part (up to $t = 300$ s), since more segments are missed, reducing on several occasions from an end-to-end delay of 70 s to 40 s. A longer segment duration improves the stability of the end-to-end delay function throughout the session. Only one change in the $e(t)$ function occurs for 20s-long segments ($t = 150$ s).



(a) Different adaptation mechanisms



(b) Different duration of segments

Figure 5.24: End-to-end latency throughout the session for scenario 2.

5.3.4 Discussion

The second network scenario was intended to test the behaviour of the adaptation algorithms under more frequent variations of the available bandwidth. The conservative mechanism selected more appropriately the bit-rates levels, whereas the aggressive mechanism produced unnecessarily fluctuations, especially in the first part of the experiment. The mean mechanism performed with a mixed behaviour between that of the aggressive and the conservative mechanisms, as it was able to select the appropriate quality level but required an additional switching step. The most significant difference among the adaptation mechanism is the number of segments which are re-downloaded due to variations in the available bandwidth.

A larger size of the segment drastically reduced the reaction times, resulting in a delayed selection of the bit-rate levels. In contrast, a shorter segment duration produced the opposite effects, reducing the delay between switching operations.

5.4 Scenario 3: peaks in the available bandwidth

The third network scenario (represented in figure 5.25) supplies a constant low bit-rate of 250 kb/s with high bit-rate peaks (more than ten times higher, 3 Mb/s) at intervals of one minute. The duration of these peaks is increased over time, starting from 5 s up to 35 s. This scenario aims to measure the reaction time (detection) of the client upon extreme variations of bit-rate. This scenario is especially tricky and challenging for the adaptive mechanisms: detecting the end of the peak becomes the most important issue in order to avoid an empty buffer.

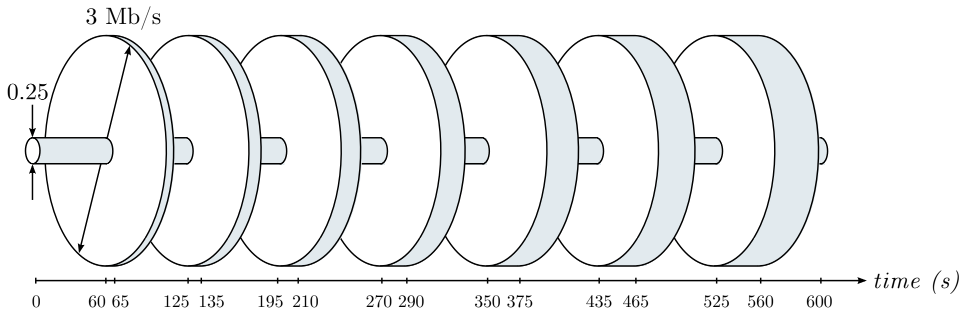


Figure 5.25: Series of Dummynet pipes for the third network scenario over time.

5.4.1 Performance of the adaptation mechanisms

The quality level selected for the three adaptive mechanism is depicted in figures 5.26, 5.27, and 5.28. Under this scenario, the aggressive and conservative mechanisms illustrate a equal behaviour in terms of detecting the bit-rate peaks. Neither of these two methods switches to a higher quality level for the first peak (of 5 s duration). However, adaptation is successfully performed in the following six peaks.

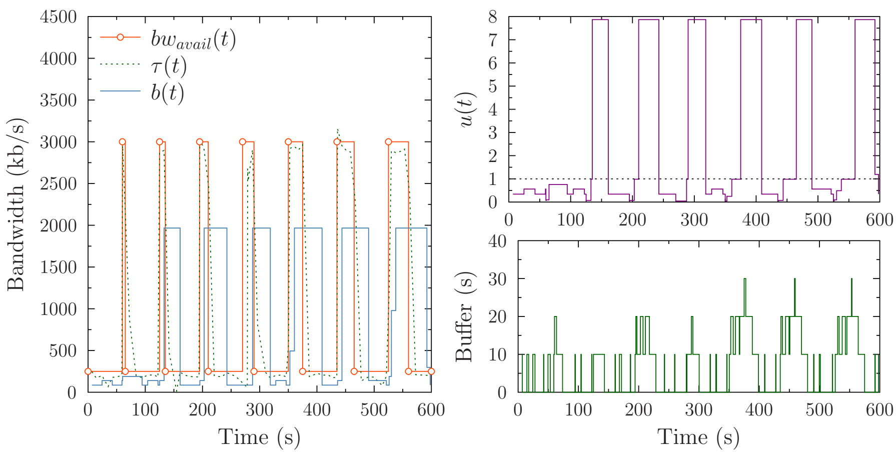


Figure 5.26: Performance of the aggressive mechanism over the scenario 3.

The most inadequate behaviour was observed in the case of the mean algorithm (figure 5.28), which presents an irregular adaptation throughout the session. It is capable of detecting all the peaks, but the switching down is performed too late as compared to the other two mechanisms.

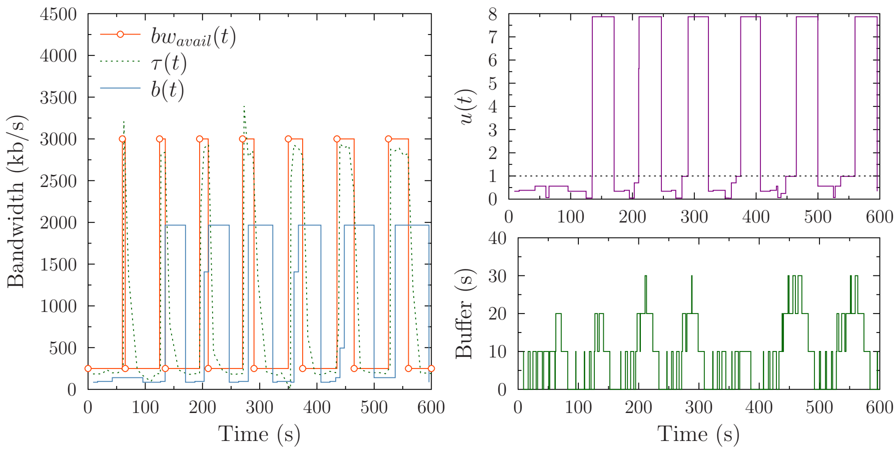


Figure 5.27: Performance of the conservative mechanism over the scenario 3.

The worst situation happens at $t = 290\ s$, when bit-rate is reduced from 3 Mb/s to 250 kb/s. The algorithm maintained the quality level for 62 s, leading to an empty buffer - hence no playout.

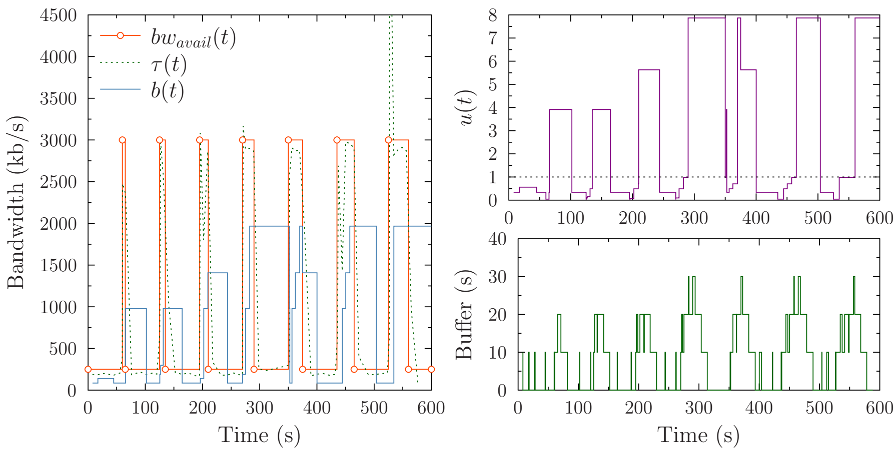


Figure 5.28: Performance of the mean mechanism over the scenario 3.

5.4.1.1 Impact on the metrics

Results of this experiments are listed in table 5.13 and represented graphically in figure 5.29. The stops during playback and the number of missed segments are frequent occurrences in this network scenario, consequently reducing ε_{retry}, ε_{fetch}, and ε_{buf}.

Table 5.13: Metrics comparison under network scenario 3 for aggressive, conservative, and mean adaptive mechanisms.

Mechanism	ε_{bw}	ε_{buf}	ε_{fetch}	ε_{retry}	ε_{active}	$\varepsilon_{start-up}$	ε_{up}	ε_{down}
Aggressive	38.87%	84.37%	80%	77.31%	100%	74.25%	85.71%	40.23%
Conservative	34.54%	94.22%	91%	83.33%	100%	68.89%	68.74%	36.63%
Mean	31.22%	83.07%	78.33%	77.31%	100%	71.43%	80.85%	34.60%

Figure 5.29: Graphical comparison under network scenario 3 for aggressive, conservative, and mean adaptive mechanisms.

The conservative mechanism provides the best management of the fluctuations produced by the scenario, but it has poor reaction (ε_{up} = 68.74%, ε_{down} = 36.63%) and start-up efficiencies ($\varepsilon_{start-up}$ = 68.89%). In particular, ε_{up} shows better values for the aggressive (85.71%) and the mean mechanisms (80.85%).

5.4.2 Performance with different duration of segments

Figure 5.30 and figure 5.31 depicts the performance of the conservative mechanism with 5s-long and 20s-long segments respectively. All peaks are successfully detected in both cases (as the throughput curve denotes). However, the essential difference is present in the first peaks (shorter in time, produced at t = 60 s and t = 125 s) which are adequately utilized with shorter segments (figure 5.30) in comparison with longer segments (figure 5.31).

