Multimedia Content Adaptation for Internet Protocol Television Services in the IP Multimedia Subsystem

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Declaration

I declare that the above thesis is my own unaided work, both in concept and execution, and that apart from the normal guidance from my supervisor, I have received no assistance except as stated in the text of this document.

This work is being submitted for the Master of Science Degree in Electrical Engineering at the University of Cape Town. Neither the substance nor any part of the above thesis has been submitted in the past, or is being, or is to be submitted for a degree at this university or at any other university.

Robert John Marston

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2008

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Synopsis

Internet Protocol Television (IPTV) is an umbrella term for a number of multimedia services deployed over an IP infrastructure. IPTV is expected to be a major application in the next-generation of IP based networks and will grant service providers capabilities and opportunities through the inter-working of IPTV services with other IP services such as presence, instant messaging (IM) and voice over IP (VoIP). Many operators are currently deploying, or investigating, the deployment of IPTV services to counteract dwindling voice services revenues by opening up new revenue streams and attracting customers to their networks. Consumer interest in IPTV services is currently being fuelled by Video on Demand (VoD) IPTV services, which provide users with access to a potentially large range of content including Hollywood titles and TV network series. IPTV services provide a highly interactive viewing experience and go beyond what current broadcast television services can offer.

Given the recent interest in IPTV a number of standardisation bodies have defined end-to-end, IPTV architectures. These include ATIS's IPTV Interoperability Forum (ATIS-IIF), ETSI TISPAN and the ITU's IPTV focus group. Generally two IPTV architectures are defined: The first architecture makes used of existing protocols and systems in order to deliver IPTV services while the second is based on next generation networks, such as the IP Multimedia Subsystem (IMS).

The IP Multimedia Subsystem (IMS) is an architectural subsystem for the control and provisioning of IP multimedia services over an all-IP, packet based core. The IMS was first standardised by the 3rd Generation Project Partnership but it has since been adopted by other standardisation bodies, including 3GPP2, ATIS, ITU-T and ETSI-TISPAN, who contribute to its development. The IMS promises a scalable, integrated platform that enables the creation of new services and facilitates the convergence of telecommunications and Internet services. The IMS also promises to achieve fixed mobile convergence by enabling the seamless distribution of services over fixed and mobile broadband networks. However, this requires that IMS services need to be able to cater for heterogeneous access networks and a wide range of user end-terminal capabilities. This presents a significant challenge for IPTV services as the adaption of video content is no easy task.

There are numerous ways in which adaption of multimedia content can be achieved; these include precoding, scalable video encoding (SVC), Multiple Description Encoding (MDC) and/or trans-coding. This thesis involves an investigation into which of these techniques are best suited for use in IMS-based IPTV services with each adaptation method evaluated in terms of its relative strengths and weaknesses. Precoding has low processing requirements for the adaption of video but requires excessive amounts of disk storage; this is to ensure sufficient catering for the diverse range of video bit-rates and capabilities of user end-terminals. SVC and MDC adaptation techniques are highly scalable but require widespread adoption from industry of a specific SVC or MDC coding format; since only a handful of devices support these codecs, SVC and MDC based content adaption methods will be of little use in present IMS-IPTV deployments. Trans-coding is highly flexible in adapting video content in order to cater for varying user and session requirements but does not scale well, due to its high processing and memory requirements. A hybrid scheme is then proposed in this thesis that leverages the advantages of precoding and trans-coding, with a goal of enabling more efficient adaption of content in IMS-IPTV systems. A proof of concept IMS-IPTV test-bed was developed to demonstrate the benefits of the proposed hybrid content adaptation scheme and provided a good platform for several evaluations to take place. The test-bed and its implementation is described in order to enable duplication of it and the results obtained during this thesis; as well as to enable extension of the test-bed for future research. Findings from test-bed show that the proposed hybrid scheme does enable more efficient adaptation of content and that integrating the scheme will benefit operators deploying IMS. Future work includes extending the test-bed architecture to allow for dynamic adaptation to be performed and optimisation of the trans-coding process.

Contents

List of Figures

List of Tables

List of Acronyms

2G - 2nd Generation

3G - 3rd Generation

3GPP - 3rd Generation Partnership Project

AIN - Advanced Intelligent Network

API - Application Programming Interface

ATIS - Alliance for Telecommunications Industry Solutions

CDMA - Code Division Multiple Access

CPU - Central Processing Unit

CoD - Content on Demand

CSCF - Call Session Control Function

CSV – Comma Separated Values

DRM - Digital Rights Management

DVB - Digital Video Broadcasting

EPG - Electronic Program Guide

ETSI - European Telecommunications Standards Institute

GPRS - General Packet Radio Service

GSM - Global System for Mobile communications

HD - High Definition

IM - Instant Messaging

IMS - IP Multimedia Subsystem

IETF - Internet Engineering Task Force

IN - Intelligent Network

IP - Internet Protocol

IPTV - IP Television

ITU - International Telecommunications Union

LD - Low Definition

MCF - Media Control Function

MDC - Multiple Description Coding

MDF - Media Delivery Function

MPEG - Motion Picture Experts Group

NGN - Next Generation Network

OSA - Open Service Architecture

QoE - Quality of Experience

QoS - Quality of Service

RFC - Request for Comments

RTP - Real-time Transport Protocol

RTSP - Real Time Streaming Protocol

SCF - Service Control Function

SD - Standard Definition

SDF - Service Discovery Function

SDO - Standards Definition Organisation

SDP - Service-Delivery Platform or Session Description Protocol

SIP - Session Initiation Protocol

SSF - Service Selection Function

SVC - Scalable Video Coding or Scalable Video Codec

TISPAN -Telecoms & Internet converged Services & Protocols for Advanced Networks

TS - Transport Stream

UE - User Equipment

UPSF - User Profile Selection Function

URI - Universal Resource Indicator

VoD - Video on Demand

VoIP - Voice over IP

XCAP - XML Configuration Access Protocol

XML - Extensible Markup Language

Chapter 1

Introduction

The Internet has revolutionised modern living with an estimated 1.5 billion [6] users making daily use of Internet services to conduct business, do research, shop for bargains, socialise and more.

Services have been the key to the success of the Internet with electronic mail and the World Wide Web being the first two Internet services to have found widespread use in both the business world and in the home. With the advancements in Internet access technologies and computer processing, a number of new services have begun to emerge to take advantage of these new developments including, Voice over IP (VoIP), Internet Protocol TV (IPTV), presence and social networking.

The use of open protocols has played a large role in the success of Internet services; anyone interested in developing Internet applications can find the necessary information and tools available for free on the Internet. There are also numerous books and publications devoted to the subject, as well as on-line communities and forums that allow developers at no cost the opportunity to gain insight from experts across the globe. All this has enabled, with relative ease and minimal cost, to create and deploy new and innovative services for use by the Internet community as a whole.

The success of the Internet and its services has been so much so that many traditional telecommunication operators have deployed IP access technologies allowing connectivity to the Internet and thus access to its services. With the increased proliferation of broadband networks, more and more businesses and individuals are starting to make use of sophisticated, low-cost, third-party Internet communications services such as VoIP and instant messaging (IM). The result of this has been the steady erosion of traditional voice services revenues and the possibility of operators being reduced to mere bit-pipes, while the bulk of user revenues are taken by third-party service providers. The telecommunication industry now finds itself at a crossroads as it tries to regulate IP services while at the same time attempting to foster the same level of service innovation as seen with the development of Internet services.

1.1 Evolution of service architectures in the telecommunications industry [1]

In the past the development of telecommunication services was complex and time-consuming. New services had to be implemented directly within the core switch systems, with each new service requiring the development of a corresponding architecture which describes the various functional components as well as interfaces and signalling protocols used to interconnect these components. Each service thus had its own infrastructure and functional systems, which were to a large extent independent of other services. This lead to the so called 'silo' or 'stovepipe' architecture, see Figure 1.1, as the operator's network became a clutter of overlaid service architectures and components. Furthermore development of these new services was costly due to the requirement of specialized telecoms engineers and lengthy time periods needed to develop the service.

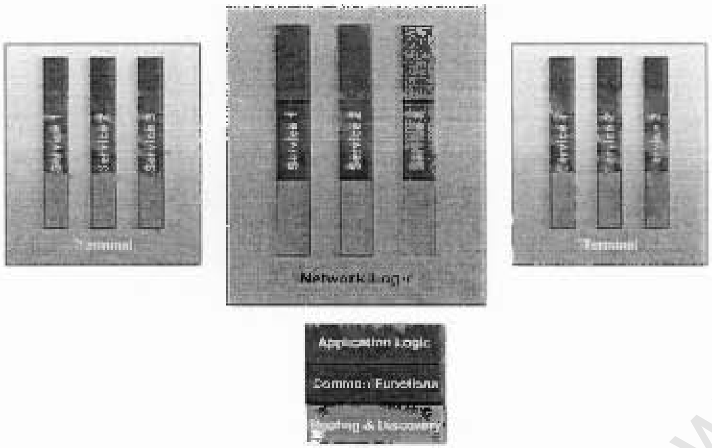

Figure 1.1: 'Silo' or 'Stovepipe' architecture

In the early 1980's the competitive need for shorter development times and the potential of the blooming IT industry led to the concept of Service-Delivery Platforms (SDPs). The idea behind these SDPs was to provide an extensible set of reusable components allowing for the rapid and efficient implementation of new services within an operator's network. In addition to this standardized components and interfaces would enable network operators independence from vendor specific service solutions and proprietary network technologies.

The first major shift was the International Telecommunications Union's (ITU) standardization of the Intelligent Network (IN), which was aimed at providing a more flexible way of developing new telecommunications services. IN defined an overlay service architecture on top of the Signalling System 7 (SS7) network and extracted the service intelligence from the legacy network switches into dedicated central service control points. By defining a number of Service Independent Building Blocks (SIBs), which can be chained together in order to form newer and more sophisticated services, the IN network allowed to some extent service independence. These SIBs allowed operators to develop value-added services for their own networks and without the need to request the core switch manufacturer to develop the new service. Other variants of the IN platform were standardized and deployed, including the Advanced Intelligent Network (AIN) in the United States as well as European and Japanese variants, owing to the SS7 standards differing slightly from region to region. There are also IN services for mobile networks which were standardized under the name Customized Applications for Mobile Enhanced Logic (CAMEL), maintained by the 3rd Generation Partnership Project (3GPP). Although many network operators have deployed IN, AIN and CAMEL services around the globe, they have not developed into the open market services the IN community initially envisaged. This can be attributed to the IN's IT systems being too specific and not following mainstream programming practices. Furthermore, the telecoms industry was at the time still very closed and monopolistic in nature.

In the mid-1990s Application Programming Interfaces (APIs) were introduced. These APIs were based on object-oriented and middle ware technologies such as C++ and Java. Telecoms API standards such as Parlay, the 3GPP Open Service Architecture (OSA), and the Java APIs for Integrated Networks (JAIN) were aimed at making telecoms service implementations easier then IN implementations by abstracting the signalling protocol details from the underlying networks. The APIs feature telecoms-related capabilities such as call control, conferencing, messaging, etc.

The concept of APIs was to host the application logic on application servers which would be able to access the provider's service gateway via a secure network connection. The network operator would then be offer services from its own application servers or make use of a third party's services through service-level agreements. Both network operators and third parties would benefit as the network operators could provide customers with a wider variety of services, while the third parties would receive revenue for access to their services. It was also possible to add new API operations and interfaces to cope with market trends and newer technologies. Framework interfaces were also defined to allow registration and discovery of new service interfaces and perform the necessary service-level agreements.

However, the uptake of APIs has been slow with network operators not wishing to open up their networks to third parties and due to these APIs being still far too complicated for use by developers unfamiliar with the telecoms architecture.

In the year 2000, the Parlay Group noted how, through the use of open protocols, the Internet had enabled its own open services market. The Parlay Group then developed a simplified version of the Parlay/OSA APIs, named Parlay X, which was a simplified API based on Web-services technologies. Parlay X was much simpler to use then previous APIs and enabled developers familiar with creation of Web-services to construct new and innovative telecommunication services. Today, most Parlay/OSA platforms feature a Parlay X gateway for exposing telecoms interfaces to third parties, while operators still use the classic Parlay/OSA APIs for the implementation of internal operators services on top of a rapidly changing network architecture.

More recently the IP Multimedia Subsystem (IMS) has emerged and promises a scalable, integrated platform to support the creation of new services as well as a means to facilitate the convergence of telecommunication and Internet services.

1.2 The IP Multimedia Subsystem

1.2.1 Development of the IMS

As the popularity of Internet services grew so did the need for mobile access to them. Cellular networks were ideal candidates for this as their coverage areas typically encompassed virtually all urban areas as well as extending into key rural areas. The General Packet Radio Service (GPRS) was developed to provide 2G cellular systems with data capabilities and gave cellular handsets the ability to connect to the Internet. GPRS data rates were however significantly lower than those offered by fixed line access and other wireless access methods. The Universal Mobile Telecommunications System (UMTS), a third-generation (3G) cellular network system, sought to increase the bandwidth available in cellular networks by using improved spectral efficiency techniques and allowing users access to a wider range of more advanced services.

The 3GPP, a collaboration of a number of telecommunications standards bodies, was formed with the goal of producing technical specifications for a 3G Mobile System based on evolved GSM core networks and the radio access technologies these networks support [7]. Initially the 3GPP focused on increasing the available bandwidth in 3G networks by improving the radio access methods. However, they found these networks still inherited many of the limitations of the circuit and voice-oriented GSM network on which they were based. In order to provide newer services that could take advantage of the higher bandwidths, the 3GPP realised that the network architecture needed to evolve [8]. This evolution came in the form of the IP Multimedia Subsystem which defines an architectural framework for the delivery of IP services in an all-IP packet based core.

1.2.2 Motivations for the adoption of the IMS technology by operators

The IMS has attracted much attention from the telecommunications community and there are a number of reasons for this. The key one being that the IMS delivers innovative multimedia services over both fixed and mobile networks using open standards. Furthermore, the IMS addresses modern issues such as convergence, service creation and delivery as well as service interconnection. With the IMS the operator is able to either retain its existing business models or evolve towards new ones.

Quality of Service

The main driver for the development of a new network architecture, as opposed to simply providing improved connection speeds to the Internet, is due to the inability of the Internet to provide acceptable Quality of Service (QoS) to users. The Internet only provides a best effort service in that it cannot guarantee delivery of all information transmitted between two end-points. Network congestion and network failures could mean the loss of information packets as they pass through the network, with end-points needing to keep track of packets in order to re-transmit any packets that did not make it through the network. Although this may be acceptable for traditional Internet, non-real-time services such as web-browsing and file-transfers, it is unacceptable for more modern, real-time multimedia services, such as voice calling and video streaming. Delays in the transmission of packets could cause these modern services to become unusable by customers. Even without congestion or failures in the network, the best effort nature of the Internet means it may not be able to provide acceptable QoS levels to all services at all times.

To highlight this problem, consider that modern digital video transmission utilises video encoding techniques to reduce the bit-rate of the video and enable it to be transmitted over present day communication links. The encoding of the video utilises the fact that many frames of the video are spatially redundant[1] and the encoder generates a reference frame to be utilised by the decoder for the prediction and decoding of subsequent predicted frames. Loss of these reference frames during transmission is likely to lead to interruptions in the playback of the video and could cause the user to abandon using the service.

The IMS however ensures end-to-end QoS for IP Multimedia packet flows by identifying session flows at the bearer level and prioritising the routing of packets, as to ensure reliable delivery of real-time IP-based multimedia services.

Separation of session signalling and media flows

The IMS separates the session signalling from the media flows. The session signalling is routed through the IMS core for purposes such as authentication, authorisation and accounting, while the media flows take advantage of IP routing protocols to ensure optimised delivery of the media data across the network.

Session-based charging of IP services

At present operators typically charge users for the amount of information transferred while connected to their network. The operator cannot charge the user for the use of individual services as the contents of the IP datagrams are not known. The IMS on the other hand provides information about the service being used by the subscriber and allows the operator to charge for the service using a method that is appropriate for that particular service; flat rate, time-based, QoS-based, or any other type of new charging method introduced. For example a VoD service may be charged on a flat-rate, or per session basis, and not on the amount of time he/she spends watching the video.

IMS as support for a service delivery platform

The IMS defines standard interfaces for use by service developers for integration of new services into the IMS network. Standardisation of common services such as voice, presence, instant messaging (IM) and others, allow for their use in the

[1] Apart from major scene changes and scenes involving a good deal of movement there is little change in the information in successive frames of a video sequence.

creation of newer, more sophisticated services. The IMS facilitates the convergence of telecommunications and Internet services through the use of common Internet protocols and Web technologies. This allows developers, who are already familiar with these technologies, to quickly develop and deploy new IMS services. In addition to this the standard interfaces allow operators to obtain services from different vendors, thus avoiding being locked to a single vendor for the acquisition of new services.

The all-IP core enables the inter-working of different access networks as well as facilitating fixed mobile convergence by allowing seamless distribution of services over both fixed and mobile broadband access networks.

By separating out the application services from the core network infrastructure the IMS allows independent scaling of the network and applications services; the number of Call Session Control Functions (CSCFs) can be increased, enabling a higher number of network subscribers to be served, while the number of application servers which handle a particular service can be changed according to the demand for the service.

The range of services is expected to be a major draw card to IMS networks. The term "triple play" is often used when talking about the IMS and refers to the integration of voice, video and data services over a single network and from a single network operator. The IMS allows network operators to provide bundled service offerings to attract customers to their networks, while customers benefit from the reduced cost and simplified billing associated with these offerings.

1.2.3 IMS standardization

The 3GPP first released IMS specifications in 2002 as a small part of their Release 5 specification. These documents did not however discuss voice or any other services in detail, leading to some confusion regarding the interoperability and realization of these services. This prompted large corporations, such as Ericsson, to take it upon themselves to define IP telephony for the IMS and to have it included in Release 7 of the 3GPP's standards. Presently Release 8 is scheduled to be finalized and published in December 2008/January 2009 [9].

Supporting 3GPP's vision of an all-IP core, a number of other Standard Definition Organizations (SDOs) have contributed to the IMS standards. These include:

- The 3rd Generation Project Partnership 2 (3GPP2) - a collaboration between North American and Asian standardising bodies, who have adapted the 3GPP standards for CDMA2000 networks.

- Telecoms & Internet converged Services & Protocols for Advanced Networks (TISPAN) - a standardisation body of the European Telecommunications Standards Institute (ETSI), who have contributed to the IMS standards for operators in the fixed-line access domain.

- CableLabs's, a non-profit research and development consortium consisting of members from the cable industry, have adapted 3GPP, Release 6, IMS standards for cable networks in their PacketCable project.

Other SDOs concerned with the IMS are the International Telecommunications Union (ITU), whose NGN focus group has included the IMS as part of its Next Generation Network (NGN) set of standards, as well as the WiMax Forum and the Broadband Forum, who have specified requirements for the interaction and integration of IMS with their own architectures.

1.3 Emergence of Web 2.0

The first IMS network deployments are only expected to be seen in 2011. Presently however there are a number of Web application frameworks and technologies, such as Ajax: Adobe Flash: Adobe Flex; Microsoft Silverlight and Curl, which enable developers to create sophisticated web services which can be accessed via web browsers with an appropriate plug-in. As a result of these the Internet has seen an explosion of innovative and sophisticated services emerging, such as social networking and Web Mashups[2]. The transition from the World Wide Web being a number of static pages to its use for the deployment of highly interactive and sophisticated services has often been referred to as 'Web 2.0'.

Given the open nature of the Internet, many Web 2.0 services are offered free of charge to users and the predominant source of revenue is gained from personalised advertising. An example of this would be a user requesting a content related to

[2]A web application combining data from several sources in order to create a new and unique service

motorcars may be interested in a special offer for a new vehicle. This presents a problem for operators looking to deploy the IMS as they are required to provide new services that improve upon these Web 2.0 services to the extent that users will want to pay for them.

1.4 IP Multimedia and Internet Protocol Television

The transmission of high quality multimedia data over IP-based communication links, has been made possible by significant increases in network bandwidth over the last few years along with the improvement of video coding technologies. This has lead to the development of new video services such as multimedia messaging, video conferencing and Internet Protocol Television.

The ITU-T Focus Group on IPTV provides the following definition for IPTV: "multimedia services, such as television, video, audio, text, graphics, and data, delivered over IP-based networks managed to provide the required level of quality of service (QoS), quality of end-user experience (QoE), security, interactivity and reliability"[10]. IPTV systems go beyond traditional broadcast television systems, as the use of the IP protocol allows for two way communication between service provider and IPTV user. Traditional television services were seen as push services, since they only allowed the television content to be pushed to the user. IPTV services however can take advantage of the two way communications channel between the user and the service to gather information from the user. This allows highly interactive and sophisticated services such as Interactive game shows, television shopping and content rating systems to be developed. Service providers are also able to note the viewing habits of IPTV subscribers allowing them to customize advertising content for individual subscribers as well as give subscribers access to a community of subscribers with similar interests, or suggest other content that may be of interest to the subscriber using this information. The main driver behind consumer interest in IPTV services is the potential for Content on Demand (CoD), or Video on Demand (VoD), offerings. With this CoD services users are able to stream a wealth of Hollywood and television network content to their end terminals. The user is also able to control the delivery of the content by using VCR-like functionality such as PLAY, PAUSE, FAST-FORWARD and so on.

IPTV is considered to be one of the major services in next-generation networks [10]. As voice service revenues continue to dwindle, many operators have dedicated IPTV projects investigating the deployment and roll out of IPTV services in order to attract new customers to their networks and retaining existing ones. A recent forecast by the Multimedia Research Group Inc. in April 2008, predicted that the number of subscribers to IPTV services worldwide will grow from 20.4 million in 2008 to 89.1 million in 2012. The group also predicts that global revenues from IPTV services will rise from 5.9 billion USD to 29.6 USD in 2012. Figure 1.2 and Figure 1.3 show the breakdown of projected IPTV subscribers and IPTV service revenues for the different world regions [11].

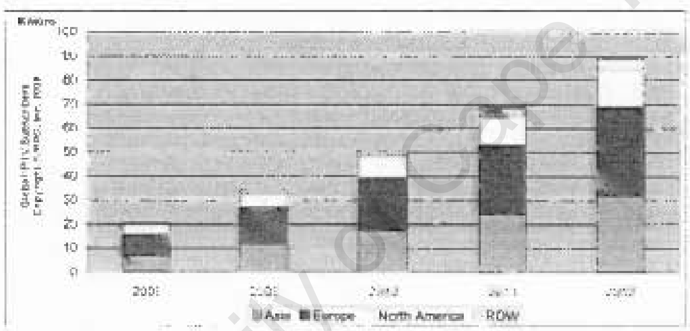

Figure 1.2: Forecast of global IPTV subscribers for the period 2008 - 2012

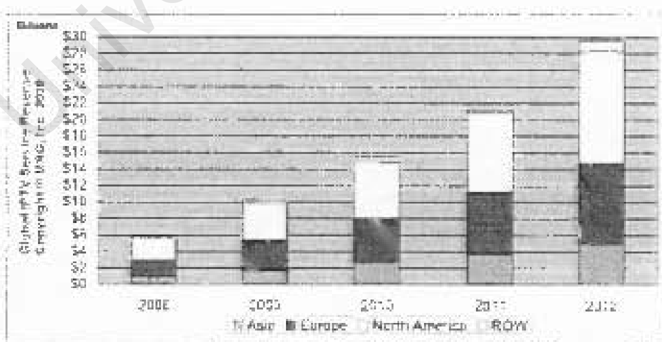

Figure 1.3: Forecast of global IPTV service revenues for the period 2008 - 2012

1.4.1 IPTV standardization

Given the large amount of interest in IPTV, a number of SDOs have begun standardisation of end-to-end IPTV service architectures. Some of the major IPTV standards activities are listed below.

ITU-T IPTV

The ITU-T created the IPTV Focus Group (FG) with a mission "to coordinate and promote the development of global IPTV standards taking into account the existing work of the ITU study groups as well as Standards Developing Organisations, Fora and Consortia" [12]. The FG ended its activities in January of 2008 and transferred its documents to the relevant study groups within the ITU-T for the development of draft Recommendations. The ITU has since formed the IPTV - Global Standards Initiative (IPTV-GSI) with the goal of facilitating the interaction between the relevant study groups and accelerating the standardisation process. The ITU-FG settled on three distinct IPTV functional architectures: A Non-NGN architecture and two NGN-based architectures, one of which utilises IMS as the control architecture.

The ITU-T standards include the IPTV functional architectures, requirements for support of IPTV services, overview of IPTV terminal devices, high-level specification of meta-data for IPTV, quality-of-experience requirements for IPTV services, as well as home networks supporting IPTV services. The final set of ITU-T Recommendations on IPTV is expected to be finalised by mid-2009 [10].

ATIS IPTV Interoperability Forum

The Alliance for Telecommunications Industry Solutions (ATIS) initiated the IPTV Interoperability Forum (IIF) with the aim of developing standards that enable the interoperability, interconnection, and implementation of IPTV systems and services. Important aspects are investigated by ATIS, within the IIF, including the defining of IPTV logical domains, IPTV references architectures (IMS- and non-IMS-based, as well as their coexistence), content delivery concepts with QoE, digital rights management (DRM) requirements, and the testing requirements of the components, reliability, and robustness of service components [13].

DVB over IP Networks

The Digital Video Broadcasting (DVB) Project is a consortium which includes broadcasters, manufacturers, network operators, software developers, regulatory bodies and other organisations from around the world. The DVB project is responsible for creating standards for digital television in Europe, as well as many other regions of the world, and publishes its specifications though ETSI. Work related to IPTV is standardised under DVB-IPTV which is defined by the DVB as a "set of open, interoperable technical specifications that facilitate the delivery of digital TV using the Internet Protocol over bi-directional fixed broadband networks". Specifications published to date define the transport of MPEG-2 TS-based services over IP based networks [14], the provision of Broadband Content Guide (BCG) information over IP [15], as well as remote management and firmware update systems for DVB IP services [16]. ETSI TISPAN incorporates these DVB standards into their own IPTV architectures.

ETSI TISPAN IPTV

TISPAN NGN R2 contains several specifications that address IPTV service requirements and architectures. Both a non-IMS IPTV subsystem and an IMS-based IPTV subsystem are specified. The standards for the IMS-based IPTV subsystem [4] provide a comprehensive specification of the architecture and functions, including service discovery functions, service control functions and end-user terminal functions as well as the corresponding protocols and interfaces used for signalling and media flows between these components.

3GPP MBMS

The 3GPP does not offer any specific IPTV specifications but their Multicast/broadcast (MBMS) specifications [17] define an efficient way of delivery and controlling multicast and broadcast services over 3G networks. These multicast and broadcast services allow the optimisation of broadcast and multicast IPTV services delivery. Although limited at present to cellular networks, there is ongoing research to extend MBMS services to apply to the IMS architecture as a whole [18].

OMA BCAST

The Open Mobile Alliance (OMA) contains a Broadcasting Working Group that deals with functional issues that are generic enough to be common to many broadcast services and that can be defined and implemented in a bearer-independent way. This includes service guides, file distribution, media stream distribution, service protection, DRM, service interaction, service provisioning, terminal provisioning, as well as notifications, and so on [13]. All are key aspects related to IPTV services supporting broadcast/multicast and which enable the distribution of rich, interactive media content to a large number of mobile viewers.

1.5 Problem Definition

The IMS enables users to access services from a range of user end-terminals and access network technologies. Fixed line access technologies allow for the data rates in the range of a few hundred kbit/s per second to several Mbit/s, depending on the access technology used. Similarly, wireless access technologies have varying data rates but are further complicated due to the mobility of users which impacts on the received signal strength of the end terminal. In addition to this the number of end-terminals that can connect to these networks is limitless. These range from portable multimedia players and cell phones to high-end computers and set-top boxes. Each of these devices has its own set of capabilities with screen size, processing power and supported coding formats varying from device to device.

This need for IMS services to be able to adapt to meet differing user requirements presents a significant challenge for IPTV services. This is because video encoded with a particular codec, bit-rate, spatial resolution and temporal resolution may not be suitable for delivery over all access networks and to all user end-terminal devices. As an example of this we have 4 different IPTV subscribers who accessing the service from different networks and different end terminal devices; Figure 1.4 depicts the scenario. In this example the video server is only able to provide content at a fixed aspect ratio and at fixed bit rate (typical of today's video services where video is encoded using specific parameters that cannot be changed unless the video is re-encoded). For user A this is a not a problem as his end-terminal has a screen size in-line with the video's spatial resolution and the

Wi-Fi connection he is connected to has sufficient bandwidth to support the video stream. The portable media player of user B however cannot handle the bit-rate of the video and the video skips and pauses frequently. User C is connected to a Digital Subscriber Line (DSL) service with sufficient bandwidth but the video is not of a suitable quality for his HD capable device. Finally user D is connected via his operators 3G network but due to his mobility the signal strength frequently drops causing the video to buffer and playback to stop.

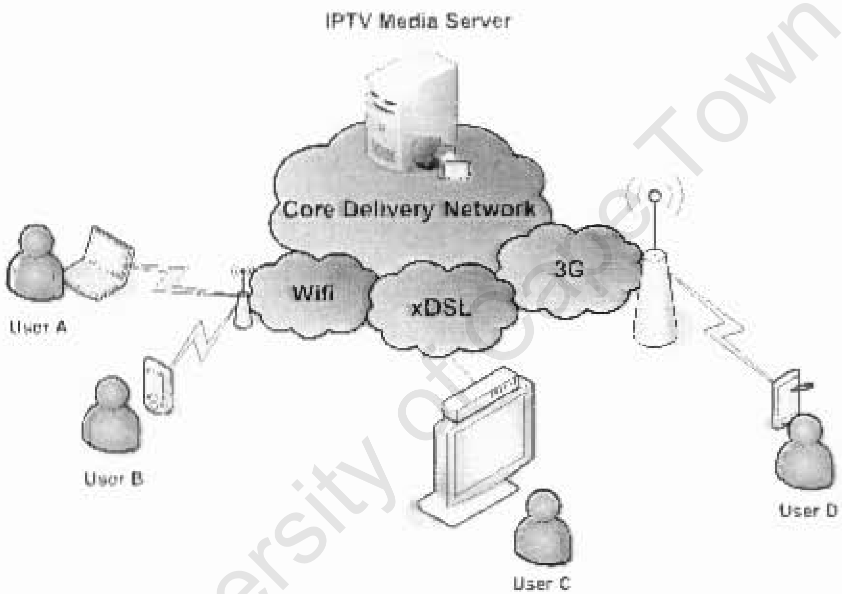

Figure 1.4: IPTV problem scenario

Although this example is somewhat exaggerated, it highlights the need for IPTV services in the IMS to adapt video streams to best suit the requirements of each individual client and facilitate the convergence of fixed and mobile broadband access networks for IPTV services.

1.6 Thesis Objectives

The development and standardization of IMS and its associated services is an ongoing process. There are many issues that need to be resolved before a full scale IMS-based IPTV deployment can be realised. Without the ability to adapt

IPTV video services to meet the needs of individual users, consumer interest in these services will be severally limited. This thesis primarily analyses multimedia content adaption methods to support the adaptation of multimedia content in IMS-IPTV deployments. It aims to showcase the advantages and disadvantages of each adaptation method and determine their suitability for use in a present day IMS-IPTV deployment. The most suitable content adaptation methods will be incorporated into a content adaptation scheme for use in IMS-IPTV deployments. This functional architecture will then be implemented in a test-bed architecture, to enable testing and evaluation of the proposed content adaptation scheme. In particular the storage and processing requirements of the stream will be analysed in order to access the scalability of the scheme. The software developed for this test-bed will be released as open-source, allowing for its extension and to serve as a basis for future IMS-IPTV research.

1.7 Thesis Scope and Limitations

There are a number of IPTV architectures currently being deployed by service providers [19, 20, 21] or standardised by standards organisations [13]. Although the concepts presented in this thesis may be used in order to achieve video adaptation in other IPTV solutions, this thesis focuses on applying them to an IMS-based IPTV architecture.

In order to create a successful IPTV infrastructure a number of issues need to be addressed. Quality of Service (QoS) mechanisms need to be in place in order to ensure a reliable and acceptable IPTV experience for users; billing and charging mechanisms need to be present, and flexible enough to support the various charging models operators may wish to use (pre-paid, on line charging, time-based charging, etc); Security and Digital Rights management issues need also to be addressed in order to protect users and ensure that content copyright laws are not infringed. These are complex peripheral issues and there is much standardisation work being done in 3GPP and other bodies to ensure that appropriate mechanisms are in place to handle these issues; as such they are beyond the scope of this thesis.

The IMS uses many protocols standardised by the IETF: SIP, SDP, RTP and RTSP are the four application layer protocols used extensively in this thesis. A full discussion regarding the inner-workings of these protocols is beyond the scope of this thesis and readers are advised to consult the relevant Request for Comments (RFC) documents.

Simulations are an effective tool for the prediction of network performance. They are however limited, in that they often do not take into account real-world factors and do not truly reflect the behaviour of physical networks. As such the conclusions drawn from simulations often cannot be generated in the real world and for this reason the decision was undertaken to implement the adaptation architecture in the form of a physical test-bed. This does however limit the research as it is not as easy to adapt a physical test-bed to new technologies or to new experiments.

1.8 Thesis outline

The remainder of this thesis is outlined below:

Chapter 2 describes and analyses the main content adaptation methods found in the literature with relevant literature relating to each method highlighted and reviewed. The chapter then gives an overview and an analysis of the ETSI TISPAN IMS-Based IPTV architecture, chosen as the reference architecture for this thesis.

Chapter 3 begins with a discussion on the relevant design considerations for a content adaptation scheme for IMS-IPTV architectures, after which each of the content adaptation methods presented in chapter 2 is reviewed as to their suitability for use in IMS-IPTV deployments. A hybrid content adaptation scheme, which makes use of two of content adaptation methods, is then proposed. The advantages associated with the use of the hybrid scheme are highlighted and the integration of the hybrid adaptation scheme with the ETSI TISPAN IMS-Based IPTV architecture is discussed.

In Chapter 4 the architectural design of an evaluation framework for the adaptation of video content for IMS-IPTV services is given. Initially, the objectives, requirements and limitations of the test-bed are presented and after which the software used and/or developed for the test-bed is discussed. Finally the test-bed's topology is given along with a description of the hardware used in the implementation of the test-bed.

Chapter 5 begins with a discussion on the tools and video sequences used during the tests performed using test-bed constructed for this thesis. This is followed by presentation and analysis of the results obtained using the test-bed.

Chapter 6 presents a set of conclusions that were drawn from the evaluations in Chapter 5, after which recommendations and future work is presented.