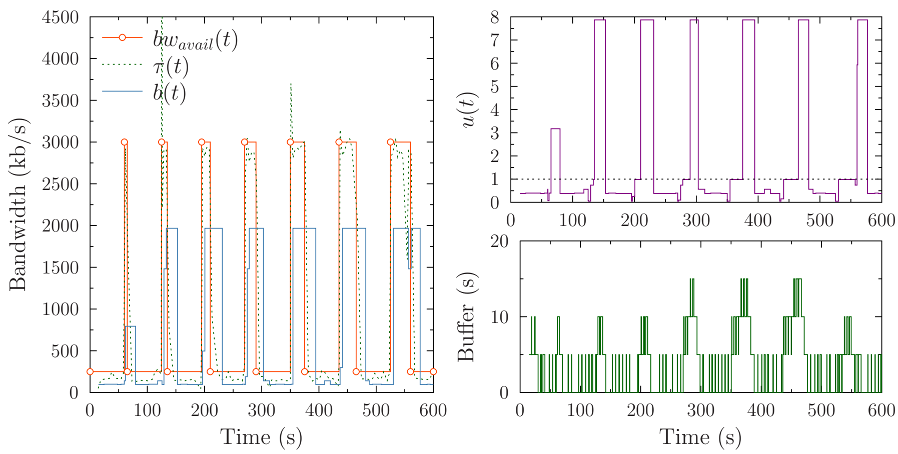


Figure 5.30: Performance of the conservative mechanism over the scenario 3 with a segment duration of 5 s.

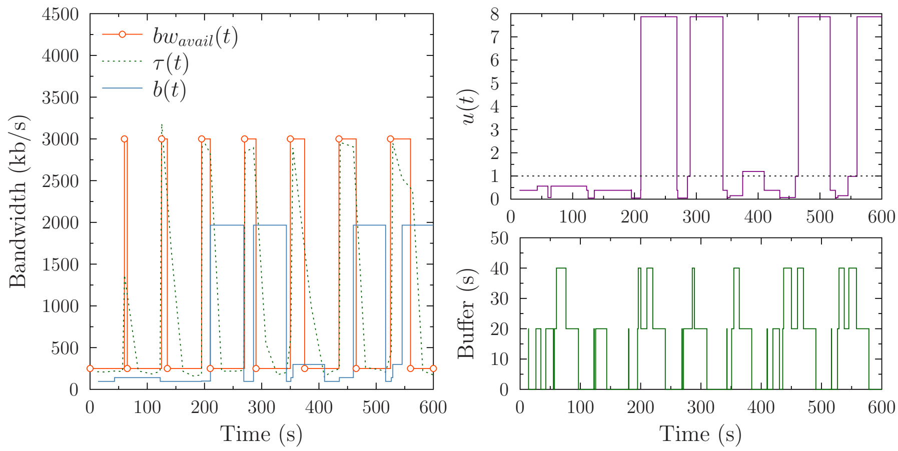


Figure 5.31: Performance of the conservative mechanism over the scenario 3 with a segment duration of 20 s.

5.4.2.1 Impact on the metrics

Results of this experiments are expressed in table 5.14 and figure 5.32. Results present high disparity for all the coefficients, meaning that the segment duration has a strong importance on this type of scenario.

The reaction (up and down) efficiencies and the active efficiency are the metrics which presents more variation (more than 50%), whereas the buffering efficiency (ε_{buf}) is less affected by the length of segments (from 6.97% to 19.43%, greater for shorter segments).

Table 5.14: Metrics comparison under network scenario 3 for 5s-long, 10s-long, and 20s-long segments.

S_d (s)	ε_{bw}	ε_{buf}	ε_{fetch}	ε_{retry}	ε_{active}	$\varepsilon_{start-up}$	ε_{up}	ε_{down}
5	42.87%	86.4%	79.16%	86.70%	100%	50.88%	85.01%	53.83%
10	34.54%	94.22%	91.66%	83.33%	100%	68.89%	68.74%	36.63%
20	30.27%	90.63%	93.33%	74.25%	55.8%	58.90%	62.03%	29.05%

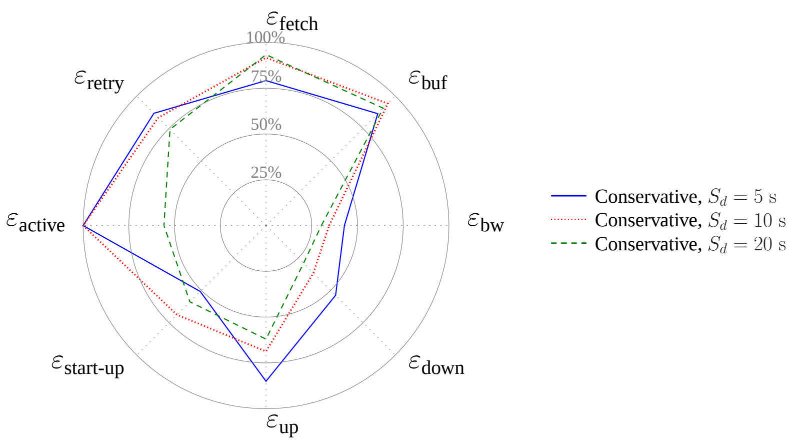


Figure 5.32: Graphical comparison under network scenario 3 for 5s-long, 10s-long, and 20-seconds long segments.

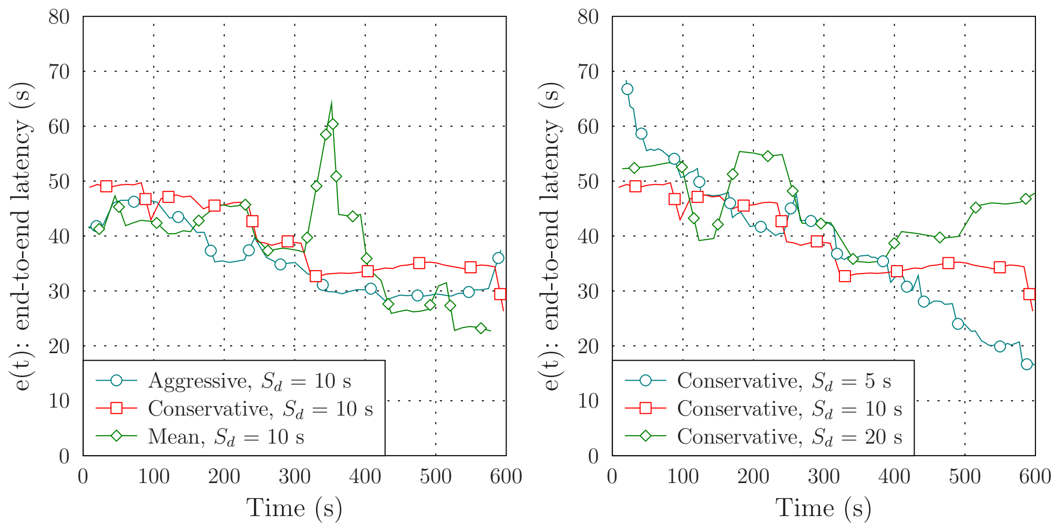
5.4.3 Analysis of the end-to-end latency

Figure 5.33 depicts the end-to-end latency in the experiments in the third scenario. The most noticeable occurrence is the irregularities of the end-to-end delay throughout the session for all cases, independent of the chosen adaptation mechanism or segment duration. Results of these experiments show that peaks in the network's available bandwidth have a strong influence on the segments' playback time-stamps, i.e., when the media segments are played.

The longest playback interruption can be seen in the mean mechanism at $t = 323$ s (see figure 5.33a), when the player was buffering content during 27 s. As a result, the end-to-end delay was abruptly increased from 37 s to 64 s. Several segments are missed after this occurrence, as it can be seen in a reduction of 26 s in the end-to-end delay ($t = 436$ s).

The aggressive and conservative mechanisms only achieve stability in the $e(t)$ function in the second part of the experiment (from $t = 300$ s), when bit-rate peaks last for more than 20 seconds.

Reducing the segment duration does not improve the end-to-end delay stability. As depicted in figure 5.33b, using 5s-long segments decreases monotonically the $e(t)$ function more than 50 s. A longer segment duration (20 s) produces strongest variations in the end-to-end delay for the shorter peaks (as it can be seen at $t = 96$ s, $t = 143$ s and $t = 240$ s). Stability in the $e(t)$ function for 20s-long segments is poorly achieved in the second part of the experiment due to playback interruptions ($t = 404$ s and $t = 410$ s).



(a) Different adaptation mechanisms

(b) Different duration of segments

Figure 5.33: End-to-end latency throughout the session for scenario 3.

5.4.4 Discussion

The third scenario was intended to test the behaviour of the adaptation algorithms under high-bandwidth-peaks of different durations. The overall performance of the aggressive and the mean algorithm was worse than the conservative mechanism, although the latter showed a notable irregularity in the selection of bit-rate levels. The computed metrics show that the conservative mechanism prevented playback interruptions better than the other mechanisms.

The size of the segments has a significant influence in this scenario. The use of shorter segments improves the reaction times, but increases the probability of missed segments due to `HTTP-404` responses. A larger size of the segments leads to a more inactive player, since fewer HTTP requests need to be sent. As a consequence, peaks in the available bandwidth are not used appropriately (peaks are detected but segments are more probable to be re-downloaded since the segment size in bytes is much bigger).

5.5 Scenario 4: troughs in the available bandwidth

The fourth network scenario presents the opposite situation of the third scenario (evaluated in section 5.4). Scenario 4 supplies a continuous high bit-rate of 3 Mb/s, with low bit-rate troughs (more than ten times lower, 250 kb/s) every minute, as shown in figure 5.34. Duration of these troughs is also increased over time, the first troughs lasts 5 s increasing up to 35 s in the last one.

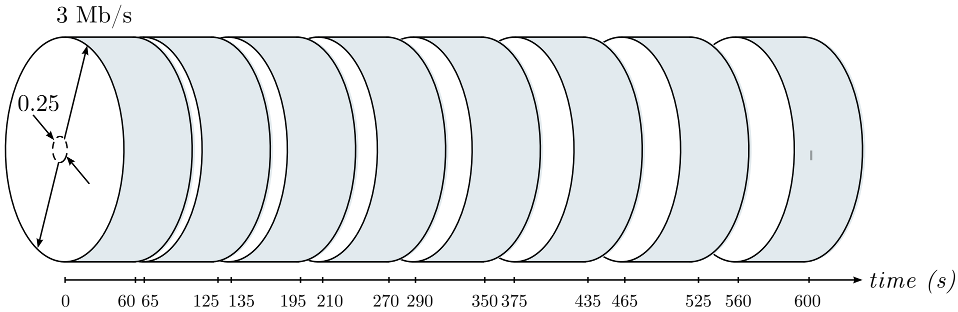


Figure 5.34: Series of Dummynet pipes for the fourth network scenario.

5.5.1 Performance of the adaptation mechanisms

Figure 5.35, 5.36, and 5.37 show the performance of the three adaptive mechanisms under the network conditions of the fourth network scenario. The most noticeable difference is the election of the representation level during the abrupt decreases in the available bandwidth.

The aggressive mechanism (figure 5.35) reduced the quality level following the first trough. However, for the following two troughs it maintained the maximum level, keeping the bandwidth utilization at 100% between troughs. After the fourth, fifth, and sixth troughs ($t = 270$ s, $t = 350$ s, and $t = 435$ s, respectively) the level was switch down to an intermediate level, since the measured throughput was imprecise (the bandwidth detected during troughs was (incorrectly) estimated to be over 500 kb/s). During the last trough ($t = 525$ s), the buffer reached the minimum due to several segments which were re-downloaded. As a result, the level was set to the minimum and the bandwidth was correctly estimated.

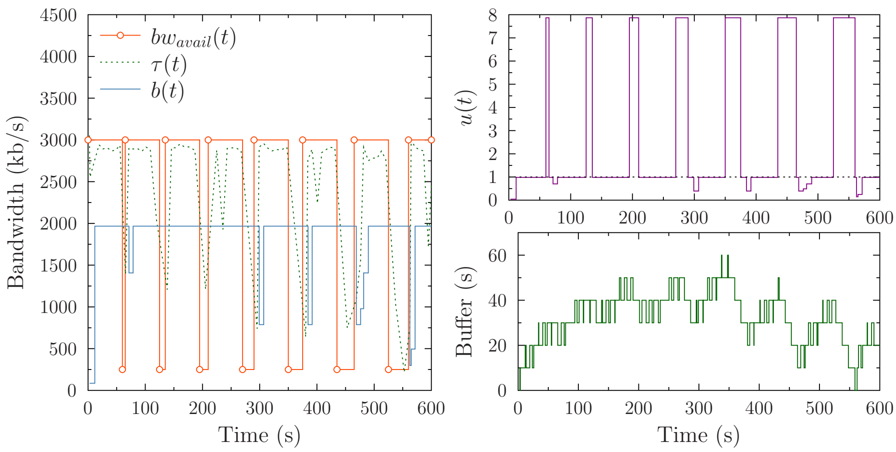


Figure 5.35: Performance of the aggressive mechanism over the scenario 4.

Figure 5.36 depicts the behaviour of the conservative mechanism. As expected, quality levels are selected slightly below the optimal. For all troughs this mechanism successfully lowered the bit-rate level to 1 Mb/s or below. The buffering procedure was significantly better for the overall session in comparison with the aggressive mechanism. From the start until $t = 100$ s the buffer fluctuates between one and two segments stored (after 10 s and 20 s respectively). After that interval, the buffer is never empty.

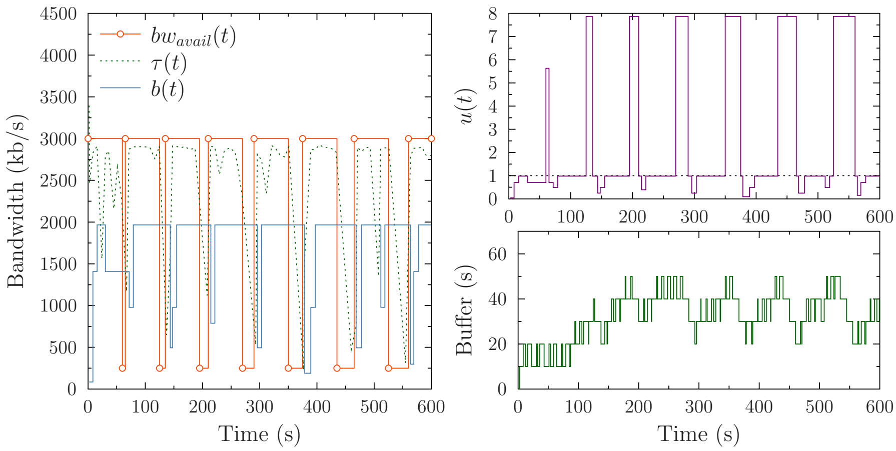


Figure 5.36: Performance of the conservative mechanism over the scenario 4.

The mean mechanism (figure 5.37) presented a mixed behaviour between the aggressive and the conservative mechanisms. Troughs are correctly detected, but the selected bit-rate level is greater than the client's measured throughput. Only in the trough produced in $t = 435$ s does the quality level reached the minimum, while in the rest of the troughs the quality selected was between 1 Mb/s and 1.5 Mb/s. In addition, the buffer becomes empty in the last three troughs.

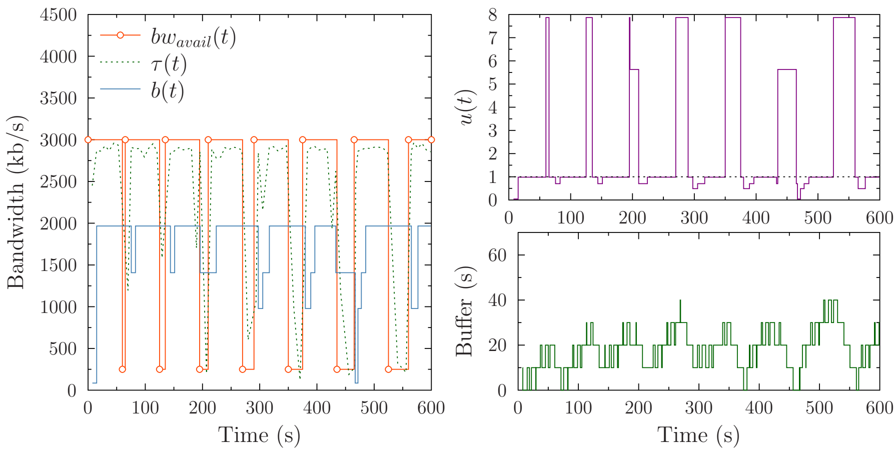


Figure 5.37: Performance of the mean mechanism over the scenario 4.

5.5.1.1 Impact on the metrics

Table 5.15 and figure 5.38 summarize the experimental results under this scenario. The active efficiency (ε_{active}) is the metric which shows the biggest variation for the different algorithms (more than 50%) being the best case $\varepsilon_{active} = 86.74$ for the mean mechanism. However, the bandwidth utilization efficiency (ε_{bw}) is quite similar in the three cases (around 70%), meaning that the mean algorithm spent more time downloading segments without a clear adaptation improvement. This leads to a reduction of the segment-retry efficiency (ε_{retry}).

Table 5.15: Metrics comparison under network scenario 4 for aggressive, conservative, and mean adaptive mechanisms.

Mechanism	ε_{bw}	ε_{buf}	ε_{fetch}	ε_{retry}	ε_{active}	$\varepsilon_{start-up}$	ε_{up}	ε_{down}
Aggressive	72.15%	100%	96%	92.59%	39.41%	87.26%	58.77%	40.59%
Conservative	66.45%	100%	100%	90.36%	36.9%	89.69%	61.12%	58.06%
Mean	69.94%	99%	98%	85.22%	86.74%	71.91%	58.06%	45.96%

Figure 5.38: Graphical comparison under network scenario 4 for aggressive, conservative, and mean adaptive mechanisms.

5.5.2 Performance with different duration segments

Figure 5.39 and figure 5.40 illustrate the behaviour of the conservative mechanism with a segment duration of 5 s and 10 s. Reaction times are significantly improved for the case of shorter segment duration, as it is shown in figure 5.39. The quality level is reduced in all troughs accordingly, less than 500 kb/s except for the shortest trough (t = 60 s), where the bit-rate is only decreased to 800 kb/s. It is important to point out that the periods when the quality was lowered are shorter.

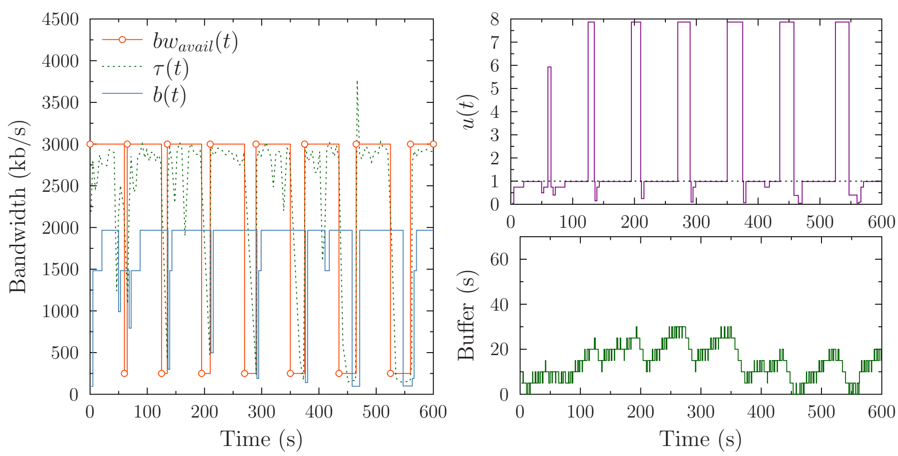


Figure 5.39: Performance of the conservative mechanism over the scenario 4 with a segment duration of 5 s.