Chapter 2

Background and Literature Review

The previous chapter introduced the IMS and discussed the need for IMS-IPTV services to be able to adapt multimedia content, to cater for heterogeneous access networks and differing client terminal capabilities. This chapter provides a brief overview of different content adaptation methods found in literature as well as discussion on each. As video content adaptation has been a popular research topic in the past, with a number of publications related to the subject, the aim of this chapter is not to provide an exhaustive list of such literature but rather to highlight the most up-to-date and relevant publications in the context of video adaptation methods. This chapter also includes a discussion on ETSI TISPAN IPTV-IMS architecture used as the reference architecture for the deployment of content adaptation methods in this thesis. The reader may consult Appendix 1 to familiarise themselves with digital video compression and factors that influence the video adaptation process.

2.1 Video content adaptation methods

2.1.1 Precoding

Precoding is a method for video adaption that has the content server support multiple versions of the multimedia content, with different formats and bit-rates.

When the media server receives a request from a client, it determines the client's requirements and capabilities, and uses this information to select a suitable version of the content for the client.

B. Shen *et al.* mention precoding in [3]. The authors analyse the strengths and weaknesses of using precoding for video content adaption and contrast its use against scalable video encoding (Section 2.1.2) and trans-coding (Section 2.1.4). The authors state that the major advantage associated with such a scheme is little to no quality degradation between the encoded video and the original video, since the encoded sequence is based on the original sequence of video frames. Another advantage is that end-to-end security need not be affected when switching between different versions of the video, since each video sequence can be encrypted independently. The disadvantages of precoding, according to the authors, is the difficulty at the source of anticipating what codec, bit-rate and so on, the content should be precoded with as well as the inability to support real-time video applications such as telephony or group communication.

D. Miras *et al.* [22] also mention precoding; referred to as *multiple versions*. The authors describe how the method can support dynamic switching among the multiple streams, by using synchronisation points in the bit-streams; however, this has a disadvantage of complicating the encoding process. Other disadvantages of the method, according to the authors, are that it requires extra space to store the multiple encodings and that the granularity of the rate adaption is limited by the number of available streams.

J. Chakareski *et al.* [23] include precoding as a method for the rate adaptation of streaming media and explain how it can be implemented. The authors however fail to mention any advantages of disadvantages associated with its use.

2.1.2 Scalable video encoding

Scalable video encoding allows adaption of the output video stream by allowing parts of the video bit-stream to be removed. Video is encoded into a number of layers with successive layers representing an improvement in the quality of the decoded video. The base layer represents the lowest quality version of the video bit-stream and each successive layer refines the video and improves the videos perceived quality. Decoding of each successive layer requires the presence of the lower layers.

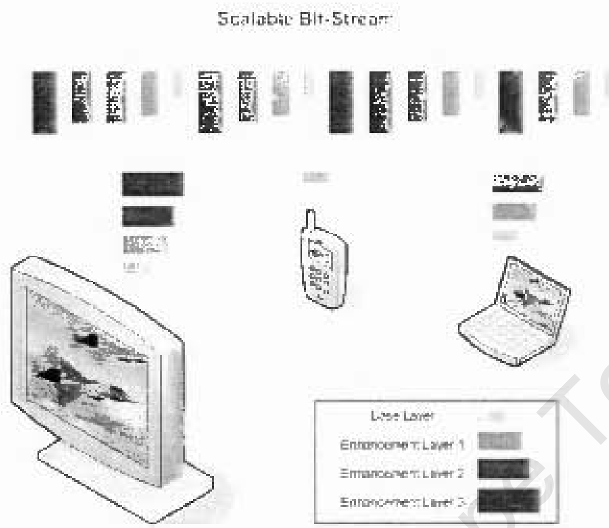

Figure 2.1: Scalable Video Encoding allows devices to subscribe to bit-streams in-line with their capabilities

Previous coding standards such as the H.262/MPEG-2, H.263, and MPEG-4 Visual standards all supported tools that at least provided some form of scalability; however, the scalable profiles of these standards were rarely used and not widely adopted by industry. The reasons for this include: the characteristics of traditional video broadcasting systems, the significant loss in encoding efficiency (when using spatial and quality scalability features), as well as the large increase in the complexity of the scalable decoder (when compared with the corresponding non-scalable profiles) [24].

The most recent and advanced scalable codec is the Scalable Video Codec (SVC), which appears as an amendment to the H.264/MPEG-4 AVC standard[1], jointly published by both the ITU-T and ISO/IEC standardization bodies. The H.264 standard has proved popular with industry and other standards bodies and today many products and systems support the standard. Because of the high adoption rate of industry and the investments made in developing H.264 compliant products, the ITU-T and ISO/IEC video groups found it appropriate to develop an SVC extension and reuse the key features of the H.264/AVC standard. This

[1] Note: H.264/MPEG-4 AVC, H.264 and AVC are used interchangeably throughout this thesis.

SVC[2] extension allows for the temporal resolution, spatial resolution and/or the fidelity of the video to be reduced in order to adapt the video content. H. Schwarz *et al.* give a good overview of the H.264 scalable extension in [24]. They discuss how the SVC extension incorporates techniques that can be used to reduce the loss in coder efficiency and bring it closer to the original H.264 standard. They also show how the decoder complexity can be decreased while still maintaining reasonable rate-distortion performance. The performance of the SVC extension is also analysed in [25], where the authors compare it the original H.264 standard as well as the MPEG-4 ASP standard. They conclude that, although the use of SVC leads to a compromise in either quality or bit-rate of the video, the gap between SVC and H.264 encoder efficiency can be relatively small. Also the performance impact, of the increased scalability functionality, can be optimized by tailoring the layer configuration of the scalable stream to the needs of the application.

D. Miras *et al.* [22] give a brief summary of the advantages and disadvantages of using scalable encoding. Advantages according to the authors, are that scalable encoding allows the video bit-rate to be adapted to the network conditions, by dropping or adding layers, as well as improving the reliability of video playback by prioritizing lower layers. Additionally, unequal error protection can be employed to protect the lower layers of the video bit-stream. Disadvantages include deciding what partition method to use, the efficiency cost, deciding how many layers to use and/or how to distribute the bandwidth among the layers.

J. Chakareski *et al.* *[23]* include the use of scalable encoding as a means of achieving bandwidth adaptation by rate scalability of the media stream. The authors mention the advantages of its use being for on-demand media applications or in one-to-many streaming architectures where the same media, or a finite set of streams, serves a number of different clients. The authors also mention the disadvantage of the generally smaller compression efficiency associated with the use of scalable encoders.

J. Kim *et al.* [26] propose the use of H264/MPEG-4 SVC in Mobile IPTV services and justify its choice through its use in applications such as multi-resolution content analysis, content adaptation, complexity adaptation, and bandwidth adap-

[2] SVC always refers to the scalable amendment to the H.264/MPEG-4 AVC standard in this thesis.

tation. The authors mention a disadvantage of the use of SVC as the relatively high complexity of the SVC encoder which makes real-time encoding very difficult to achieve.

One major disadvantage of the SVC scheme, not mentioned in [24, 25, 22, 26], is the requirements of each individual user terminal to be capable of decoding the SVC bit-stream. If IPTV services were to be deployed using H.264/MPEG-4 SVC then all devices would need the compliant decoder or playback of the video would not be possible[3].

2.1.3 Multiple Description Coding (MDC)

Multiple description coding (MDC) encodes video into two or more independent streams. In contrast to SVC, where the lower layers need to be present in order to decode higher layers, MDC streams can be independently decoded. These streams, called descriptions, are sent to the decoder over separate communications paths and in the event of one, or more, descriptions failing to arrive, the decoder uses the available descriptions to approximate the original signal. The distortion in the reconstructed video decreases with the reception and decoding of any additional descriptions. This distortion is lower bounded by the distortion attained by single description coding (SDC), operating at the same overall bit-rate, in an error-free transmission scenario. MDC allows video to be adapted, by providing the client with only the relevant descriptions required to achieve the clients desired, spatial resolution, temporal resolution and/or fidelity.

M-T Lu *et al.* [27] use MDC in their Peer to Peer (P2P) IPTV architecture. MDC allows users of the system to subscribe to streams with different resolutions, as well as to ensure that users receive the highest quality achievable with the available network resources. The authors propose a new MDC scheme that uses four bit-streams: even and odd streams consisting of original video sequence, as well as even and odd streams consisting of the video sequence scaled by a quarter. The authors show how this scheme outperforms previous spatial and temporal interpolation schemes for a single video sequence. The authors also compare MDC with

[3]Note: Devices supporting the non scalable profiles of H.264/MPEG-4 AVC are able to decode just the base layer of the SVC bit-stream but this base layer will typically be of a very low quality and hence not suitable for devices requiring higher quality versions of the video.

SVC and SDC, stating the major drawbacks of layer coding to be its high complexity and lack of error resilience, while SDC achieves higher coding efficiency than MDC since MDC cannot fully exploit the temporal relationship between frames.

Similar to SVC, MDC would require that all devices have the appropriate MDC decoder, which at present is highly unlikely since there is no standardized MDC format.

2.1.4 Trans-coding

Trans-coding is the process of decoding a video signal and re-encoding the frames to another coding format or to the same coding format but with altered parameters of the video. Early uses of this method mapped the bit rate of the video to the available capacity of the communications channel. With the advent of mobile computers and smart phones, the technology has evolved to support spatial resolution, frame-rate and/or bit-rate adaption of a video sequence.

Figure 2.2 shows the trans-coding process and represents the most basic form of trans-coding, known as cascaded pixel-domain trans-coding. The signal is first decoded and the original video frames reconstructed after which any required intermediate processing of the video frames is performed. Finally the frames are re-encoded, subject to any new constraints specified to the required output format. This trans-coding approach however is very costly in terms of computing power and memory usage and as such, more efficient techniques are typically utilized.

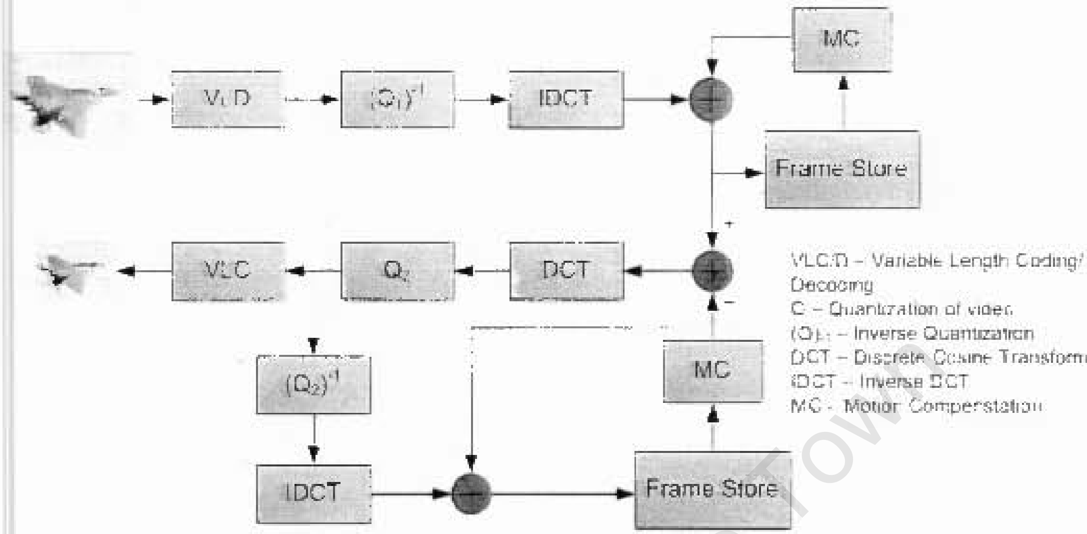

Figure 2.2: Illustration of cascade pixel-domain trans-coding.

A. Vetro *et al.* [28] provide an extensive overview of the various trans-coding architectures and techniques that have been researched and developed prior to 2003. The paper looks at trans-coding architectures that enable bit-rate, spatial resolution and temporal resolution of video to be changed. In addition to this error-resilience trans-coding is also examined. The authors provide a good overview of the trans-coding process and give an extensive list of references to other research that has been conducted to optimise the trans-coding process.

J. Chakareski *et al.* [23] mention that trans-coding in their discussion on methods for achieving rate scalability, stating that its application in practice is quiet limited due to its high computational complexity.

B. Shen *et al.* [3] argue that video trans-coding is better suited for content adaptation than precoding and scalable video coding methods in the IMS. The reason for this is due to the flexibility of trans-coding in adapting the video to any codec, spatial resolution, temporal resolution and/or bit-rate the client requires. The authors present an intelligent method of exploiting the correlation between the input and output of a video trans-coder, called xcode, which enables the trans-coding to be performed without having to fully re-encode the video and thus optimizing the trans-coding process.

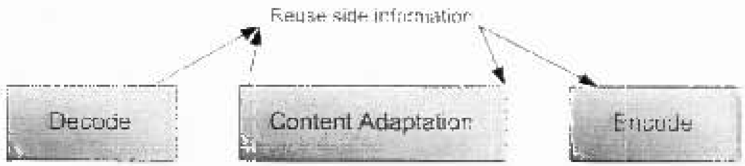

Figure 2.3: Illustration of the xcode process proposed in [3].

The xcode method reuses the motion estimation and macro-blocks of the original encoded video sequence for encoding of the adapted video sequence, thus avoiding having to recalculate the transform-domain information and reducing processing requirements. This would lead to a decrease in the processing power required to achieve the trans-coding process; however, the authors do not provide any results comparing their xcode method with the full re-encoding of video. Furthermore, they admit that the xcode method would not work when trans-coding to a coding format that does not use the same transform base as the original coding format. Other trans-coding issues addressed by the authors include dynamic trans-coding for interactive video applications and the use trans-coding for the improving video quality in lossy access networks. The authors conclude the paper with a case study analyzing the power consumption of a mobile phone for video playback of two video sequences, at a range of frame-sizes and frame-rates. Findings from the case study show that playback of video with lower frame-sizes and frame-rates takes longer to drain the phone's battery then video with higher frame-sizes and frame-rates.

2.2 ETSI TISPAN IMS-based IPTV Architecture

ETSI TISPAN standards to-date define the most comprehensive IMS-based IPTV architecture. In addition to this the standards are developed in co-operation with ITU-T FG, ATIS IIF and DVB, as to ensure interoperability between IPTV systems developed by different vendors and deployed by different network operators. For this reason, the ETSI TISPAN IMS-based IPTV standards are chosen as the reference IMS-IPTV architecture for this thesis. This section provides a detailed overview of the TISPAN IMS-based architecture and is intended to familiarise readers with the various functional components and reference points to ensure an understanding of the later chapters of this thesis.

2.2.1 Overview of the ETSI TISPAN IMS-based IPTV Architecture

Figure 2.4 shows a high-level overview of the ETSI TISPAN IMS-based IPTV functional architecture.

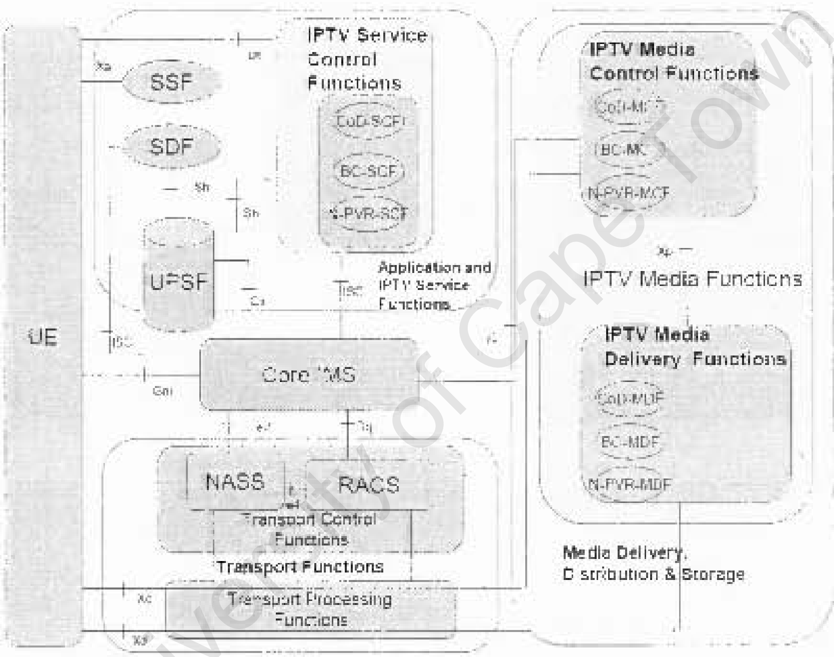

Figure 2.4: ETSI TISPAN functional architecture for IPTV services [4].

User Equipment (UE)

The IPTV enabled User Equipment (UE) is responsible for termination of IPTV control and media signals as well as displaying of the corresponding information to the user. Interaction between the user and the UE allows the selection of programme, content, and service descriptions, such as content guides for broadcast and VoD services.

user readable data related to the IPTV service such as channel names, start times and so forth. This information is usually presented to the user in the form of an Electronic Program Guide (EPG), which is a graphical on-screen display of the information.

Service Control Function (SCF)

The Service Control Function (SCF) is a SIP Application Server (AS) responsible for authorisation during session setup and modification: this includes checking the user's profiles in order to allow or deny access to the service; credit limit and credit control using on-line charging systems; as well as selection of the relevant IPTV media functions.

IPTV Media Control Function and Delivery Functions (MCF and MDF)

The IPTV Media Functions are responsible for the control and delivery of IPTV media flows to the UE. They are divided into the Media Control Functions (MCF) and Media Delivery Functions (MDF).

MCF Tasks:

- Handling media flow control of MDF.
- Managing the media processing of MDF, if required.
- Monitoring the status of MDF.
- Managing interaction with the UE (e.g. trick mode commands).
- Handling interaction with the IPTV service control function SCF.
- Keeping an accurate view on status and content distribution related to the different MDFs that it controls.
- Selecting an MDF, in the case of an MCF controlling multiple MDFs, different criteria may be applicable (e.g. location of UE, load on the Media Functions, etc).