The behaviour of the conservative mechanism with a longer duration segments (20 s) is illustrated in figure 5.40. The resulting adaptation is poor compared to 5s-long and 10s-long segments, presenting no benefit. The reduction of bit-rate clearly influences the selection of the quality level, leading to an under-estimation of the available bandwidth in the network. This effect is illustrated in the troughs produced at $t = 60$ s, $t = 125$ s, $t = 195$ s, and $t = 525$ s. After the fourth trough ($t = 270$ s), the mechanism could not reach the maximum quality level.

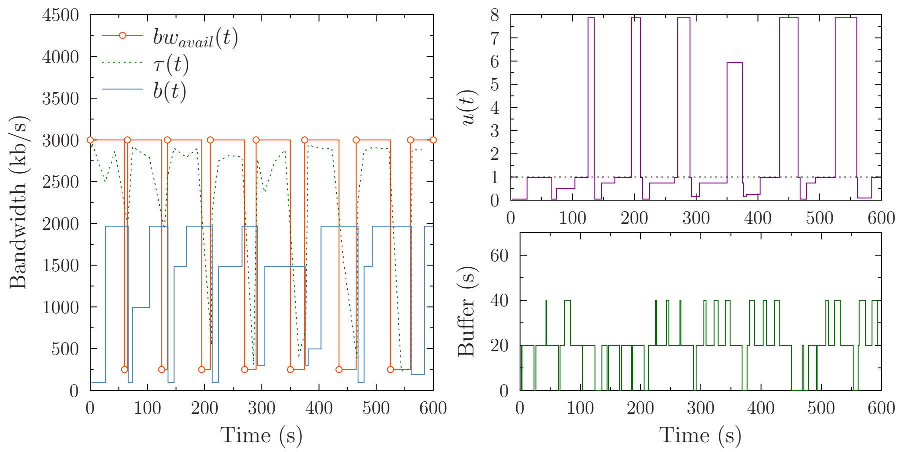


Figure 5.40: Performance of the conservative mechanism over the scenario 4 with a segment duration of 20 s.

5.5.2.1 Impact on the metrics

Results of scenario 4 with different segment durations are presented in Table 5.16 and figure 5.41. They reveal that the use of a shorter segment duration significantly improves the reaction efficiencies (ε_{up} and ε_{down}), the active efficiency, and most importantly, the utilization of bandwidth (ε_{bw}) is about 10% better than using 10s-long segments. The cost of these improvements are mainly the increased number of segments re-downloaded and missed (leading to a reduction in the ε_{retry} and ε_{fetch} efficiencies, respectively).

In contrast, using a longer duration segment (20 s) does not provide significant benefits. In comparison with 10s-long segments, only the active efficiency ($\varepsilon_{active} = 42.6$) and the switching-down efficiency ($\varepsilon_{down} = 47.58$) are slightly increased (less than 10%), while the rest of the metrics had equal values.

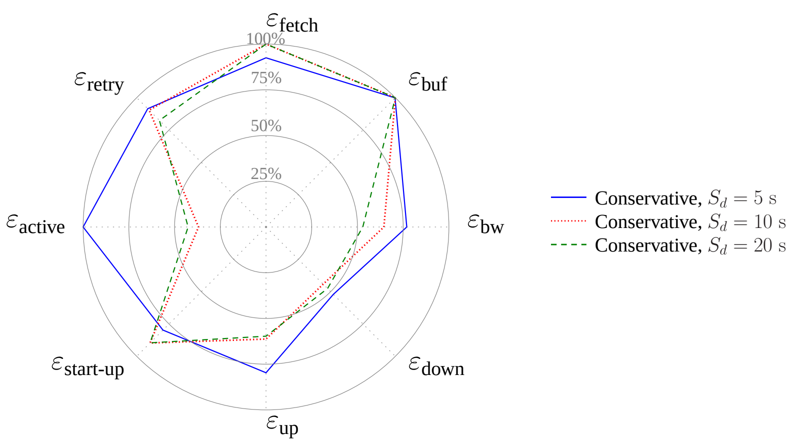


Figure 5.41: Graphical comparison under network scenario 4 for 5s-long, 10s-long, and 20-seconds long segments.

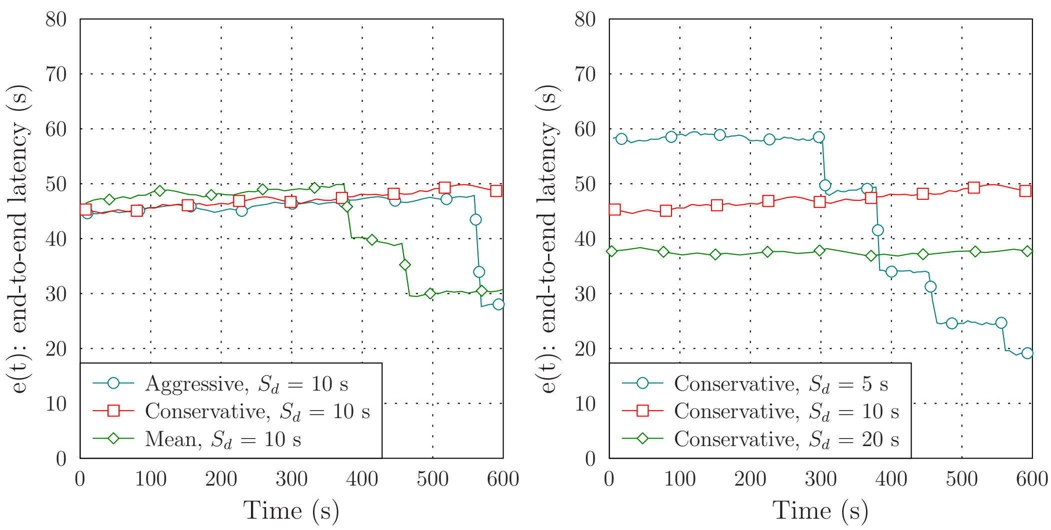
Table 5.16: Metrics comparison under network scenario 4 for 5s-long, 10s-long, and 20s-long segments.

S_d (s)	ε_{bw}	ε_{buf}	ε_{fetch}	ε_{retry}	ε_{active}	$\varepsilon_{start-up}$	ε_{up}	ε_{down}
5	76.97%	99.79%	92.5%	91.46%	100%	79.70%	79.69%	52.07%
10	64.45%	100%	100%	90.36%	36.9%	89.69%	61.26%	44.21%
20	52.92%	100%	100%	82.41%	42.6%	89.57%	59.70%	47.58%

5.5.3 Analysis of the end-to-end latency

Figure 5.42 depicts the end-to-end latency for the experiments performed under the fourth network scenario. As depicted in figure 5.42a, the three mechanisms performed equally in the first part of the experiment (from $t = 0$ s to $t = 300$ s), achieving a constant end-to-end delay. During the second part of the session (from $t = 300$ s to $t = 600$ s) the aggressive and mean mechanisms received several `HTTP-404` responses (as it can be seen in $t = 374$ s, $t = 456$ s, and $t = 558$ s), resulting in a reduction of the value of the $e(t)$ function.

Figure 5.42b represents the resulting end-to-end delay using different duration of segments. A shorter segment duration (5 s) produces more variation in the $e(t)$ function when troughs are longer than 20 s (as in the second part of the experiment, from $t = 300$ s). In contrast, choosing a segment duration of 10 s or 20 s does not affect the end-to-end latency during the whole session, meaning that segments are not missed and playback is not interrupted (the buffer is never empty when the next segment must be played).



(a) Different adaptation mechanisms

(b) Different duration of segments

Figure 5.42: End-to-end latency throughout the session for scenario 4.

5.5.4 Discussion

The fourth scenario was intended to test the behaviour of the adaptation algorithms under low-bandwidth-peaks of different durations. The three proposed mechanisms performed efficiently under this scenario, preventing playback interruptions and successfully downloading the media segments when they were available on the server. Results of these experiments indicated a major difference in the number of buffered segments during the session. The conservative mechanism always filled the buffer in time, whereas the aggressive and mean mechanism experienced a reduction in the buffer level during the second part of the experiments, when troughs were significantly longer. It is noteworthy that the mean algorithm consumed most of the session time in the active state, i.e., downloading segments, whereas the aggressive and the conservative mechanism spent a significant amount of time in the inactive state, which could be used to re-download segments which were already buffered at a higher quality.

Results regarding the effects of the size of the segments are very similar to the results of the third scenario (peaks in bandwidth). A shorter duration of the segments improves most of the metrics considered in our evaluation, whereas a longer duration of segments does not lead to any significant improvement over the reference duration (10 s).

5.6 Effects of packet loss

The effects of packet loss in the underlying network was measured through a set of four experiments where packets are lost according to a probability of error p_e: 5%, 10%, 15%, and 20%. Table 5.17 enumerates the input parameters which were fixed for the experiments carried out in this section.

In order to evaluate the effects of packet losses on playback, the conservative mechanism has been chosen since this algorithm achieved better buffering efficiency in the previous experiments. The duration of the segments was fixed at the reference value (10 s). The emulated bandwidth was constant at 2 Mb/s during the whole session (from $t = 0$ s to $t = 600$ s).