- Selection of an alternative media functions (MF), in the event the MF cannot process the request (eg. Relevant content not available, load on the MF, etc) and the return of the MF information to the SCF.
- Generate charging information, e.g. for end-user charging based on the viewed content.

MDF Tasks:

- Handling the delivery of the media flows to the UE.
- Status reporting to MCF (e.g. reporting on established IPTV media streams).
- Storage of media (e.g. CoD assets) and possibly also storage of some service information related to the media.
- In particular, it may be used for storage of the most frequently accessed content or user specific content (e.g. recording PVR, Time-shift TV, BC service with Trick mode, user generated content) if the same tasks are not performed by UE.
- May additionally process, encode or trans-code (if required) media to different required media formats (e.g. TV resolution depending on terminals capabilities or user preferences).
- May perform content protection functionalities (e.g. content encryption).
- May support content ingestion of IPTV media.
- For BC services MDF may act as source for multicast streams of BC media streams.
- May collect QoE reports (e.g. from UE using Xd).

Transport Functions

The Transport Control Functions (TCF) enable policy control, resource reservation and admission control, as well as IP address provisioning, network level user authentication and access network configuration, as defined in TISPAN.

The Transport Processing Functions (TPF) represents the IP core and access network transmission links.

ETSI TISPAN IMS-IPTV Reference points

Table 2.1 shows the names given to each of the interfaces shown in Figure 2.4, as well as the signalling protocols that are used on each interface.

FE/ Ref.point (protocol)	UE	IMS core	UPSF	SDF	SSF	SCF	MCF	MDF	ECF/ EFF
UE	---	Gm (SIP/SDP)	---	via Core IMS (SIP/SDP)	Xa (HTTP, DVBSTP, FLUTE)	Ut (HTTP), via Core IMS (SIP/SDP)	Xc (RTSP) (Note 1)	Xd (UDP/RT) (Note 1)	Dj, Di IGMP/ MLD
IMS core	Gm (SIP/SDP)	---	Cx (Diameter)	---	---	ISC (SIP/SDP)	y2 (SIP/SDP)	---	---
UPSF	---	Cx (Diameter)	---	Sh (Diameter)	---	Sh (Diameter)	---	---	---
SDF	via Core IMS (SIP/SDP)	---	Sh (Diameter)	---	---	---	---	---	---
SSF	Xa (HTTP, DVBSTP, FLUTE)	---	---	---	---	---	---	---	---
SCF	Ut (HTTP), via Core IMS (SIP/SDP)	ISC (SIP/SDP)	Sh (Diameter)	---	---	---	via Core IMS & y2 (SIP/SDP)	---	---
MCF	Xc (RTSP) (Note 1)	y2 (SIP/SDP)	---	---	---	via Core IMS & y2 (SIP/SDP)	---	Xp (not defined)	---
MDF	Xd (UDP/RTP) (Note 1)	---	---	---	---	---	Xp (not defined)	---	---
ECF/ EFF	---	---	---	---	---	---	---	---	---
NOTE 1	As described in TS 182 027 [2] clauses 6.4 and 6.5, Xc and Xd are logical reference points that can be decomposed into Dj and possibly Di, Ds or Iz reference points depending on the location of the MCF or MDF.								

Table 2.1: IMS-based IPTV functional entities and protocols used on reference points [5].

The following reference points, discussed below, are considered relevant to this thesis:

UE - SSF (Xa)

The Xa reference point is used by the UE for the selection of IPTV services from the SSF.

UE - IPTV Service Control Functions (Ut)

The Ut reference point is used for configuration of the subscriber's IPTV profile.

Core IMS

The IMS Core provides functionality for authentication, authorisation, and signalling for the setup of the service provisioning and content delivery. It routes signalling messages to the appropriate application server or triggers the applications based on settings maintained in the database of user profiles. For resource reservation and admission control this function interacts with the Resource and Admission Control Subsystem (RACS).

User Profile Selection Functions (UPSF) and Subscription Locator Functions (SLF)

The User Profile Selection Function (UPSF) (equivalent to the Home Subscriber Server (HSS) in 3GPP specifications) is a database that stores the user profiles as well as any IPTV specific profile data. The IPTV SCF communicates with the UPSF over the Diameter based Cx interface in order to obtain this information during session setup and modification. If multiple UPSFs exist (for load balancing) the IPTV SCF may make use of a Subscription Locator Function (SLF) in order to obtain the address of a relevant UPSF.

Service Discovery and Selection Functions (SDF and SSF)

The role of the Service Discovery Function (SDF) is to generate and/or provide service attachment information for users, as well as to provide a means for discovering personalised IPTV services in the IMS network. The service attachment information consists of Service Selection Function (SSF) addresses in the form of URIs and/or IP-addresses which the user agent can use to contact an appropriate SSF.

The SSF provides service selection information such as a list of available servers that the UE can browse and select from. The SSF either generates this information itself or relays the information from another source. The information provided for each service includes an identifier associated with the IPTV service, for CoD services this is a single content item while for broadcast services this is a set of channels; optional network parameters required by the UE to activate the service, e.g. information related to the content delivery and control channels; as well as

UE - Media Control Functions (Xc)

The Xc reference point is a logical end-to-end reference point between the UE and the IPTV Media Control Function, used for the exchange of media control messages, for the control of IPTV Media flows (eg. starting playback, stopping playback, etc).

UE - Media Delivery Functions (Xd)

The Xd reference point is a logical end-to-end reference point between the UE and the IPTV Media Delivery Function, used for the delivery of media data.

The Xd reference point may also carry QoS and/or QoE reports from the UE to the MDF if the IPTV system architecture supports QoS and/or QoE. These reports relate to the quality of the IPTV Media Data that the MDF is providing to the UE.

Core IMS - IPTV Service Media Functions (y2)

The y2 reference point is used by the IPTV SCF for the control of the MCF.

IPTV Media Control Function - IPTV Media Delivery Function (Xp)

The Xp reference point is used by the MCF for the control of the MDF and supports session setup when content is distributed across one or more MDFs.

2.2.2 An analysis of the ETSI TISPAN IMS-based IPTV Architecture

The ETSI TISPAN IMS-based set of standards for IPTV is to date the most comprehensive set of standards defined for the implementation of IPTV services in an IMS architecture. The protocols and functional entities are well defined and the standards provide a complete end-to-end IPTV solution. The easy and rapid development of a test-bed, for use in the evaluation section of this thesis, is possible due to the fact that the protocols defined have been IETF standardised for a number of years now and open-source implementations are available for use in development. One drawback however to the use of these protocols is that some were built to be independent, such as RTSP which duplicates the session setup of SIP, and some are dated and in need of extensions to be capable of taking advantage of new IP technologies and new developments in IP multimedia.

Chapter 3

Proposed content adaptation IMS-based IPTV architecture

Chapter 2 discussed four methods which can be used to adapt video content for IPTV services. Chapter 2 also introduced the ETSI TISPAN IMS-Based IPTV architecture, which at present is the most comprehensive set of specifications related to IPTV services for the IMS, with the relevant functional components and reference points highlighted and discussed.

This chapter begins by looking at the desired design requirements of a content adaptation architecture suitable for IMS-IPTV systems before evaluating each of the content adaptation methods, presented in Chapter 2. A hybrid content adaptation scheme is then proposed, which makes use of precoding and trans-content methods in order to provide a complete, scalable content adaption architecture for IMS-IPTV services. Methods to integrate the relevant content adaptation methods into the ETSI TISPAN IMS-Based IPTV architecture are then discussed.

3.1 Design considerations for a content adaptation architecture for IMS-IPTV architectures

Before selecting an appropriate content adaption method for IMS-IPTV architectures, the functional requirements of such an architecture need to be documented and understood. These requirements will help identify the functionalities needed to ensure that i) users are satisfied with the service. ii) network operators are easily able to integrate the scheme within their network and iii) the costs of implementing the scheme will be offset by the benefits gained by the network operator.

The requirements for the proposed content adaptation scheme are listed as follows:

- The primary requirement for the content adaptation scheme is to enable the alteration of video content to ensure:
  - delivery through any access network the subscriber may access the IPTV service from.
  - the video is in a suitable format for playback on a range of user end-terminals.
- The 3GPP and ETSI TISPAN define standard IMS functional components and interfaces. This is necessary to ensure interoperability between IMS components. originating from different vendors. as well as to ensure interoperability between IMS systems deployed by individual operators. The proposed content adaptation architecture should reuse. where possible. these standardised components and interfaces thus limiting the need for new functional entities in the IMS architecture."
- Each content adaptation method comes with its own set of advantages and disadvantages. The content adaptation scheme should seek to find a balance between these advantages and disadvantages. in order to find an optimal method for content adaptation.
- In mobile environments the mobility of users is problematic for video streaming systems. As users move around interference from objects causes signal strength to vary which causes fluctuations in the bandwidth of the communications channel. The proposed content adaption scheme should be able to cater for this by dynamically adapting the video to best match the available channel capacity.
- Different clients desire different levels of interaction with the system. Clients who want minimal interaction with the system should be able to use an

automated process to determine the capabilities of the client terminal and access network, while other users who want to 'tweak' the system should be able to alter parameters according to what they desire. This process should not allow the IPTV service to be compromised and checks need to be in place to ensure that user parameters are within acceptable bounds. Also the network should be able to check that the values the users are using are in-line with the user's subscription. (Users with a subscription for SD content should not be able to access HD content for example).

3.2 Evaluation of content adaption methods

Precoding of content is difficult, even if the operator were to precode content to several popular coding formats, the characteristics of the video may not suit all clients capable of playing back the formats; small devices would require low-bit rate and low spatial resolution video content, while HD TV users would expect high-bit rate, high-quality video. Multiple copies of every title would require large quantities of disk storage. It is possible to use precoding methods for dynamic content adaptation but this leads to a complicated encoding process, since synchronisation points need to be added to all copies of the video.

Scalable video coding and multiple description encoding are perhaps the most attractive solutions for enabling content adaptation in the IMS, allowing the adaption of the video to be done with minimal processing requirements. A major drawback is however the lower coding efficiency of SVC and MDC encoders which causes SVC/MDC generated bit-streams to have higher storage requirements than single layer bit-streams. However, the amount of storage space required would be far less than that required by precoding. The H.264 scalable extension as well as MDC encoders allow the quality of the content to be dynamically altered and hence can be used to adapt video streams to varying link conditions in mobile environments. SVC allows for graceful quality degradation; in the event of higher-layer packets being lost during transmission the decoder is still able to decode the lower layers it has received. The drawback to this is that the highest quality of the video is determined by the highest successive layer received. MDC on the other hand allows for the decoding of all received streams and the MDC decoder can attempt to predict the lost stream information by using the received descriptions. This helps MDC provide possible higher picture quality than SVC in the event of packet loss.

Despite the benefits of SVC coding methods, they have found little success in industry and even with widespread adoption of the H.264/MPEG-4 AVC standard there are presently no devices, apart from computer software, which support the scalable H.264 SVC extension. Similarly, solutions that make use of MDC encoding require the device to have an appropriate MDC decoder, or additional software to be downloaded and installed. Unlike SVC however, there are currently no MDC formats standardised which requires the use of proprietary MDC formats and third-party software to be installed on subscriber IPTV terminals. The result is that SVC and MDC methods are unsuitable for content adaptation in the IMS at present as so few devices are able to support either of them.

Trans-coding is the dominant method for content adaptation at present. Unfortunately the high processing requirements of the method means that operators and service providers will need to spend large amounts of capital in order to obtain high-end systems, capable of trans-coding many multimedia streams. Trans-coding does however allow the adaptation of multimedia content to almost any format, temporal resolution, spatial resolution and/or fidelity that a client may require.

3.3 Proposed hybrid content adaptation architecture for IMS-based IPTV

From the previous section we noted that each of the content adaption methods discussed in Chapter 2 have both major advantages and disadvantages. None of the literature presented in Chapter 2 however considered a combination of different content adaptation methods. A hybrid content adaptation scheme, consisting of two or more of these methods, could benefit from the advantages each content adaptation method used.

The proposed hybrid content adaptation architecture developed for this thesis, uses both precoding and trans-coding methods to adapt video content for IPTV users. The precoding of content is achieved by creating three versions of the source content: a high-definition (HD) version, standard definition (SD) version and low definition (LD) version. The definitions of the different versions refer to the resolution of the video. HD video is considered to have a resolution of at least 1024×768, while the SD and LD versions of the video should have resolutions of around 800x600 and 320x240 respectively. These three versions of the stream cater for subscribers in different domains, for example the HD version will likely be suitable for fixed-line access networks and devices that are connected to HD displays, while the low-definition version may be suitable for cell-phones and hand-held multimedia devices in low bit-rate access networks. Each version is encoded using an H.264 encoder. The AVC standard was chosen since it has received a lot of attention from industry and has been adopted by various standards bodies for use in video applications. The AVC standard is expected to be the predominant video standard used in the near future [24] and many devices currently support the standard.

The use of H.264/MPEG-4 AVC encoded content should ensure that the majority of users are able to access the stream without requiring content adaptation to be performed. However, if none of the three versions are suitable for the user's equipment, or if the channel conditions are varying, trans-coding should be employed. The version of the content used for the trans-coding process is dependent on the subscribers connection and terminal characteristics: i.e. if the user has a mobile phone, which essentially requires a low definition version of the stream, then the LD version should be used for the trans-coding process. To further improve this scheme, methods such as those used in [3] could be used to reduce the requirements of the trans-coding process and optimise performance.

It should be noted that H.264 scalable profiles could also be incorporated into this content adaptation architecture. An SVC copy of the stream could be located at the MDF and used to cater for any client capable of supporting the scalable coding format. However, since only a handful of devices currently support the scalable standard, its use is unlikely to add value to the proposed content adaption scheme in the very near future.

3.3.1 Integrating the hybrid content adaptation scheme with TISPAN IMS-IPTV architecture

The previous section outlined the proposed hybrid content adaptation scheme presented in this thesis. This section now discusses how this scheme can be integrated into the ETSI TISPAN IMS-Based IPTV architecture outlined in the previous chapter.

Selection of precoded content

Section 7.1 of ETSI TS 182 027 [4] classifies the user data information stored by the IMS operator. Of interest are the User Equipment (UE) settings, stored under each subscriber's IPTV Profile. These UE settings include information related to the capabilities of the all the UEs that an IPTV subscriber is associated with (a subscriber may have more than one UE thus unique ID is given to each). The UE capabilities may be used by the Service Selection Function (SSF) for customising IPTV services to suit the subscribers current UE. For instance, an IPTV user on a SD-only device would not be provided with information related to HD services. Figure 3.1 depicts the process whereby the UE attaches to the IPTV service and then selects the relevant SSF.

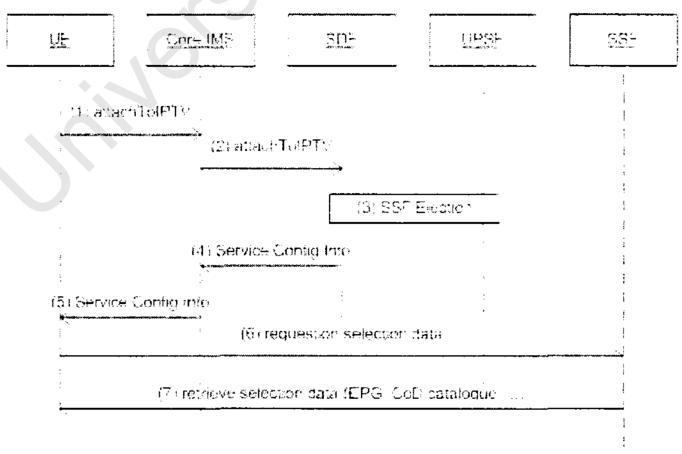

Figure 3.1: IPTV service attachment and selection in pull mode [4].

Referring to Figure 3.1:

1-2) UE sends initial invite towards the SDF which includes the UE's unique ID.

3) The SDF determines the SSF (or SSFs) that are appropriate for the UE's capabilities, the user's profile and also the location of the UE. The SDF accesses the UPSF for the subscriber's IPTV service information or any other entity which holds the information.

4-5) The SDF replies with the service configuration information, including the SSF that the UE should request the selection data from.

6) The UE requests the selection data from the SSF.

7) The SSF provides the UE with customised electronic program guides (EPG) and CoD catalogues that point towards the appropriate precoded versions of the content. Other information may be present in the EPG and CoD catalogues, including information about the content media streams such as encoded format, profile level, bit-rate, etc.

When selecting an IPTV broadcast service or IPTV on-demand content, the UE maps the information received from the SSF into SIP and SDP parameters as shown in Figure 3.2. The identifier for the appropriate precoded content is placed in the Request-URI field of the initial SIP INVITE message (highlighted in red in Figure 3.2), which the core IMS uses to determine the SCF the INVITE should be forwarded to. Other parameters are mapped to the media description lines in the SDP message, attached to the initial SIP INVITE.

```
INVITE sip:james_bond_casino_royale_ld@iptv.open-ims.test SIP/2.0
Via: SIP/2.0/UDP 137.158.125.235:5060;rport;branch=z9hG4bK222324647
Route: <sip:orig@scscf.open-ims.test:6060;lr>
From: "Robert Marston" <sip:rob@open-ims.test>;tag=214361087€
To: <sip:james_bond_casino_royale_ld@iptv.open-ims.test>
Call-ID: 310038184
CSeq: 20 INVITE
Contact: <sip:rob@137.158.125.235:5060>
Content-Type: application/sdp
Max-Forwards: 70
User-Agent: UCT IMS Client
Subject: IMS CoD Session
Expires: 120
Content-Length: 317

v=0
o=- 0 0 IN IP4 137.158.125.235
s=IMS IPTV CoD Session
c=IN IP4 137.158.125.235
t=0 0
m=audio 6628 RTP/AVP 14
b=AS:128
a=rtpmap:14 MPA/90000
m=video 6630 RTP/AVP 97
b=AS:1600
a=rtpmap:97 MP4V-ES/90000
a=framesize:97 320-240
a=framerate:15
m=application 9 TCP iptv_rtsp
a=setup:active
a=connection:new
```

Figure 3.2: Initial SIP INVITE for IPTV CoD service with attached SDP content.

Trans-coding

If the UE requires the video content to be adapted (e.g. does not support the H.264/MPEG-4 AVC standard, has an uncommon display resolution, etc), it should place additional attributes in the media description lines indicating how the content should be adapted. Figure 3.2 shows an example of the SDP content message where the user requests the video of the content to have a resolution of 320 by 240 pixels and a frame-rate of 15 frames per second (highlighted in blue). The bandwidths for the audio and video streams are also present (highlighted in green).

Upon receipt of the INVITE, the MCF examines the SDP and determines that the client is requesting adaptation of the video content, it instructs the MDF to trans-code the content to a format suitable to the client device. In the example shown in Figure 3.2, the original content has a frame-rate of 24 frames per second but the client's device would prefer a frame rate of 15 frames per second. The MDF then selects the LD version of the content, since it notes the Request-URI indicates the client terminal requires low definition content. It may not always be the case that the UE has the information concerning the content delivery channels; ETSI TISPAN standards allow the network operator to choose whether or not to provide this information. If this is the case, then the UE may configure the media description lines in the SDP to be in-line with its capabilities. Also, RFC 2327 [29] allows for multiple payload types to be specific for the different media components of the stream. So for example, if the client supports both H.264/MPEG-4 AVC and MPEG-4 ASP coding formats, it could specify this under the video m-line in the SDP. Figure 3.3 shows an example of this.

```
m=video 6530 RTP/AVP 96 97
b=AS:52000
a=rtpmap:96 H264/90000
a=fmtp:96 packetization-mode=1;profile-level-id=640028;sprop-parameter-sets=Z2QAKKw8QJAUAW6EAAAPpAAC7gI8YMYm,a06yvLA=;
a=framesize:96 320-240
a=rtpmap:97 MP4V-ES/90000
a=fmtp:97 profile-level-id=3; config=000001200008687ffff0aad89c2160a31;
a=framesize:97 320-240
a=framerate:15
```

Figure 3.3: Multiple video payload types in SDP.

Figure 3.4 shows the complete signalling for setting up of an IPTV session; from the initial INVITE all the way through to the delivery of the media flows from the MDF to the UE.

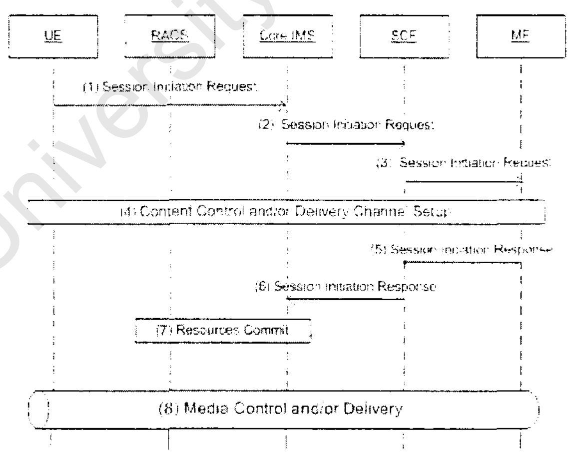

Figure 3.4: Signalling flow for initialisation of an IPTV session [4].

3.3.2 Relating the functionalities of the proposed hybrid adaptation scheme back to design requirements

In section 3.1 design considerations of a content adaptation scheme were discussed. This section analysis the hybrid content adaptation scheme, outlined in the previous two sections, and relates the functionalities back to these design considerations.

Comprehensive content adaptation procedure

The three precoded versions of the content allow it to be viewed by the majority of IPTV subscribers; however, the use of trans-coding allows for the adaptation of content to multiple formats, profile-levels, bit-rates, spatial resolutions and temporal resolutions. This allows for the hybrid content adaptation scheme to be able to cater for all envisaged client requirements.

Reuse of standard IMS components and reference points

The hybrid content adaptation scheme does not add any new functional entities to the IMS architecture. Instead it reuses the existing Service Selection functions, Media Functions and User Equipment and merely defines how the interaction between these entities should take place.

Optimal content adaptation scheme

The use of H264/MPEG-4 AVC precoding allows for a reduction in the overall processing requirements of the scheme, since the trans-coding process need not be invoked for the majority of users. By limiting the number of precoded versions of the content to three, the storage requirements of the hybrid scheme are kept at a minimum.

Dynamic content adaptation for mobile environments

The trans-coding of content allows the scheme to cater for mobile environments with varying link conditions. QoS reports from the UE allow the MDF to adapt the trans-coding process to cater for any changes in the link bandwidth. Also by using synchronisation points, the MDF can select between the three versions of the content in order to optimise the trans-coding process while ensuring the client receives the best achievable video quality.

Customisation of content adaptation scheme

Customisation of the hybrid adaptation of video content is performed in two stages. The first stage involves storing information related to the subscribers UEs in the profile information stored in the UPSF. This allows the SSF to select the precoded version of the content that is most suitable for the UE which subscriber is currently using. The second stage involves the media negotiation between the UE and the MCF in order to allow for further customisation of the IPTV content (eg. to reduce the frame-rate to ensure playback on devices with low processing power). In both cases the network is able to determine if the users' preferences are acceptable by checking that the values are within an acceptable range and that they are in-line with the users' subscription.

3.4 Chapter Summary

This chapter has discussed the various requirements of a content adaptation architecture for IPTV services in the IMS. Each of the various content adaptation methods were evaluated and each of their advantages and disadvantages described. The precoding and trans-coding methods were determined to be the most suitable for current IMS deployments and a hybrid content adaptation scheme that utilises both methods was proposed. This hybrid scheme allows for a more efficient and scalable content adaptation architecture to be developed. The integration of the proposed scheme with the ETSI TISPAN IMS-Based IPTV architecture was then discussed and the functionalities of the scheme related back to the design requirements outlined in the beginning of the chapter.

Chapter 4

Architecture and Implementation of an Evaluation Framework

In Chapter 3 the ETSI TISPAN IMS-IPTV functional architecture was outlined and a hybrid content adaption scheme was proposed for use within this architecture. This chapter presents the architectural design and implementation of a suitable test-bed for evaluating the trans-coding IMS-IPTV architecture in a practical scenario. First the objectives, requirements and limitations of the evaluation framework are presented. This is followed by a description of the software used to implement the test-bed. Thereafter, the architecture and hardware elements of the test-bed are described.

4.1 Evaluation Framework Objectives

The key objectives of the evaluation framework are:

- To determine the storage requirements of the three precoded versions of the content required by the hybrid content adaptation scheme
- To evaluate the performance of the trans-coding process when adapting different precoded versions of the content.
- To show how the use of the hybrid content adaption scheme will improve the scalability of IMS-IPTV video content adaptation.

- Finally to determine the feasibility of using the hybrid content adaptation scheme proposed in this thesis for the adaption of video content in IMS-IPTV architectures.

4.2 Requirements of the Evaluation Framework

In order to satisfy the objectives of the evaluation framework a suitable test-bed architecture must be designed. This test-bed should satisfy the following requirements:

- It must be integrated with an IMS framework.
- It must implement an IMS-IPTV client conforming to ETSI TISPAN IMS-Based IPTV standards.
- It must implement the IPTV SCF function and Media Functions conforming to ETSI TISPAN IMS-Based IPTV standards.
- It must integrate the hybrid content adaption scheme proposed in this thesis and be capable of adapting video content to different coding formats, spatial resolutions and temporal resolutions.

4.3 Limitations of the Evaluation Framework

Hardware test-beds are limited in size due to associated costs and limitations of the hardware required to construct the test-bed. Simulations are often used for the testing of scalability issues since they allow the number of systems or users to be scaled to amounts that could not be practically implemented in a test-bed. Simulations however do not often take into account all the real world factors (i.e. processing requirements of servers, accurate delay estimates and so forth) but instead choose to focus on those factors which are regarded as the most important. Practical implementations on the other hand allow for a more realistic analysis to be performed and can show trends with just a handful of entities that can be scaled up to draw conclusions on a large number.

4.4 Software

This section provides details on the software used in the test-bed architecture.

4.4.1 UCT IMS Client

The UCT IMS Client is an open-source IMS client developed by students in the Department of Electrical Engineering, at the University of Cape Town. The client is written in ANSII-C and supports a number of functions including voice and video calling, instant messaging, Presence, uploading/fetching of XCAP documents and, before the completion of this thesis, basic IPTV viewing capabilities. The client however required the use of a proprietary IPTV server and lacked standards compliant IMS signalling. Hence one of the outcomes from this thesis has been to modify the client to support ETSI-TISPAN IMS-based IPTV signalling. The client is now capable of interacting with a ETSI-TISPAN compliant SCF and MCF for the setup, control and termination of IPTV services. The extensions made to the client are listed below.

EPG capabilities

The UCT IMS client has been extended to extract, parse and display an XML formatted EPG which contains listings of IPTV broadcast channels and on-demand content. Users can use the EPG to browse and select the broadcast channel or CoD item they wish to view. Figure 4.1 shows the display of the EPG in the client.

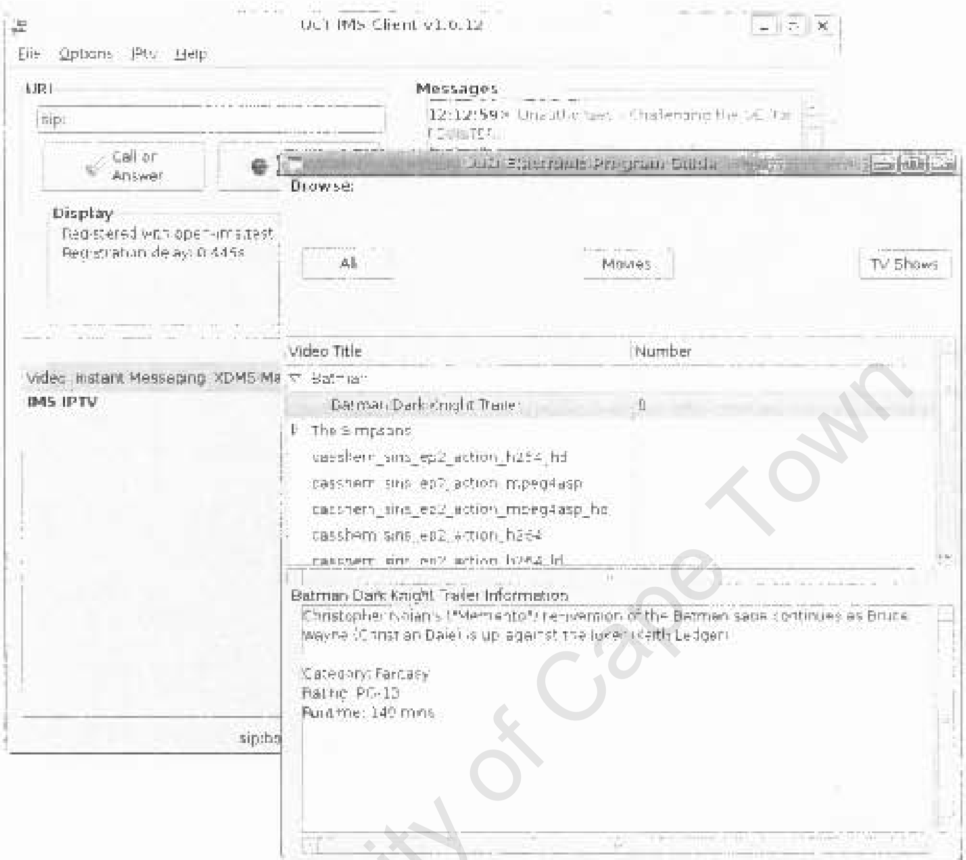

Figure 4.1: UCT IMS Client EPG

ETSI TISPAN IMS SIP signalling

For communication between the SCF and MCF the client was modified to use SIP signalling as defined in the IMS-IPTV ETSI TISPAN standards [5]. This includes the establishment and teardown of an IPTV session with the SCF, as well as the negotiation of media delivery and control channels between the client and the MCF.

In addition to the SIP signalling, RTSP signalling was implemented in the client. ETSI TISPAN standards define two methods for RTSP control of IPTV media streams. This is because RTSP was developed as a media session setup and control protocol and hence the full session establishment procedure of RTSP duplicates the session setup achieved with the SIP protocol in the IMS. Method 1 uses a subset of RTSP signalling messages for the trick-play functions and omits the session setup commands. Method 2 uses the full range of signalling messages defined in the RTSP RFC and the client uses the libvlc [30] library. For Method 1 it was not possible to extract out just the trick-play functions of the libvlc library, and separate RTSP signalling methods were added to the client to handle trick-play functions when the client communicates with the MCF using Method 1.

Figure 4.3 and 4.3 show the signalling for the two methods.

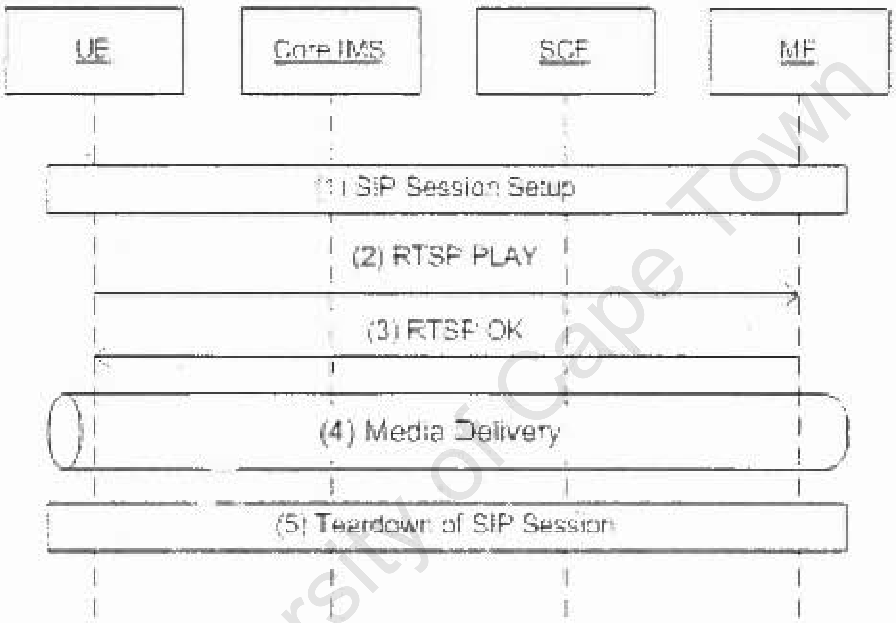

Figure 4.2: ETSI TISPAN IPTV - RTSP Method 1 signalling.

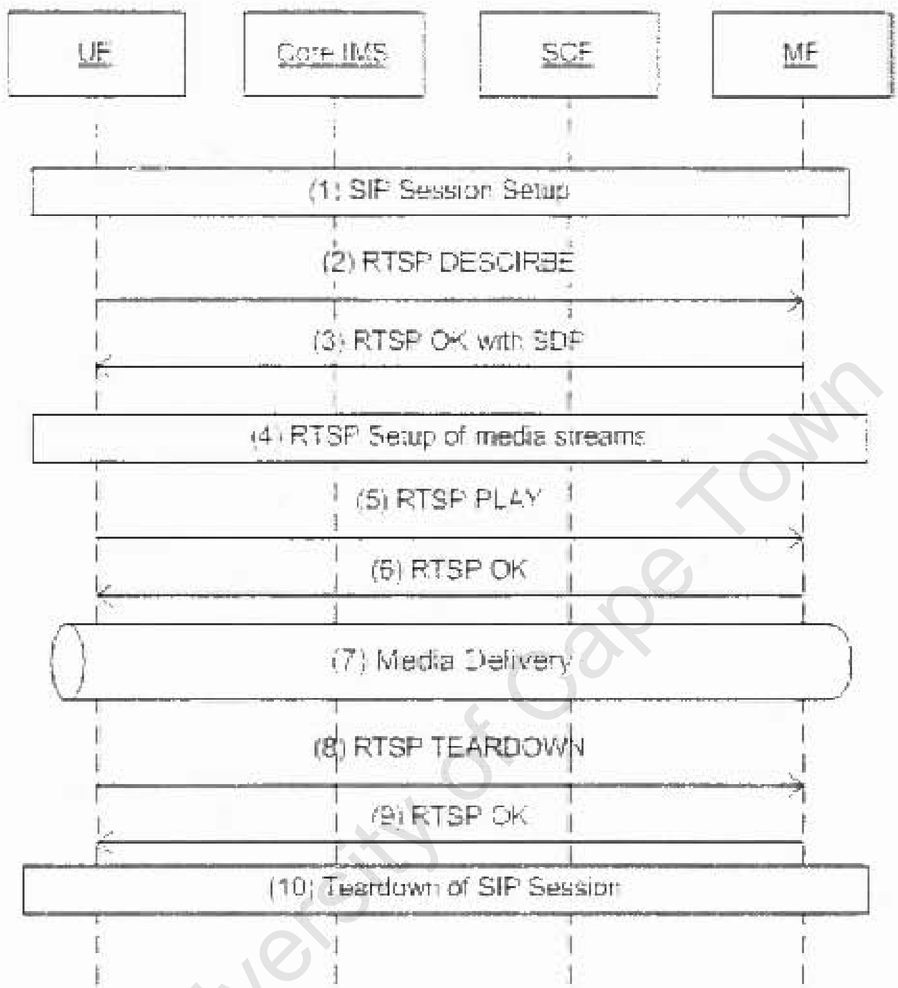

Figure 4.3: ETSI TISPAN IPTV - RTSP Method 2 signalling.