Table 5.17: Input parameters.

Mechanism	T (s)	R	bw_{avail} (Mb/s)	S_d (s)
Conservative	600	10	2	10

5.6.1 Impact on the metrics

Table 5.18 and figure 5.43 show the results of these experiments. Note that the reaction efficiencies have not been considered, since the network's available bandwidth is constant throughout the session (i.e., there are no bandwidth fluctuations that the client may detect.).

Table 5.18: Metrics comparison under different probability of packet losses.

p_e	ε_{bw}	ε_{buf}	ε_{fetch}	ε_{retry}	ε_{active}	$\varepsilon_{start-up}$
5%	51.17%	100%	100%	100%	41.80%	88.97%
10%	14.72%	95.18%	93.33%	70.66%	100%	83.71%
15%	5.66%	51.70%	55%	80%	100%	59.83%
20%	4.14%	54.64%	56.66%	98.66%	100%	55.02%

In general, increasing the probability of packet loss has a strong influence on all the metrics defined in our evaluation. ε_{bw}, ε_{buf}, ε_{fetch}, and $\varepsilon_{start-up}$ showed a clear dependency on p_e. The higher the probability of error, the lower these metrics are. The computed values for the active efficiency (ε_{fetch}) are rather simple to deduce: with low probability of error, the client is able to fetch the segments on time, switching to the inactive state when all available segments are downloaded. Increasing the probability of error leads to a more active client, as it must spend more time fetching the segments (due to packet losses, the HTTP transactions last longer).

Interesting results can be observed for the segment-retry efficiency (ε_{retry}). Up to $p_e = 10\%$ the efficiency decreases since more segments are being discarded. However, for $p_e = 15\%$ and

p_e = 20% the efficiency is increased. Although it seems to be an improvement, these values simply indicate that the segments are not being fully downloaded or are downloaded at the lowest quality level, therefore the client cannot discard them.

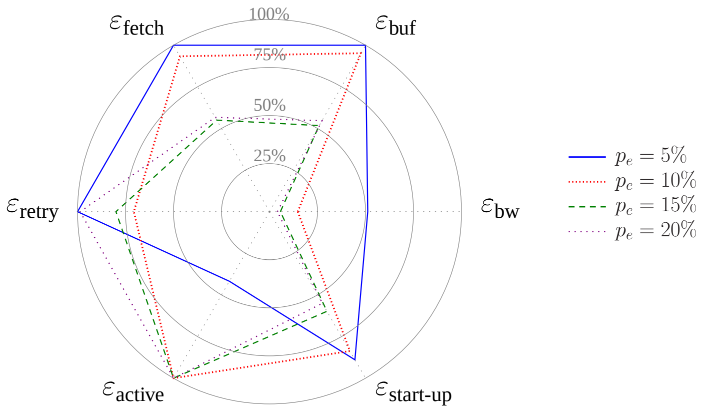


Figure 5.43: Graphical comparison under different probability of packet losses.

5.6.2 Discussion

Results of these experiments have shown that the player avoids playback interruptions and fetches the segments in time for packet loss rates of up to 10% of the network traffic transferred between the server and client. When the probability of error reaches 15% of network traffic, then the quality of service is drastically reduced, since the playback is interrupted 50% of the session time and half of the segments are not fetched in time. Therefore, in a 2 Mb/s channel, the client's application is able to maintain an acceptable quality of service with up to 10% probability of packet losses.

5.7 Evaluation with real live events

The performance of the client's application under real live scenarios is evaluated in this section. A survey has been carried out to determine which TV channels currently offer an HTTP-based alternative following any of the adaptive streaming protocols introduced in section 2.4. During this master's thesis project, there was no TV channel offering content using the MPEG-DASH protocol, although there were several using Apple-HLS and Microsoft-LSS.

The final match of the FIFA Women's World Cup (2011) was selected as the live event for this experiment. The match was broadcasted by the Eurosport channel. Eurosport provides free live content using Apple-HLS[5]. Table 5.19 summarizes the characteristics the offered media stream. Eurosport provides segments with a duration of 10 s (S_d = 10 s) and an available shifting time of 30 s (T_{shift} = 30 s), i.e., only three media segments are available at any point of time.

Table 5.19: Characteristics offered by the Eurosport channel over Apple-HLS.

S_d (s)	T_{shift} (s)	R	Avg. bit-rate (kb/s)	CODEC	Media container
10	30	1	900	H.264	MPEG-TS

The experiment was carried out from 21:00:33 to 23:05:10 on the 17th, July 2011, resulting in a session time of 02:04:37 (T = 7477 s). Listing 5.1 shows the extended M3U playlist offered by

[5] Eurosport's live channel can be found at `http://live.iphone.eurosport.com/uk1/stc_0_0.m3u8`.

the server. This playlist was updated every 10 s (as specified by the `#EXT-X-TARGETDURATION` tag). Since only one quality level is offered on the server's side (with a bit-rate of 900 kb/s), this experiment focused on the long-term behaviour of the client's application with a live event. In particular, the costs of the transcoding step (see section 4.1.1) in terms of the quality of service during playback was analyzed.

Listing 5.1: Extended M3U playlist retrieved from the Eurosport's HTTP server.

```
#EXTM3U
#EXT-X-TARGETDURATION:10
#EXT-X-MEDIA-SEQUENCE:
#EXTINF:10,
2011-07-17/VOA0/14/Media_20110717_19452220_19453220_0_0.mp4
#EXTINF:10,
2011-07-17/VOA0/14/Media_20110717_19453220_19454220_0_0.mp4
#EXTINF:10,
2011-07-17/VOA0/14/Media_20110717_19454220_19455220_0_0.mp4
```

5.7.1 Impact on the metrics

Table 5.20 and figure 5.44 show the results of the experiment on a real live event. The bandwidth, buffering, fetch, and retry efficiencies reached their maximum value (ε_{bw}, ε_{buf}, ε_{fetch}, and ε_{fetch} respectively), hence not a single segment was missed during the match and there were no interruptions in playback. The active efficiency (ε_{active}) was about 50%, meaning that the client was able to fetch the segments in time and spent a significant amount of time in the inactive state. The start-up efficiency ($\varepsilon_{start-up}$) indicates that the client is able to start playback rapidly even though media files needed to be converted into a format supported by Stagefright.

Table 5.20: Metrics comparison in a real live event.

ε_{bw}	ε_{buf}	ε_{fetch}	ε_{retry}	ε_{active}	$\varepsilon_{start-up}$
100%	100%	100%	100%	50.06%	74.67%

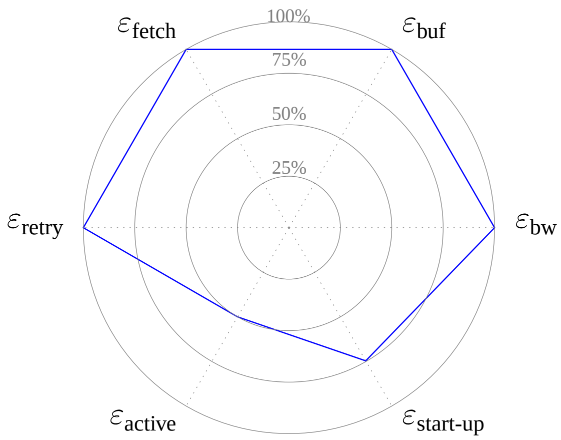


Figure 5.44: Graphical comparison of metrics in a real live event.

The average downloading time for the 10s-long segments throughout the whole session was 3.84 s (38.4% of the segment duration, S_d = 10 s), including the conversion. The average time consumed by conversion step was 2.41 s (24.1% of the segment duration, S_d = 10 s). Hence, the time required for the client to store Stagefright-compatible segments into the buffer was increased 1.43 s on average (14.3% of the segment duration, S_d = 10 s).

5.7.2 Discussion

This experiment was intended to evaluate the behaviour of the client's application with a real live event. Results show that the quality of service provided on the client's side is very high since there were no interruptions or missed fragments. The client's application consumes a significant amount of time converting media files. However, this procedure was performed in the background sufficiently quickly that the quality of playback was not decreased.

Chapter 6

Conclusions

"The world always seems brighter when you've just made something that wasn't there before."

– Neil Gaiman

In this chapter the findings of this master's thesis project are presented. Section 6.1 presents a discussion of the results achieved in the previous chapter. Section 6.2 enumerates some of the possible improvements to this project which should be considered in future works.

6.1 Discussion

In this master's thesis project a full service has been proposed in order to evaluate the benefits of different adaptive streaming mechanisms using HTTP as a delivery protocol. Three mechanisms were proposed to provide bit-rate adaptation on the client's side: the *aggressive*, *conservative*, and *mean* algorithms.

Results of the experiments with heterogeneous network scenarios have shown that the adaptation mechanisms efficiently utilize the available bandwidth of the network. The aggressive mechanism produces an adequate adaptation to short-term bandwidth fluctuations, although this mechanism increases the probability of discarded and missed segments due to bandwidth underestimation. The conservative mechanism prevents playback stops, at the cost of a non-optimal utilization of available bandwidth in the short-term, although experiments have shown that bandwidth utilization is equal to that of the aggressive mechanism in the long-term. The mean mechanism presents a similar performance to the aggressive mechanism, although it consumes more time downloading segments.

Major differences can be seen in the level of activity of the media player. While using the mean algorithm the player remains mostly in the active state throughout the session, the aggressive and conservative mechanisms spent a considerable time in the inactive state. These inactive intervals could have been used to re-download some segments at a better quality, hence improving the overall use of the available bandwidth.

The reaction times to switch between media bit-rates were significantly better when the available bit-rate was increased rather than reduced, since segments were downloaded faster. As a consequence, the throughput measurements occur earlier and the adaptation mechanisms decide upon the next appropriate level of quality sooner.

Reducing the size of the segments improves the reaction times to variation of the network bit-rate, at the cost of increased activity by the client. More time is consumed sending and receiving HTTP messages, hence the probability of missing a segment, i.e., the probability of receiving a `HTTP-404` response, is increased. The bit-rate adaptation is also improved for shorter segments since the bandwidth measurements occur more often. The opposite situation happens with larger segments, hence the network's available bandwidth is used less efficiently since measurements are taken less frequently. In addition, the switching operations are performed significantly later. However, downloading longer segments leads to a more inactive client, since fewer segments need to be downloaded in order to play the media content. The client spends more time in the inactive state, which could be used to perform other actions, for example, to fetch segments at higher bit-rates.

Finally, experiments indicate that the client's application is able to maintain an acceptable quality of service with up to 10% packet losses in the underlying network.

6.2 Future work

This section explores some of the limitations and assumptions made in the prototype developed for this master's thesis project, in order to be improved in future works.

For simplicity, the algorithms defined in section 4.4.2 are only invoked when segments are fully downloaded, thus restricting the adaptation procedure to the boundaries of segments. This leads to infrequent adaptation in the case of larger segments (as shown throughout the evaluation in the previous chapter). The algorithms use the throughput calculated for each media segment as an input parameter. The throughput measurement could be computed several times in the case of larger segments, in order to more frequently inform the adaptation algorithms about the characteristics of the underlying network.

The conservative mechanism can be improved by always fetching the next lower level (than the current long-term bandwidth allows) instead of multiplying the throughput measurement by a sensitivity parameter (see algorithm 2). Then, if there is additional time fetch the delta to this lower level that would bring up the quality level. Thus, there will be always segments to play, improving the quality when there is extra bandwidth. This improvement meets the first and third requirements defined in section 4.4.2, while approximating the second requirement.

In section 4.4.6 a skipping mechanism was defined to discard media segments upon reduction of the network's available bandwidth to re-download segments at a lower quality level. This procedure considers a fixed fraction of the segment duration (a downloading timeout is defined as the 80% of the segment duration), rather than a time estimated by subtracting the estimated download time from the segment duration. This estimated download time could be computed in terms of median observed download time or maximum observed time. Improvements to the re-download mechanism would lead to a more efficient use of the network resources, since the bytes received during the HTTP transaction are simply discarded and never played by the client.

The MPEG-DASH protocol allows clients to request a single range of bytes of the media segments, as HTTP/1.1 supports the transmission of a partial entity-body by means of the `Content-Range` header field. This feature has not been considered in our prototype, instead, the full body of the HTTP responses is always assumed to be transferred. Requesting partial segments (*subsegments*) could be useful to improve the re-download mechanism and to enable faster switching between rates.

In our client's application, segments which are stored in the buffer are played sequentially. This leads to a reduction of the end-to-end delay function (as it was defined in section 5.1.6) if the client is not able to fetch some segments on time, since the next segment will be played without synchronization with the server. Segments do not have to be played sequentially. The specification of the MPD in the MPEG-DASH standard allows the clients to determine the time-stamp at which the segments have to be played. Since server and client are synchronized with an external NTP server, this feature could be added to the prototype, improving the stability of the end-to-end delay function presented in the evaluation chapter.

Apple-HLS is natively supported on Android 3.0 onwards (section 2.10.2 provides a summary of the adaptive protocols supported on Android). It would be interesting to evaluate the capabilities of the official implementation of Apple-HLS on Android OS, in order to determine the behaviour of the adaptation mechanism developed by Google and potentially providing a comparison with the results achieved in our evaluation.

The evaluation presented in this work does not cover situations where the client disconnects from the streaming server and restarts the connection (for instance, due to a network failure). The adaptation mechanisms could be improved by storing information about the previous state of the network, in order to avoid the repetition of the switching operations at the beginning of the session. Additionally, other reliable transport protocol such as SCTP could be used instead of TCP. SCTP can improve the measured throughput [60] and provides better protection against Denial of Service attacks.

Finally, it would be interesting to study the scalability of the system. Scalability could be improved if a *caching proxy* is displaced between clients and servers. This caching proxy would store the most frequent content requested by the end-users. Once a client requests a media stream, following clients which request the same content will experience better throughput measurements, reducing the influence of the selected adaptation mechanism.

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Appendix A

Demonstration of the client's application

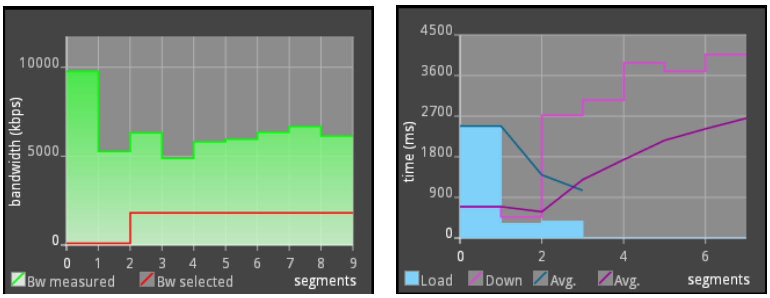
This appendix explains in more detail the features of the client's application. Section A.1 and section A.2 introduce the graph generator and the logging system. Section A.3 presents the elements of the client's graphical user interface (GUI).

A.1 Graph generator

The client's GUI shows several graphs which are dynamically generated during the streaming session. These graphs are created with AndroidPlot [9], a Java API designed exclusively for the Android platform[1]. Two types of graphs are available in our prototype (see figure A.1):

Bandwidth graph Compares the measured bandwidth (green plot) with the selected media bit-rate (red plot).

Download graph Represents the segments' downloading time (pink plot) and the segments' loading time (blue bars). The average of each plot is calculated and depicted in purple and dark blue colours, respectively. Note that the first blue bar corresponds to the start-up delay, whereas successive bars illustrate the time it takes to load the media segment into the *fake* player.



(a) Sample bandwidth graph.

(b) Sample download graph.

Figure A.1: Available graphs in the client's application.

A.2 Logging system

The client's application creates several plain-text files (*logs*). These files are updated throughout the session for each measurement (segments' downloading times, measured throughput, selected media bit-rate, size of the buffer...), as defined in section 5.1.6 on page 66. At the end of the session, a summary file is generated. Listing A.1 shows an example:

[1] A feature comparison of graph libraries for Android can be found at `http://androidplot.com/wiki/Feature_Comparison`.

Listing A.1: Summary log file.

```
# Session log file
# Starting at 20110709T210900
# Model: GT-S5830
# Brand: Samsung
# Id: FROYO
# Display: FROYO.XWKA9
# 602.482 END SESSION. TOTALS:
# HTTP requests (manifest): 138
# HTTP requests (segments): 78
# Start-up delay: 8.802 s
# Total pause (excluding initial pause): 0.0 s
# Segments played: 70
# Segments missed (404 errors): 1
# Segments skipped: 0
# Segments skipped (due to FFmpeg error): 0
```

A.3 Overview of the client's GUI

The following subsections explain the features of the client's GUI.

A.3.1 Adding media sources

Media sources can be added into the list using the *options menu* (see figure A.2a). *New manifest URL* opens a new text dialog, where the URL of the manifest can be inserted (figure A.2b). The list of media sources is stored in a private database (i.e., only accessible from the client's application) using SQLite [68], since it is fully supported on Android.

(a) Options menu. **(b)** New manifest dialog. **(c)** List updated.

Figure A.2: Adding media sources.

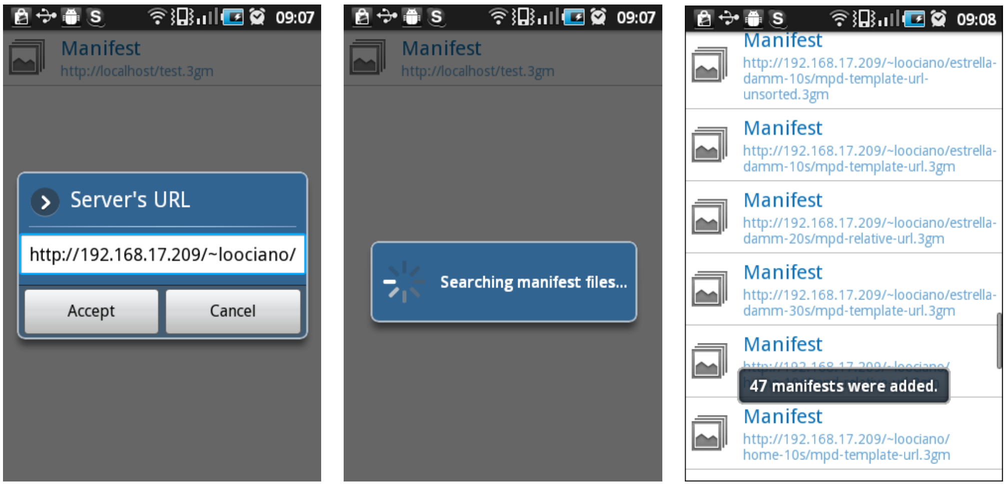
The application checks the existing entries on the database to avoid duplicate URLs. If the inserted URL was not registered on the database, the list of media sources is updated with the new URL (figure A.2c). The application distinguishes between MPEG-DASH and Apple-HLS sources (`.3gm` and `.m3u8` file extensions, respectively), displaying them with different icons.

A.3.2 Importing multiple media sources

Multiple URIs can be added using the option *Import list*. This option opens a new dialog to select the file which contains a list of URIs. This list is parsed and the database is updated.