Decoding and display of multimedia streams

The client was extended to make use of the libvlc library for receiving, decoding and display of multimedia content. The libvlc API supports a range of coding formats, including [31]:

- MPEG-1/2
- DIVX (1/2/3)
- MPEG-4, DivX 5, XviD, 3ivX D4

- H.264
- Sorenson 1/3 (Quicktime)
- DV
- Cinepak Theora (alpha 3)
- H.263 / H.263i
- MJPEG (A/B)
- WMV 1/2

Configuration of video preferences

Options for bandwidth, spatial resolution, frame-rate and encoding format were added as configurable preferences in the client. These allow users to customise the video they wish to receive and for content adaptation testing in the test-bed.

4.4.2 Hybrid MCF/MDF

The MCF/MDF software consists of an MCF module and MDF module both written in ANSII-C for the Linux platform. Although ETSI TISPAN standards separate out the two functions and point to the fact that the MDF need not be co-located on the same hardware as the MCF, this functionality was not needed for this thesis and it was decided to implement the MCF and MDF as a single software entity.

MCF Module

The MCF module is responsible for handling the SIP signalling, using the eXosip libraries [32], as well as parsing SDP messages from clients to determine the clients required video parameters such as bit-rate, spatial resolutions, etc. The MCF makes use of the osip libraries [33] for the SDP parsing and editing.

MDF Module

The MDF module is responsible for trans-coding and streaming multimedia content to the client and uses the libvlc library [30].

4.4.3 Franhofer Fokus Open IMS Core [2]

The Open IMS Core consists of a number of software programs to implement the three IMS Call Session Control Functions (CSCFs), as well as a lightweight Home Subscriber Server (HSS). Together these elements form the core elements of all IMS/NGN architectures which are specified within 3GPP, 3GPP2, ETSI TISPAN and the PacketCable initiative. Clients register with the Open IMS core before interacting with the IPTV service. The Open IMS core is responsible for the routing IMS SIP signalling between the client, SCF and MCF. It is developed by the Franhofer Fokus Institute and released as open-source software intended to create interest and further development of IMS technologies. The Open IMS core has an active community of developers and has become popular for use in IMS test-beds, which makes it a reliable reference architecture to test with. Figure 4.4 shows the Open IMS core architecture.

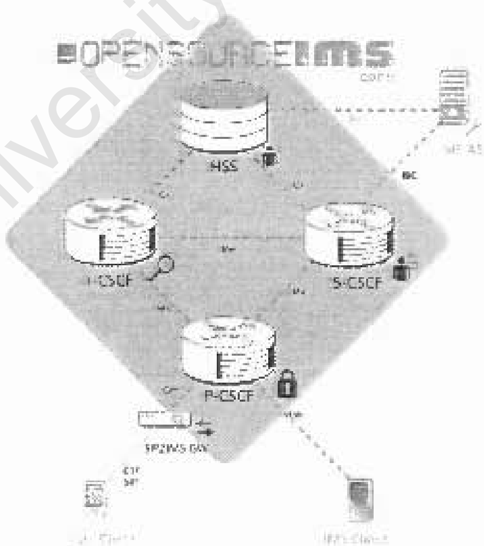

Figure 4.4: Fraunhofer Open IMS core architecture [2].

4.5 Hardware

The test-bed architecture is shown in Figure 4.5. It consists of seven machines connected via a 10Mbit/s Ethernet connection. One machine runs all CSCFs and HSS of the Open IMS core with separate machines used to run the simple SCF software and the hybrid MCF/MDF software. Four machines are used for the client terminals, all of which run the extended UCTIMSClient software to support the features called for in this thesis.

The details of each machine used can be found in Appendix 2.

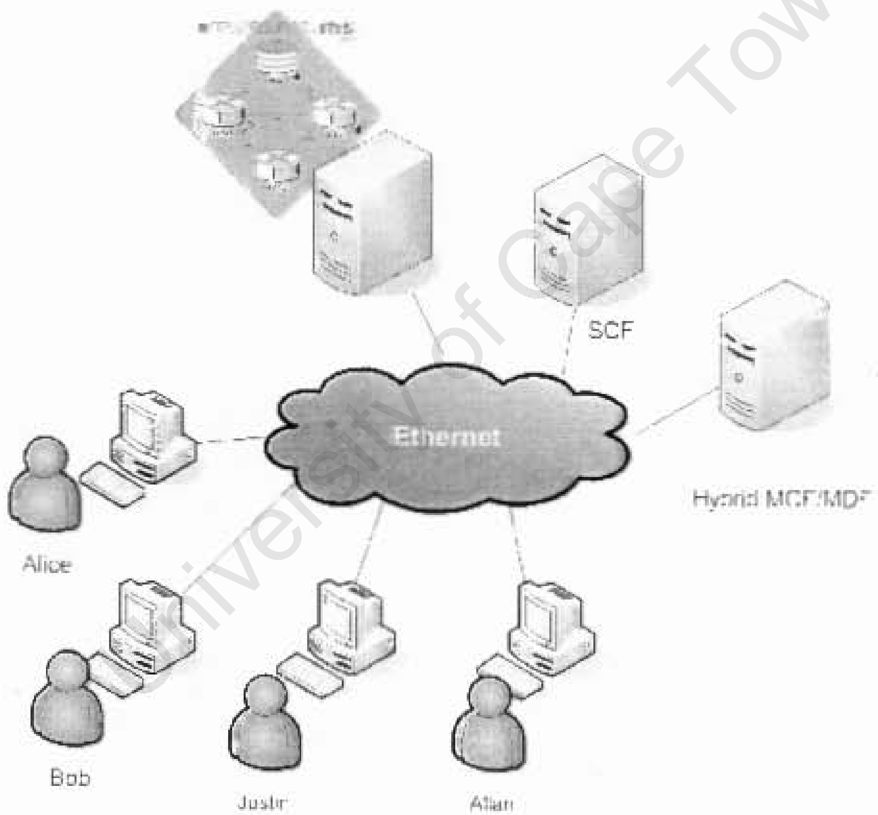

Figure 4.5: IMS-IPTV content adaptation test-bed architecture.

4.6 Chapter Summary

This chapter outlined the design objectives of the evaluation framework presented in this thesis and discussed the requirements and limitations of the framework.

The test-bed architecture was given and the software and hardware used for implementation of the test-bed discussed. These details should allow for the replication of the test-bed architecture, duplication of tests and results and for further IMS-IPTV research to be conducted.

Chapter 5

Evaluation Results and Analysis

In Chapter 4 the architectural design of the IMS-IPTV hybrid content adaptation test-bed was presented. Using this test-bed a number of tests were performed to assess the hybrid content adaptation scheme and determine the feasibility of its use for adapting video content in IMS-IPTV architectures.

The chapter begins with a discussion of the tools and video sequences used in the testing. This is followed by a discussion on the influence of precoding on the three different versions used in the hybrid content adaptation scheme. Trans-coding to and from H.264 and another popular coding format is contrasted. This is followed by an outline and discussion on two further test scenarios: the first analyses the trans-coding process when adapting each of the three precoded versions of the test sequences with the second further analysing the trans-coding of HD and SD content.

5.1 Tools used for testing

Avidemux [34] is a free open-source program used for multi-purpose video editing and was used to create the precoded video sequences.

The Dstat [35] tool was used to capture the CPU load on the machine used for the hybrid MCF/MDF.s. For all test involving the load on the CPU the idle state column was subtracted from a hundred to get overall load on the CPU. All tests were run for a period of 50 seconds with Dstat reporting on the CPU load at one second intervals. Dstat also allowed the saving of reports to a Comma

Separated Value (CSV) file format, which were imported into Microsoft Excel for the generation the graphs shown in this chapter. The disadvantage of using Dstat is that it reports only the total CPU usage and not the individual CPU loads of each process. Other tools that allowed this were trialed but it was found they did not report on the trans-coding processes that were forked by the MCF/MDF.

5.2 Video test sequences

Three video sequences were used in the testing of the hybrid content adaptation scheme. Each sequence was chosen for its unique characteristics in order to gather results from a range of video sequences. The first sequence, taken from the movie "Die Hard 4", involves a car chase sequence with a large amount of motion and many scene changes. This type of content is typically associated with high-bit rates as the video encoder cannot exploit temporal locality well, due to the rapid movement of characters/objects and constant scene changes. The second scene is taken from the movie "Gladiator" and involves dialogue between two characters with little movement and only occasional scene changes. This generally allows for a lower bit-rate, since very little information changes between successive frames. The last sequence is taken from an animated series called "Casshern Sins - Episode 2" and is an action sequence with a good deal of movement and scene changes. Animation sequences are typically not as detailed as real-world captured content.

Each of the HD sequences was captured using Avidemux from HD 720 H.264/MPEG-4 AVC sources. Each HD sequence was then re-encoded using the Avidemux tool to generate a further two versions: the SD sequence with resolution of 800x600 and the LD sequence with resolution of 320x280. The frame-rate of the LD and SD versions were kept identical to the original HD sequence at 23.976 fps. Additionally each of the LD test sequences were converted MPEG-4 ASP sequences for the comparison between the differences in trans-coding from H.264 and MPEG-4 encoded video. Each of the video sequences has a length of approximately 60 seconds[1].

Appendix C gives more details regarding each video test sequence.

[1]Since the content was taken from an encoded video source, the sequences were broken on key-frames as close to the minute mark as possible.

5.3 File sizes of the test sequences

The main disadvantage associated with the precoding of content, is that it requires the content server to have more storage space to hold the different versions of the video sequences. The hybrid content adaptation scheme aims to minimise the amount of storage space required by limiting the number of copies, per sequence, to three. This section describes the storage space required to store each of the test sequences discussed in Section 5.2. Table 5.1 shows the files sizes of each precoded test sequence, with Figure 5.1 showing the total storage space required for each copy of the three video sequences.

Version	HD	SD	LD
Die Hard Sequence	17506804	10300526	3455188
Gladiator Sequence	3625290	2032798	1292616
Animation Sequence	7579772	4177920	2014494

Table 5.1: File sizes (in bytes) of the various precoded test sequence.

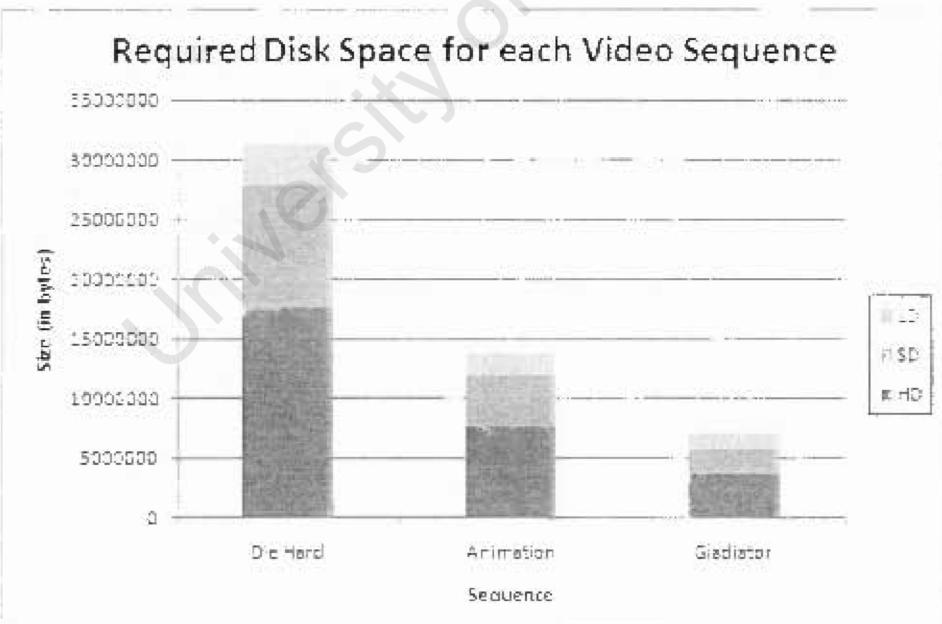

Figure 5.1: Graph showing the contribution of each precoded version to the total disk size required for each sequence.

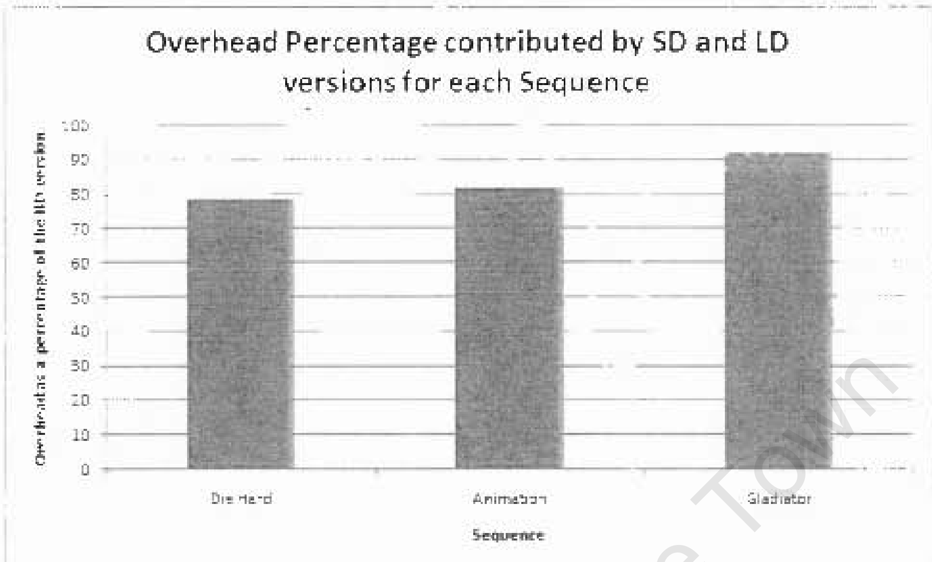

Figure 5.2: Overhead as a percentage of the HD version contributed by the SD and LD versions of the content.

As HD television services have become popular and many content providers wish to offer true HD video content, in such a case a single HD version of the content would typically be stored when not using precoding. The combined total of the SD and LD file sizes can therefore be seen as the storage overhead associated with the use of precoding in the hybrid adaptation scheme. Figure 5.2 shows the storage overhead for each of the test sequences.

The storage overhead associated with the LD and SD precoded versions of the Die Hard 4, animation and the Gladiator sequences are 78.57, 81.7 and 91.73 respectively. From these results it can be estimated that the total disk space required for the hybrid content adaptation scheme is roughly double that of a scheme that only uses one single-layer HD encoded sequence. This is however still significantly less then what would be required for a precoding scheme that did not use a maximum of three precoded sequences.

There are a few interesting points to note from the file size variations of video sequence. The Die Hard - action sequence has the largest file sizes which can be attributed to the large amount of motion and scene changes in the sequences. The animation sequence ranks second in terms of file size as it also contains a large amount of motion and scene changes. The file sizes of the Gladiator sequences are considerably smaller, due to the low amount of movement and very few scene changes in the sequence, allowing the sequences to be highly compressed. Following on from this it can be seen that the percentage overhead increases with a decrease in the file size. This can be attributed to the loss of information being greater for higher bit-rate sequences when converted to a lower quality.

5.4 Comparison between trans-coding from MPEG4-ASP and H.264/MPEG4-AVC

The justification for the use of H.264 encoded content in the hybrid adaptation scheme is due to the widespread adoption of the standard by industry and the large amount of devices that already support the standard. The Advanced Simple Profile (ASP) of the MPEG-4 Visual standard is another well supported standard with which the video could be encoded. This section contrasts the trans-coding from each of the two formats and is intended to show why the use of H.264 encoded content is preferable.

5.4.1 Experimental procedure

Initially a single client was used to request the H.264 encoded LD test sequences to be converted to MPEG-4 ASP streams, with the load on the server being measured during the trans-coding process. This is then repeated but with the client requesting each of the LD MPEG-4 ASP encoded test sequences to be converted to H.264 encoded video streams instead. In order to investigate how each of the encoders CPU requirements changes with an increase in the number of clients, the tests were repeated three times with the number of clients increasing in increments of one.

5.4.2 CPU load graphs for H.264 and MPEG-4 ASP coding comparison

The following three figures show the difference in load on the CPU of the MDF when trans-coding from H.264 to MPEG-4 ASP, and vice versa.

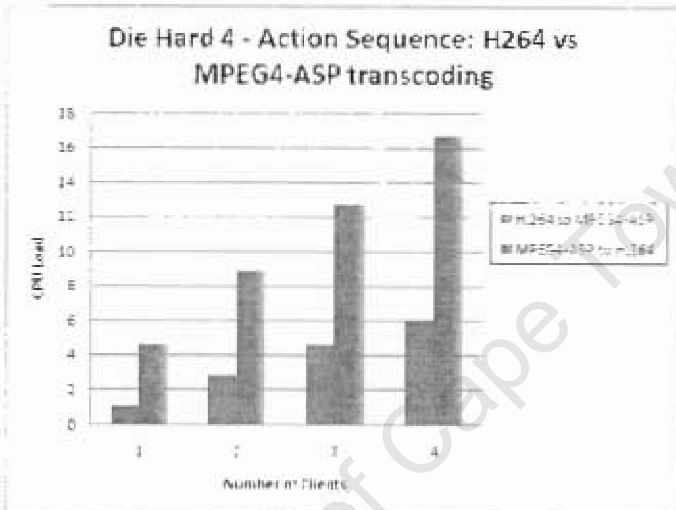

Figure 5.3: Comparison between trans-coding from H.264/MPEG4-AVC to MPEG4-ASP and vice versa for the "Die Hard 4" - action sequence.