A.3.3 Searching for media sources

The option *Search manifests* opens a new dialog to specify a web server's URI (figure A.3a). The client's application performs an HTTP request and parses the HTTP response, searching for MPEG-DASH or Apple-HLS sources, i.e., URIs with `.3gm` or `.m3u8` extensions (figure A.3b). If so, URIs are added to the list of sources (figure A.3c).



(a) Text dialog. **(b)** Loading widget. **(c)** List updated.

Figure A.3: Searching for media sources.

A.3.4 Modifying and deleting media sources

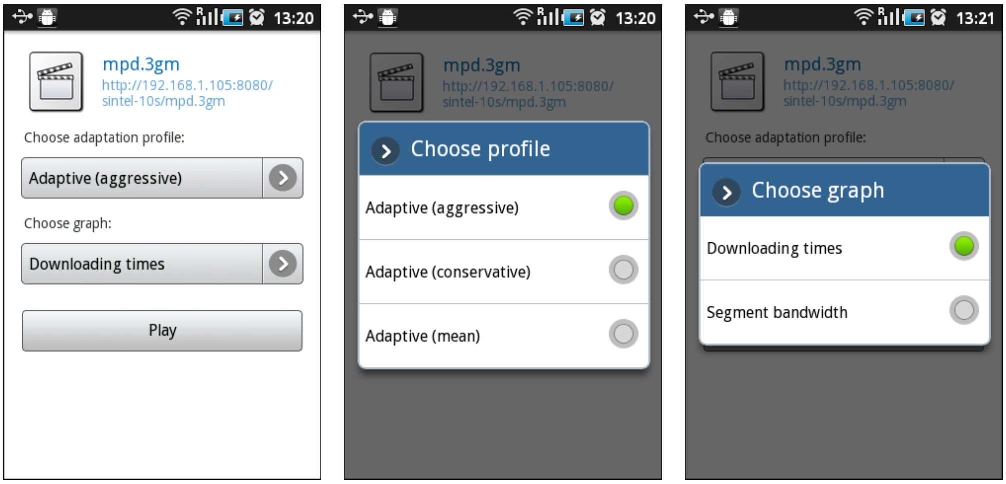
Media sources can be modified of deleted using the *context menu* as shown in figure A.4a. The context menu is triggered by long-pressing any element of the list. Figure A.4b shows how to delete all the media sources. This action is followed by a confirmation dialog (figure A.4c).

(a) Context menu. **(b)** Options menu. **(c)** Confirmation dialog.

Figure A.4: Modifying and deleting media sources.

A.3.5 Opening a media source

If a media source is selected from the list, the application shows the *Settings Activity* (figure A.5a). There are two parameters that can be selected in this view: (1) the adaptation mechanism and (2) the graph to be displayed on the screen. Note that the displayed graph can be switched during playback, but the adaptation mechanism can not be changed once the streaming session is started (*play* button).



(a) Settings Activity. **(b)** Adaptation dialog. **(c)** Graph dialog.

Figure A.5: Selection of the session parameters.

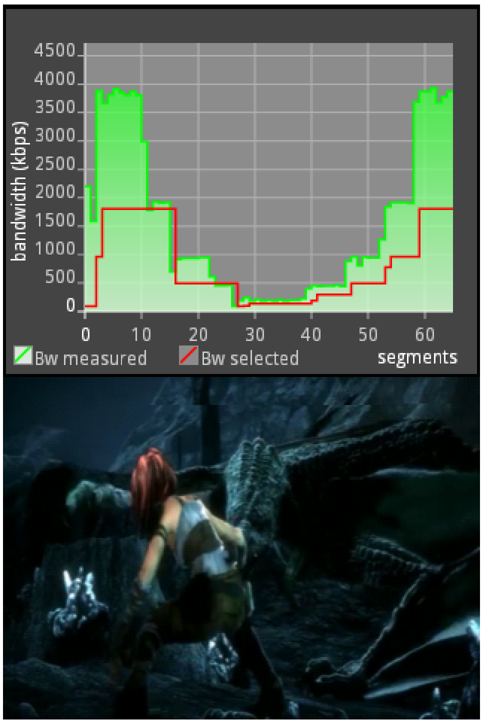
A.3.6 Playback during the streaming session

Figure A.6a shows the state of the graph at the beginning of the streaming session, when the first segment has been downloaded. The graph is updated when successive segments are downloaded, as it is shown in figure A.6b and figure A.6c.

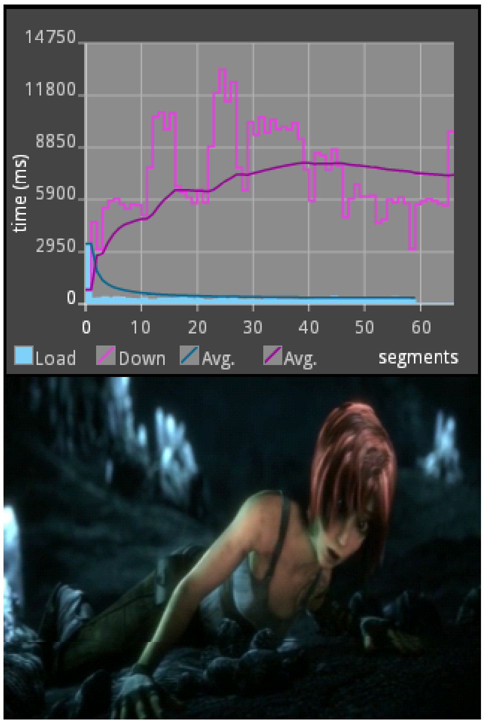
(a) Playing segment 1. **(b)** Playing segment 2. **(c)** Playing segment 3.

Figure A.6: Dynamic graphs.

Figure A.7 to A.10 show the client's application during the performance evaluation over the scenarios 1 to 4 (defined in section 5.1.7 on page 70).

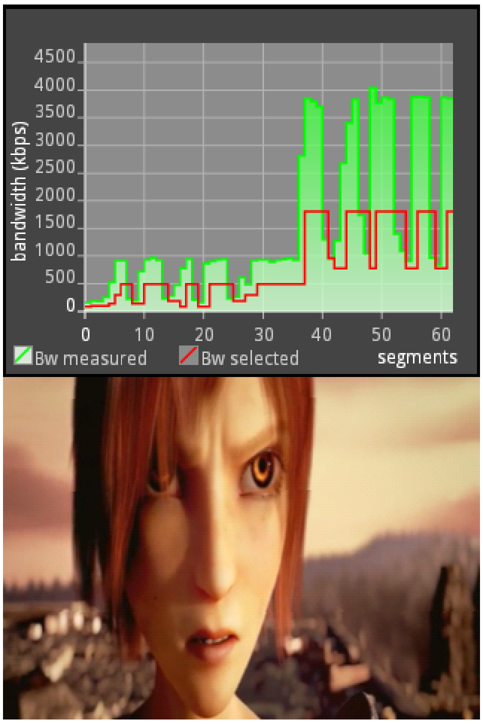


(a) Displaying bandwidth graph.

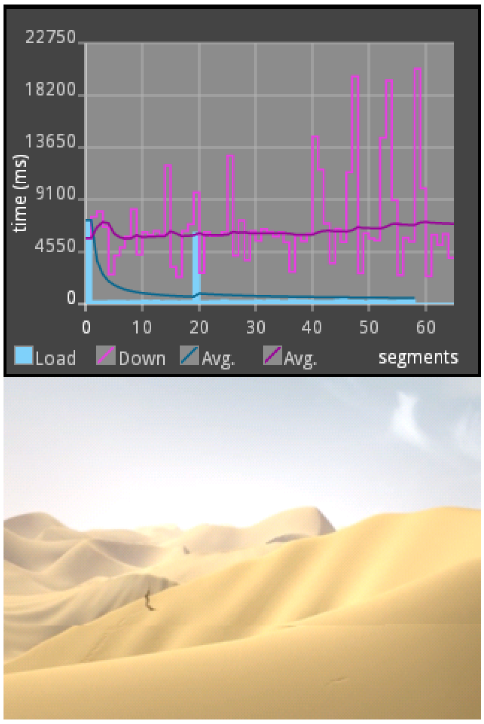


(b) Displaying download graph.

Figure A.7: Sample of playback using the conservative mechanism over the scenario 1.



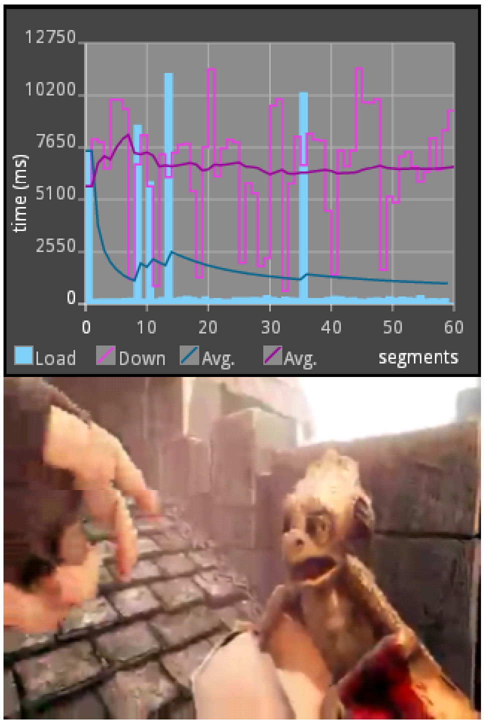
(a) Displaying bandwidth graph.



(b) Displaying download graph.

Figure A.8: Sample of playback using the conservative mechanism over the scenario 2.

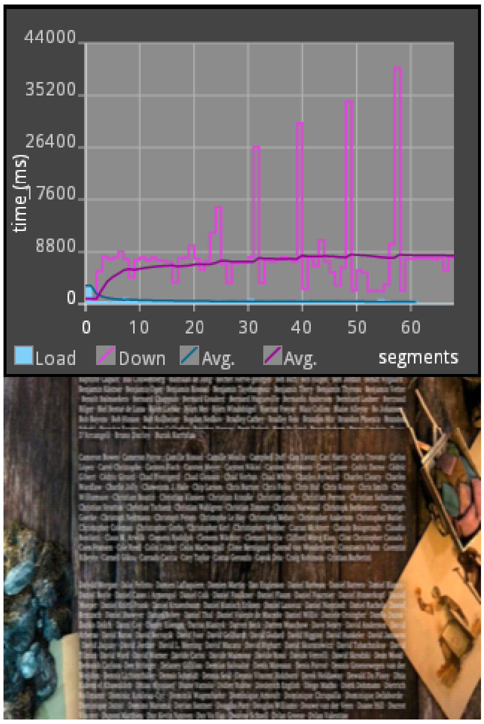
(a) Displaying bandwidth graph.



(b) Displaying download graph.

Figure A.9: Sample of playback using the conservative mechanism over the scenario 3.

(a) Displaying bandwidth graph.



(b) Displaying download graph.

Figure A.10: Sample of playback using the conservative mechanism over the scenario 4.

Note that the blue bars depicted in figure A.8b and A.9b represent interruptions on playback due to buffer underflow. If so, the client's GUI displays the last video frame on the screen and a *loading wheel animation* until the next media segment starts playing.

Appendix B

FFmpeg capabilities

This appendix includes further information about the transcoding capabilities of the FFmpeg libraries. Tables B.1 to B.3 summarize the supported audio/video CODECs and container file formats.

Table B.1: FFmpeg supported audio CODECs. Extracted from [13].

Name	Encoding	Decoding	Details
AAC	Yes	Yes	Encoding supported through external library `libfaac`
MP3 (MPEG audio layer 3)	Yes	Yes	Encoding supported through external library LAME, ADU MP3 and MP3 on MP4 also supported
Vorbis	Yes	Yes	A native but very primitive encoder exists

Table B.2: FFmpeg supported container formats. Extracted from [13].

Name	Encoding	Decoding	Details
Flash (SWF)	Yes	Yes	
Flash Video (FLV)	No	Yes	Macromedia Flash video files
MOV/QuickTime/MP4	Yes	Yes	3GP, 3GP2, PSP, iPod variants supported
MPEG-TS (transport stream)	Yes	Yes	Also known as DVB Transport Stream
MPEG-4	Yes	Yes	MPEG-4 is a variant of QuickTime
Ogg	Yes	Yes	
Raw H.263	Yes	Yes	
Raw H.264	Yes	Yes	
Raw MPEG-2	No	Yes	
Raw MPEG-4	Yes	Yes	
WAV	Yes	Yes	

Table B.3: FFmpeg supported video CODECs. Extracted from [13].

Name	Encoding	Decoding	Details
Flash Video (FLV)	Yes	Yes	Sorenson H.263 used in Flash
H.263 / H.263-1996	Yes	Yes	
H.263+ / H.263-1998 / H.263 version 2	Yes	Yes	
H.264 / AVC / MPEG-4 AVC / MPEG-4 part 10	Yes	Yes	Encoding supported through external library `libx264`
MPEG-2 video	Yes	Yes	
MPEG-4 part 2	Yes	Yes	
VP8	Yes	Yes	fourcc: VP80, encoding supported through external library `libvpx`
Theora	Yes	Yes	Encoding supported through external library `libtheora`

Appendix C

Integration of FFmpeg libraries using the Android NDK

This appendix explains in more detail the process of integration of the FFmpeg libraries into the client's application for Android. Listing C.1 shows the configuration steps that are needed to compile the FFmpeg libraries for the ARM architecture[1] (see the specifications of the client's device in table 5.1 on page 62). The `./configure` command (listing C.1, line 12) simply specify the flags which enable or disable the features provided by the libraries. Note that the most important arguments are `--arch=arm` and `--enable-cross-compile` which allow the compilation for the ARM architecture. Additional flags are provided by the arguments `--extra-cflags` and `--extra-ldflags`.

Listing C.1: Integration script.

```
1 #!/bin/bash
2 # Path to the prebuild directory of the Android NDK
3 PREBUILT=/usr/share/android-ndk/toolchains/arm-eabi-4.4.0/prebuilt/linux-x86
4
5 # Path to the platform directory of the Android NDK
6 PLATFORM=/usr/share/android-ndk/platforms/android-8/arch-arm
7
8 # Update PATH
9 export PATH=$PREBUILT/bin:$PATH
10
11 # Configure the FFmpeg libraries with the following command line
12 ./configure --target-os=linux \
13    --arch=arm \
14    --enable-version3 \
15    --enable-gpl \
16    --enable-nonfree \
17    --disable-stripping \
18    --disable-ffmpeg \
19    --disable-ffplay \
20    --disable-ffserver \
21    --disable-ffprobe \
22    --disable-encoders \
23    --disable-muxers \
24    --disable-devices \
25    --disable-protocols \
26    --enable-protocol=file \
27    --enable-avfilter \
28    --disable-network \
29    --disable-mpegaudio-hp \
30    --disable-avdevice \
31    --enable-cross-compile \
32    --cc=$PREBUILT/bin/arm-eabi-gcc \
33    --cross-prefix=$PREBUILT/bin/arm-eabi- \
34    --nm=$PREBUILT/bin/arm-eabi-nm \
35    --extra-cflags="-fPIC -DANDROID" \
36    --disable-asm \
37    --enable-neon \
38    --enable-armv5te \
39    --extra-ldflags="-Wl,-T,$PREBUILT/arm-eabi/lib/ldscripts/armelf.x -Wl,
```

[1] More information can be found at http://www.arm.com.

```
   -rpath-link=$PLATFORM/usr/lib -L$PLATFORM/usr/lib
   -nostdlib $PREBUILT/lib/gcc/arm-eabi/4.4.0/crtbegin.o
   $PREBUILT/lib/gcc/arm-eabi/4.4.0/crtend.o -lc -lm -ldl"

# Compile the FFmpeg libraries using the Android NDK
ndk-build
```

Listing C.2 shows the `Makefile` utilized in our prototype to generate the static and dynamic libraries loaded by the Java application.

Listing C.2: Android makefile (`Android.mk`).

```
LOCAL_PATH := $(call my-dir)
include $(CLEAR_VARS)
...
LOCAL_CFLAGS := -D__STDC_CONSTANT_MACROS
...
LOCAL_C_INCLUDES += \
    $(LOCAL_PATH)/../libffmpeg \
    $(LOCAL_PATH)/../include

LOCAL_SRC_FILES := \
    onLoad.cpp \
    com_media_ffmpeg_FFMpegAVFrame.cpp \
    com_media_ffmpeg_FFMpegAVInputFormat.c \
    com_media_ffmpeg_FFMpegAVRational.c \
    com_media_ffmpeg_FFMpegAVFormatContext.c \
    com_media_ffmpeg_FFMpegAVCodecContext.cpp \
    com_media_ffmpeg_FFMpegUtils.cpp

LOCAL_SRC_FILES += \
    com_media_ffmpeg_FFMpeg.c \
    ../libffmpeg/cmdutils.c

LOCAL_LDLIBS := -llog

LOCAL_SHARED_LIBRARIES := libjniaudio libjnivideo
LOCAL_STATIC_LIBRARIES := libavcodec libavformat libavutil libpostproc libswscale

LOCAL_MODULE := libffmpeg_jni

include $(BUILD_SHARED_LIBRARY)
```

Trivia

"I can't go to a restaurant and order food because I keep looking at the fonts on the menu."

– Donald Knuth

- This master's thesis report is written entirely on LaTeX 2_ε. It uses a template from the Royal Institute of Technology (KTH), available from: `system.csc.kth.se/misc/tex`.
- The following Latex packages were used: `graphicx`, `hypens`, `hyperref`, `parskip`, `subfig`, `colortbl`, `xcolor`, `multirow`, `tabulary`, `longtable`, `listings`, `caption`, `minted`, `algorithm2e`, `fourier`, and `epstopdf`. All of them are available via `ctan.org`.
- The Android logo is published by Google Inc. under the terms of the Creative Commons Attribution (CC-A) license. According to the brand guidelines: *"the android robot can be used, reproduced, and modified freely in marketing communications.* `http://www.android.com/branding.html`.
- All figures (except screenshots of Sintel) were drawn in vectorial format (SVG) using Inkscape (available from `inkscape.org`). Imported into Latex in EPS format. The Latex package `epstopdf` eases the integration.
- Plots were generated with the Graphics Layout Engine (GLE), available from `glx.sourceforge.net`.
- Source code listings were highlighted with the `minted` Latex package, available from `ctan.org/tex-archive/macros/latex/contrib/minted`. It is based on the powerful Pygments library, available from `pygments.org`.
- The word cloud showed at the end of this report was created with Wordle (available from `wordle.net`), based on the most repeated words of this document.
- This master's thesis project was presented on 27 September 2011 at KTH (Kista campus, Hörby seminar room).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conservative
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evaluation
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end-to-end
switching
devices
must
information
Video
Mb/s
used
mean
Sd
private
metrics
selected
rate
Bandwidth
H.264
servers
activity
player
bit-rate
possible
stream
Protocol
protocol
parameters
algorithm
supported
method
format
streams
MPEG-DASH
on-demand act
throughput
container
efficiency
comparison
encoding
bw
Buffer
performed
playback
utilization
playlist
times
listing
Apple
behaviour
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thesis
URL
size
Network
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Performance
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several
shown
bwavail
function
system
application
Apple-HLS
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files
provides
start
NTP
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void
manifest
next
following
packet
CODECs
new
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order
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Java
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