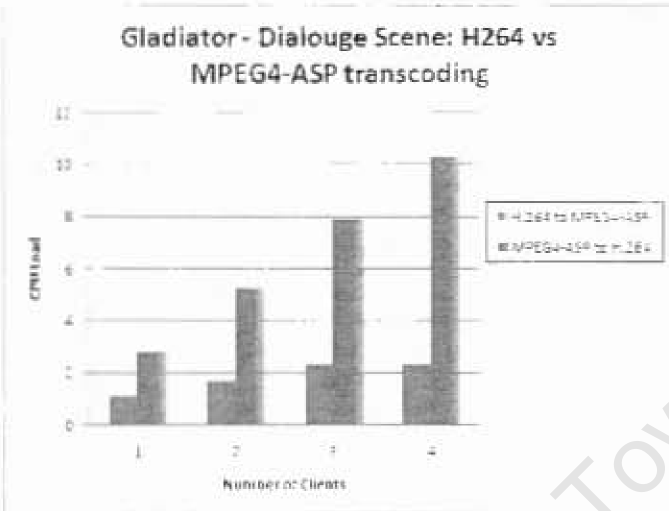

Figure 5.4: Comparison between trans-coding from H.264/MPEG4-AVC to MPEG4-ASP and vice versa for the "Gladiator" - dialogue sequence.

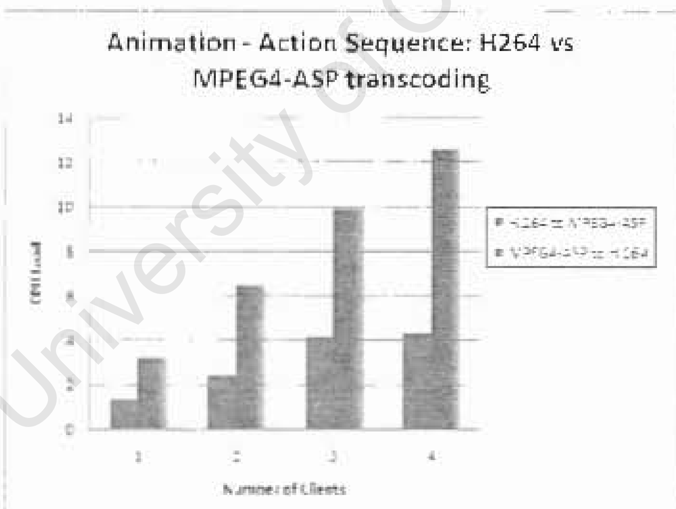

Figure 5.5: Comparison between trans-coding from H.264/MPEG4-AVC to MPEG4-ASP and vice versa for the animation - action sequence.

5.4.3 Discussion of results

We can see from the figures in the previous section, that encoding video to H.264 requires more than double the amount of processing power than when encoding to MPEG-4 ASP. The newer H.264 standard makes use of more computationally expensive coding methods than the older MPEG-4 ASP standard, allowing it to achieve higher quality video at lower bit-rates. Subscribers would thus prefer to receive H.264 encoded bit-streams given the higher quality video generally associated with them. Even if MPEG-4 ASP were to be used, the higher computational costs of trans-coding MPEG-4 ASP bit-streams to H.264 would lead to a less scalable solution as the number of devices that support the H.264 standard increases.

5.5 Trans-coding performance test results

Trans-coding is typically associated with requiring large amounts of processing power. In this section we examine the load placed on the MDF CPU when adapting each of the three precoded versions to a low definition version of the content. Trans-coding is performed using both H.264 and MPEG-4 ASP, to show the difference between trans-coding to different formats. The aim of these tests is to illustrate the advantages of using separate precoded versions of the content and not trans-coding from a single HD source.

5.5.1 Experiment procedure

In this experiment one client was used to request the trans-coding of the LD version of the stream to an H.264 encoded stream, with a spatial resolution of 320x240 and frame-rate of 30: the change in frame-rate is to ensure that the stream is trans-coded since the LD H.264 encoded version is already at the required spatial resolution and the MCF would merely stream it. The experiment was then repeated with the number of clients increasing with each experiment, and then this process is repeated with the SD and HD versions of the test sequences. These experiments were then re-run with each of the clients then requesting the video to be converted to MPEG-4 ASP.

5.5.2 CPU load graphs showing trans-coding between different precoded versions of the test sequences

Trans-coding to H.264 AVC streams

The following graphs show the trans-coding of the test sequences to H.264 AVC streams.

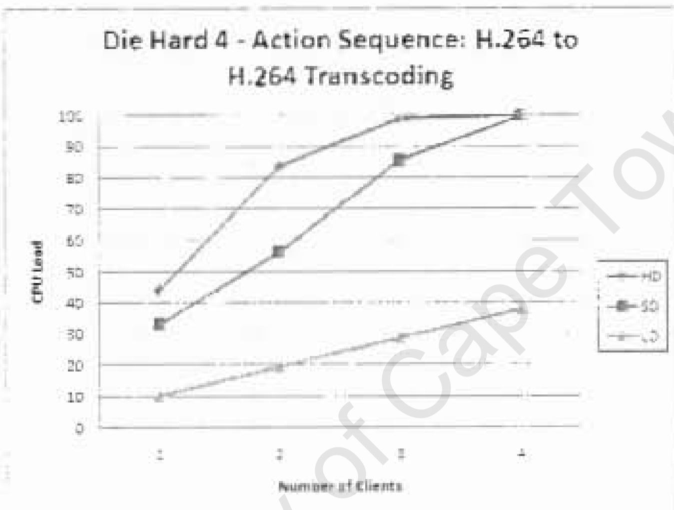

Figure 5.6: Graph showing the load on the CPU with 1 - 4 clients requiring H.264 streams of the "Die Hard 4" - action sequence.

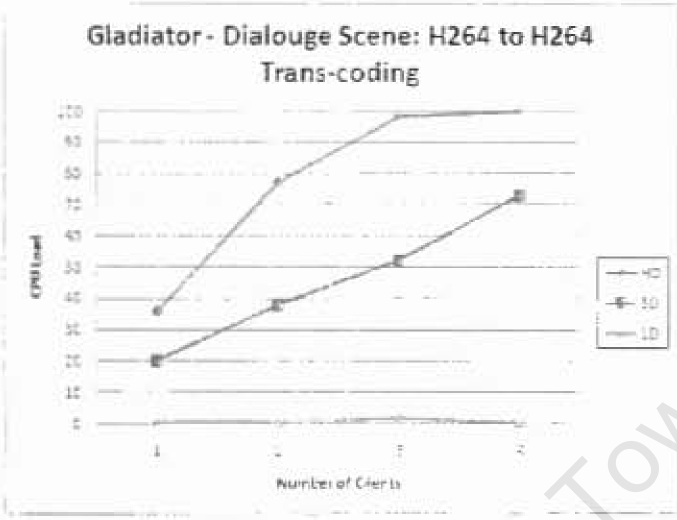

Figure 5.7: Graph showing the load on the CPU with 1 - 4 clients requiring H.264 streams of the "Gladiator" - dialogue sequence.

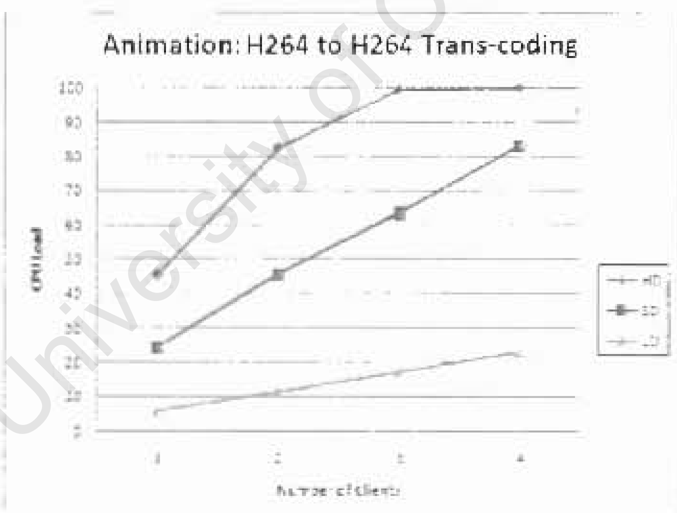

Figure 5.8: Graph showing the load on the CPU with 1 - 4 clients requiring H.264 streams of the - "Casshern Sins" animator sequence.

From the above graphs we can see that in each of the scenarios, the trans-coding from the HD and SD version of the content requires a significantly higher percentage of processing power then when trans-coding from the LD version. In the Die Hard 4 sequence trans-coding tests we see the CPU usage is near 100%

for both HD and SD sequences with only 3 clients requiring trans-coding. The Gladiator sequences was slightly less, due to the lower-bit rate of the video, but was still significantly high for both HD and SD sequences with 4 clients consuming more than 70% CPU in all the tests. Similarly the animation sequence shows very high CPU usage for both the HD and SD sequences.

Trans-coding to MPEG-4 ASP streams

The following graphs show the trans-coding of the test sequences to MPEG-4 ASP streams.

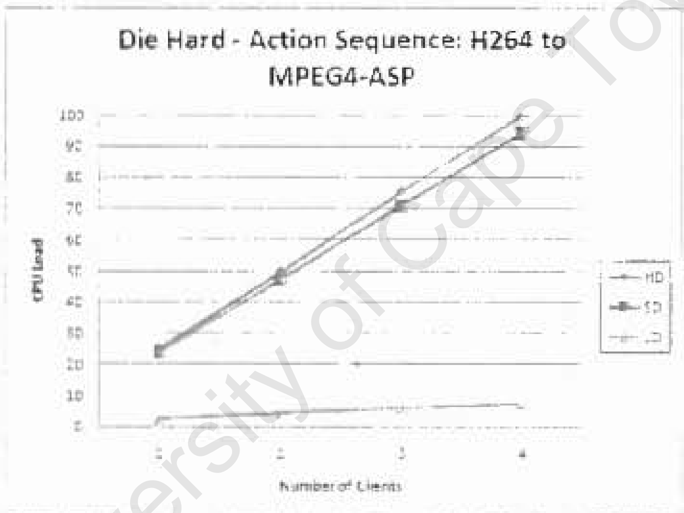

Figure 5.9: Graph showing the load on the CPU with 1 - 4 clients requiring MPEG4-ASP streams of the "Die Hard 4" - action sequence.

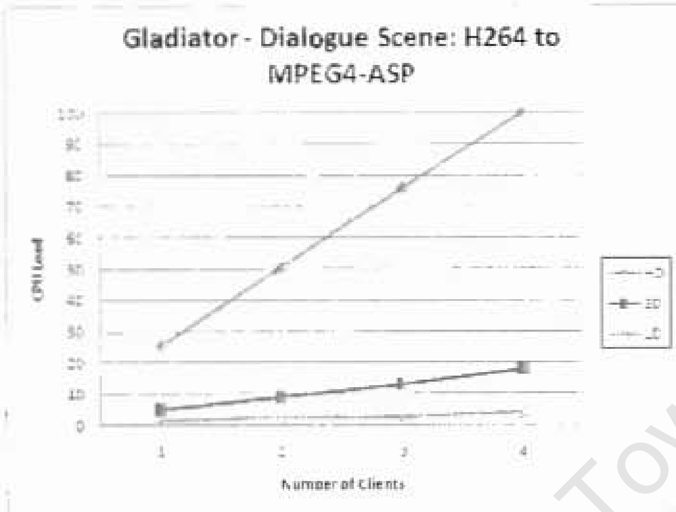

Figure 5.10: Graph showing the load on the CPU with 1 - 4 clients requiring MPEG4-ASP streams of the - "Gladiator" dialogue sequence.

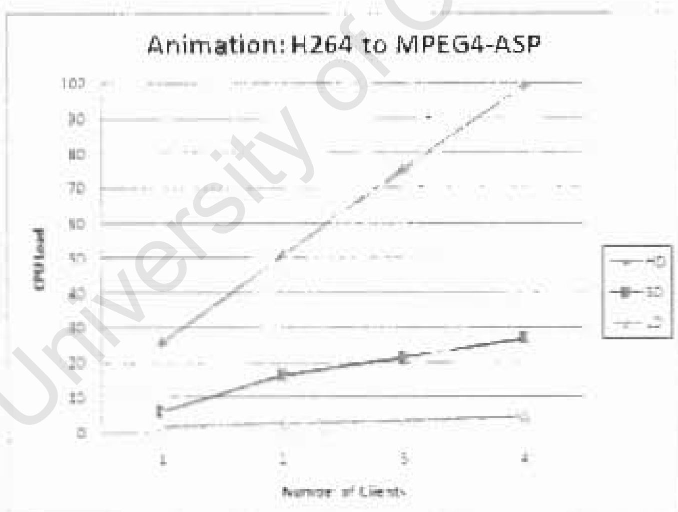

Figure 5.11: Graph showing the load on the CPU with 1 - 4 clients requiring MPEG4-ASP streams of the - "Casshern Sins" animation sequence.

Apart from the lower processing power required to encode the MPEG-4 ASP video, the same trends seen with the H.264 trans-coding can be seen here. Trans-coding from the lower definition versions of the stream are significantly lower.

5.5.3 Discussion of results

Clearly the use of three separate versions of the content allows for a major decrease in the processing power required for the trans-coding process. This results in an increase in the scalability of the system with a single MDF being able to support a number of more clients when trans-coding from lower definitions of the streams. The high processing requirements for the SD and HD versions of the streams are troubling since, although not optimised for trans-coding, the MDF machine is at present considered a high-end system. The results show that the use of the hybrid content adaptation scheme will work well in mobile environments where the limited access speeds and devices with small screen resolutions and limited processing power will require the LD version of the content to be trans-coded.

5.6 Trans-coding HD and SD test sequences to video with altered aspect ratio

In Section 5.5 it was noted that the trans-coding process for the HD and SD test sequences required a large amount of processing power. One typical application for trans-coding of the HD and SD sequences is the changing of the aspect ratio to fit the display of the IPTV user's end terminal. Typically these displays come in wide-screen (16:9) and standard (4:3) aspect ratios. The trans-coding process may be used to adapt from the wide-screen format to the standard format, or vice versa. The tests carried out in this section are intended to determine the load placed on the CPU when trans-coding wide-screen HD and SD test sequences to a 4:3 aspect ratio. The aim of which is to determine the feasibility of trans-coding HD and SD content.

5.6.1 Experiment procedure

In this experimental scenario, four clients require the wide-screen test sequences to be adapted to a standard 4:3 aspect ratio. Both the HD and SD test sequences are evaluated with the HD versions changed to a resolution of 1280x960 and the SD versions being changed to a resolution of 640x480. The frame-rate of the trans-coded stream is kept the same as the original sequence.

5.6.2 Graph showing the load on the MDF CPU when trans-coding between different aspect ratios

Figure 5.12 shows the CPU loads during the trans-coding of content to 4 clients for each of the HD and SD test sequences.

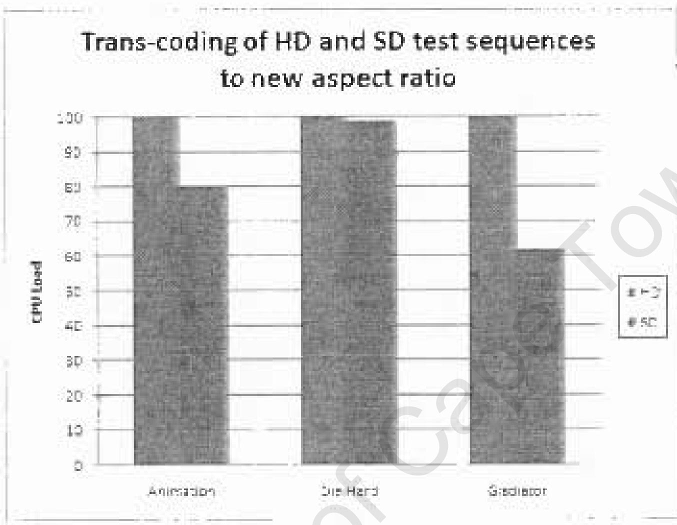

Figure 5.12: Trans-coding for HD and SD test sequences to 4:3 aspect ratio.

5.6.3 Discussion of Results

Even on the high-end machine used for the MDF, with just 4 clients the system is at approximately a hundred percent CPU usage for all the HD test sequences. The results for the SD sequences are similarly unsatisfactory as the "Die Hard" action sequence requires almost hundred percent CPU usage. From these results it can be seen that trans-coding of HD and SD content requires expensive high-end systems and trans-coding of these sequences should be avoided.

It should be noted however that many of the devices that are targeted by the HD and SD versions will be able to view the streams, but the parameters of the video will not be ideal for each of the devices. For example a device with a 4:3 display may not be able to take full advantage of the screen area with 16:9 content but the video will still be viewable. Reducing the number of clients that required the HD and SD streams to be trans-coded will be critical to ensuring scalability advantages of both content adaptation methods.

A suitable evaluation framework was developed to assess the performance of the hybrid content adaptation scheme proposed in this research and the test-bed was able to successfully perform the evaluations required for this research. The results obtained show that the hybrid content adaptation scheme suffers from higher storage overheads when compared to other methods such as using trans-coding or scalable/multiple description coding. Considerable gains however where seen when trans-coding from lower definition, precoded versions of content.

6.2 Conclusions

Based on the findings in the preceding chapters of this thesis, the following conclusions are drawn:

- No single content adaptation technique is suitable for use in IMS-IPTV implementation. Precoding is associated with large amounts of storage overhead, SVC and MDC are not well supported at present and trans-coding has significantly high processing overheads. A hybrid scheme, which incorporates more than one of these methods, is therefore needed to leverage the advantages of each.

- The use of H.264/MPEG-4 AVC encoded video in the proposed hybrid content adaptation scheme will ensure that many IPTV clients do not require adaptation of the video to a different coding format, as the H.264/MPEG-4 AVC standard is well supported in industry.

- Three separate precoded versions of the content increases the storage overhead to approximately double that of a scheme using a single-layer encoded version of the content, but it is envisaged to allow the majority of user end-terminal devices to be supported.

- The use of trans-coding requires significant processing requirements for standard and high definition versions of the streams but has acceptable performance for low definition streams, such as those using the mobile domain.

Furthermore the use of three encoded versions of the stream allows adaptation of a stream which is a closer match to the user's requirements and assists in reducing the processing overhead associated with the trans-coding process.

6.3 Recommendations and Future Work

During the course of the work for this thesis a number of avenues for further research emerged.

- Scalable video coding and multiple description coding were not used in the hybrid content adaption scheme due to little support from industry at present. This however may change in the future and if industry were to widely adopt and support a particular SVC or MDC format then the scheme presented in this thesis could benefit from the advantages provided by the SVC and MDC methods. The integration of SVC and/or MDC methods into the hybrid scheme presented in this thesis would give good insight into their benefits.

- The test-bed architecture only investigated clients requiring the adaption of video at the start of the streaming session. Once the content was being streamed the characteristics of the adapted video were kept constant. It is highly desirable however for the video to be dynamically altered for mobile environments where clients are moving and the received signal strength is fluctuating. Further research should investigate the use of the Real-Time control Protocol (RTCP) to determine the current state of the connections between the MDF and user [36]. These RTCP reports can then be used to dynamically alter the video to meet the current QoS level achievable for the client.

- The trans-coding method used in the test-bed was simple and not optimised for performance. An investigation into methods for optimising the trans-coding process, by lowering the processing power and memory requirements, would ensure better performance of the hybrid content adaptation scheme.

Bibliography

[1] T. Magedanz, N. Blum, and S. Dutkowski, "Evolution of SOA Concepts in Telecommunications," *IEEE Computer*, vol. 40, pp. 46–50, November 2007.

[2] "Open IMS Core - homepage," [Online] Available: http://www.openimscore.org/docs/FHoSS/main.html [December, 2008].

[3] B. Shen, W.-T. Tan, and F. Huve, "Dynamic Video Transcoding in Mobile Environments," *IEEE MultiMedia*, vol. 15, pp. 42–51, January 2008.

[4] ETSI TS 182.027, "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN): IPTV Architecture; IPTV functions supported by the IMS subsystem," February 2008.

[5] ETSI TS 183.063, "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN): IMS-based IPTV stage 3 specification," June 2008.

[6] "World Internet Usage Statistics," [Online] Available: http://www.internetworldstats.com/stats.htm [December, 2008].

[7] "About 3GPP homepage," [Online] Available: http://www.3gpp.org/about-3gpp [November, 2008].

[8] G. Camarillo and M. A. Garcia-Martin, *The 3G IP Multimedia Sybsystem (IMS): Merging the Internet and the Cellular Worlds*. John Wiley & Sons, Ltd, 2nd ed., May 2006.

[9] "3GPP - Releases," [Online] Available: http://www.3gpp.org/releases, 2008, [November, 2008].

[10] G.Koleyni. "Internet Protocol Television." *ITU News*. pp. 28–30. October 2008.

[11] "Semiannual IPTV Global Forecast Report of the MRG." [Online] Available: http://www.mrgco.com/TOC_IPTV_GF1008.html. 2008. [November. 2008].

[12] "IPTV Focus Group (FG IPTV)." [Online] Available: http://www.itu.int/ITU-T/IPTV/ [November. 2008].

[13] E. Mikoczy, D. Sivchenko, B. Xu, and J. I. Moreno. "IPTV Services over IMS: Architecture and Standardization," *IEEE Communications Magazine*, vol. 46, pp. 128–135. May 2008.

[14] ETSI TS 102.034. "Transport of MPEG-2 TS-Based DVB Services over IP Based Networks," October 2008.

[15] ETSI TS 102.539. "Digital Video Broadcasting (DVB): Carriage of Broadband Content Guide (BCG) information over Internet Protocol (IP)." April 2008.

[16] ETSI TS 102.824. "Digital video broadcasting (dvb): remote management and firmware update system for dvb ip services," July 2008.

[17] 3GPP TS 23.246. "Multimedia Broadcast/Multicast Service: Architecture and Functional Description." Release 6.

[18] A. Al-Hezmi. M. Knappmeyer, B. Ricks, F. C. Pinto. and R. Tönjes. "Enabling IMS with Multicast and Broadcast Capabilites." *Personal. Indoor and Mobile Radio Communications, 2007. PIMRC 2007. IEEE 18th International Symposium on*. pp. 1–5. September 2007.

[19] "Microsoft Mediaroom." [Online] Available: http://www.microsoftmediaroom.co.uk [December. 2008].

[20] ipTVnews, "Hanarotelecom Launches IPTV Service." [Online] Available: http://www.itu.int/ITU-T/IPTV/. July 2006 [December. 2008].

[21] "BT Vision." [Online] Available: http://www.btvision.bt.com/ [December. 2008].

[22] D. Miras, A. Sadagic, B. Teitelbaum, J. Leigh, M. E. Zarki, and H. Liu, "A Survey of Network QoS Needs of Advanced Internet Applications - Working Document," *Internet 2 QoS Working Group*, 2002.

[23] J. Chakareksi and P. Frossard, "Adaptive Systems for Improved Media Streaming Experience," *IEEE Communications Magazine*, vol. 45, pp. 77–83, January 2007.

[24] H. Schwarz, D. Marpe, and T. Wiegand, "Overview of the Scalable Video Coding Extension of the H.264/AVC Standard," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 17, pp. 1103–1120, September 2007.

[25] M. Wien, H. Schwarz, and T. Oelbaum, "Performance Analysis of SVC," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 17, pp. 1194–1203, September 2007.

[26] J. Kim, T.-W. Um, W. Ryu, B. S. Lee, and M. Hahn, "Heterogeneous Networks and Terminal-Aware QoS/QoE-Guaranteed Mobile IPTV Service," *IEEE Communications Magazine*, vol. 46, pp. 110–117, May 2008.

[27] M.-T. Lu, J.-C. Wu, K.-J. Peng, P. Huang, J. Yao, and H. Chen, "Design and Evaluation of a P2P IPTV System for Heterogeneous Networks," *IEEE Transactions on Multimedia*, vol. 9, pp. 1568–1579, December 2007.

[28] A. Vetro, C. Christopoulos, and H. Sun, "Video Transcoding Architectures and Techniques: An Overview," *IEEE Signal Processing Magazine*, vol. 20, pp. 18–29, March 2003.

[29] IETF RFC 2327, "SDP: Session Description Protocol," April 1998.

[30] "VideoLan - homepage," [Online] Available: http://www.videolan.org/ [December, 2008].

[31] VideoLan, "VLC features list," [Online] Available: http://www.videolan.org/vlc/features.html [December, 2008].

[32] "The eXtended osip library," [Online] Available: http://savannah.nongnu.org/projects/exosip/ [December, 2008].

[33] "The GNU oSIP library - homepage." [Online] Available: http://www.gnu.org/software/osip/ [December, 2008].

[34] "AviDemux - homepage." [Online] Available: http://www.avidemux.org/ [December, 2008].

[35] D. Wieërs, "PDstat: Versatile resource statistics tool - homepage." [Online] Available: http://dag.wieers.com/home-made/dstat/ [December, 2008].

[36] A. P. Pavlos Antoniou and V. Vassiliou, "Adaptive feedback algorithm for Internet video streaming based on fuzzy rate control," *Computers and Communications 2007. ISCC'07. 12th IEEE Symposium on.*, pp. 219–226, July 2007.

[37] "Multimedia Data - What is compression?," [Online] Available: http://www.cs.cf.ac.uk/Dave/Multimedia/node202.html#SECTION04211000000000000000, 2001. [November, 2008].

[38] P. Symes, *Digital Video Compression*. United States: McGraw-Hill/TAB Electronics, 1st ed., October 2003.

[39] Y. Q. Shi and H. Sun, *Image and Video Compression for Multimedia Engineering: Fundamentals, Algorithms, and Standards*. 2000 N.W Corporate Blvd., Boca Ranton, Florida 33431: CRC Press LLC, 2nd ed., March 2008.

Appendix A

Digital video compression and video characterstics

A.1 Brief overview of digital video compression

Raw video data requires large amounts of bandwidth in order to be transmitted. Consider a grey-scale image with a height and width of ten pixels. Each pixel is assigned an eight bit value, indicating the shade of the pixel. This image represents just one frame of a video sequence and requires eight hundred bits to be transmitted between the sender and receiver. Video frame rates are typically in the region of thirty frames per second which means that for one second of video the communications channel capacity needs to support a video bit-rate equal to at least 24 kbits/s. Scaling this up, an image with 800 by 600 pixels, with 24 bits represent the colour of each pixel, our video information rate is now 345.6 Mbits/s. This video bit-rate alone is greater than the transmission rates of current communication links; yet present day High Definition television supports resolutions of up to 1920 by 1080 pixels per frame. These high resolution video transmissions are possible due to modern day video compression methods which reduce the number of bits required to be transmitted while still providing video with an acceptable quality.

Video compression can be divided into two categories: loss-less compression and lossy compression. Loss-less compression, as the name implies, allows the video data to be compressed and reconstructed without loss of any information. Loss-less compression methods typically involve entropy encoding and are based on information theoretic techniques, such as repetitive sequence suppression and statistical encoding. Lossy compression methods on the other hand do not allow for the reconstruction of the original signal and are aimed at achieving the best fidelity for a given bit-rate, or at minimising the bit-rate in order to achieve a given fidelity measure [37]. Lossy compression methods exploit the fact that the human eye is largely insensitive to the removal of certain parts of a video signal, as well as the redundancy in successive frames of a video sequence (e.g. during a scene with two people talking, apart from the people's facial and body movements, little information is likely to change from frame to frame). Modern video transmission methods compromise by using lossy compression methods in order to reduce the bit-rate of high resolution video but exploit the latest advances in compression methods to allow for video to be reproduced at an acceptable quality [38, 39]. In practice both loss-less and lossy compression methods are used fro the reduction of video bit-rates.

A number of video compression standards have been developed by organizations such as the ISO/IEC's Moving Picture Experts Group (MPEG) and the ITU-T's Video Coding Experts Group (VCEG). Each new standard has emerged to take advantage of improvements in compression and video broadcasting technology. In addition to this, large corporations have developed proprietary video coding formats for use in their software and systems, some of which have later been adopted by other manufacturers and standardizing bodies: Microsoft's WMV and VC-1 coding formats are examples of such formats.

A.2 Factors influencing the adaptation of video content

Section A.1 introduced the concept of compression technology which reduces video bit-rates and allows transmission of video over modern day communication links. This section highlights some of the key aspects of encoded video, which may make it necessary for the adaption of video content to suit different user devices and network characteristics.

A.2.1 Coding Format

As mentioned in the previous section, there are a number of video coding formats available and although end-terminals usually support multiple formats it may be the case that the streamed video format is encoded using a format that is not supported by the user's device (usually related to licensing of the coding format). In such a scenario, adapting the video content to a format supported by the device is necessary to ensure playback of the video to the user. Furthermore, coding standards often have a number of profiles or levels, targeted at different application domains, to which various video streams may conform. Not all profiles and levels are supported by devices that support a particular coding standard. As such the adaptation of video from one profile to another may be required. As an example, the recent H.264/MPEG-4 AVC coding standard specifies, amongst others, a Baseline profile (targeted at low-cost applications with limited computing resources, such as in mobile environments) and High profile (targeted at mainstream broadcast and disc storage applications). A user may have a smart-phone only capable of supporting the Baseline profile; hence content encoded with the High profile will need to be adapted in order to cater for this particular user.

A.2.2 Display Resolution

Due to the popularity of Digital Video Disks (DVDs) and high-definition Blu-ray disks, along with the increased proliferation of broadband networks which allow high bit-rate video to be transmitted, more and more high resolution video is being captured. A the same time, the emergence of mobile, multimedia-capable devices has created a need for the reduction of such high-quality content in order to suit the smaller display sizes associated with these devices.

A.2.3 Temporal Resolution

The temporal resolution of the video refers to the frame-rate of the video and is the number of video frames displayed per second. A higher temporal resolution means more frames need to be transmitted in the same amount of time, hence requiring a higher transmission rate. A reduction in the temporal resolution can translate to a significant decrease in the bit-rate of the video or alternatively for an improvement in the quality of the individual frames. Some devices, particularly those in the mobile world, may only be capable of decoding and displaying a certain number of frames per second and hence require the video to be at a temporal resolution equal to or lower than this amount. Also mobile devices have limited battery life and may wish to conserve power by reducing the processing requirements of the video at the cost of receiving a lower frame rate.

A.2.4 Fidelity

The quality of compressed video is mainly determined by the quantisation step in the encoding process. A higher number of quantum values used to compress a range of pixel values, the higher number of discrete symbols required to transmit the video. This impacts on the video as the higher the number of symbols that need to be transmitted, the higher the bit-rate of the video. Typically both the colour and frequency components of a video frame are quantised. Colour quantisation limits the number of colours that make up a frame to a discrete set (referred to as a colour palette) while frequency quantisation exploits the fact that the human eye is not very good at distinguishing the exact strength of a high frequency brightness variation. By reducing either the number of colours or the frequency components that make up an image, the bit-rate of the video can be reduced in order to adapt it to various communication channel transmission rates.

Appendix B

Details of machines used in the test-bed

This appendix gives the details for each machine used in the test-bed architecture. The information was obtained using the following commands in ubuntu:

1. #head /proc/cpuinfo
2. #head /proc/meminfo
3. #lsb_release -a
4. #uname -a

OpenIMS core machine

- vendor_id: GenuineIntel
- cpu_family: 15
- model: 2
- model name: Intel(R) Pentium(R) 4 CPU 3.00GHz
- stepping: 9
- cpu MHz: 2992.688
- cache size: 512 KB

- physical id: 0
- siblings: 2
- MemTotal: 1016508 kB
- OS: Ubuntu - 8.10 - Intrepid
- OS Kernel: Linux 2.6.27-9-generic

SCF machine

- vendor_id: GenuineIntel
- cpu family: 6
- model: 15
- model name: Intel(R) Pentium(R) Dual CPU E2160 @ 1.8 GHz
- stepping: 13
- cpu MHz: 1803.000
- cache size: 1024 KB
- physical id: 0
- siblings: 2
- MemTotal: 1025124 kB
- OS: Ubuntu - 8.10 - Intrepid
- OS Kernel: Linux 2.6.27-9-generic

Hybrid MCF/MDF Machine

- vendor_id: GenuineIntel
- cpu family: 6
- model: 23
- model name : Intel(R) Core(TM)2 Quad CPU Q9300 @ 2.50GHzs
- stepping: 7
- cpu MHz: 2003.000
- cache size : 3072 KB
- physical id: 0
- siblings : 4
- cpu cores: 4
- MemTotal: 3111724 kB
- OS: Ubuntu - 8.10 - Intrepid
- OS Kernel: Linux 2.6.27-7-generic

Client Terminals

Alice:

- vendor_id: GenuineIntel
- cpu family: 15
- model: 3
- model name: Intel(R) Pentium(R) 4 CPU 3.20GHz
- stepping: 4 cpu
- MHz: 3191.938

- cache size: 1024 KB
- physical id: 0
- siblings: 2
- MemTotal: 1032748 kB
- OS: Ubuntu - 8.10 - Intrepid
- OS Kernel: Linux 2.6.27-9-generic

Allan, Bob and Justin:

- vendor_id: GenuineIntel
- cpu family: 6
- model: 15
- model name : Intel(R) Pentium(R) Dual CPU E2160 @ 1.80GHz
- stepping: 13 cpu MHz: 1803.000
- cache size : 1024 KB
- physical id: 0
- siblings : 2
- MemTotal: 1025124 kB
- OS: Ubuntu - 8.10 - Intrepid
- OS Kernel: Linux 2.6.27-7-generic

Chapter 6

Conclusions

6.1 Summary

The convergence of IP access networks in the IMS network architecture and the diverse range of IP-multimedia enabled end-terminal devices make it a requirement for IPTV-IMS services to be able to adapt the multimedia content to suit individual client requirements. This Thesis has involved an investigation into methods that could be used to adapt video content to cater for different access network bandwidths and client end-terminal device characteristics.

Precoding. scalable coding, multiple description coding and trans-coding were the methods investigated in this thesis to perform the adaptation of video content, with each of the methods outlined and discussed. The key issues of video content adaptation were identified as:

1. Comprehensive adaptation of video in terms of format. spatial resolution, temporal resolution and quality.

2. Optimal use of server resources such as processing power and storage space.

3. Able to be integrated with current IMS specifications.

Each of the methods investigated was evaluated in terms of these three criteria but it was found that no single method was truly suitable for the task.

This thesis then proposed and outlined a hybrid adaptation scheme that makes use of both the precoding and trans-coding methods. in order to leverage the

Appendix C

Information related to Test Sequences

C.1 Animation - Action Sequences

C.1.1 casshern_sins_ep2_action_h264_hd.avi

Format : AVI
Format/Info : Audio Video Interleave
File size : 7.23 MiB
Duration : 1mn 0s
Overall bit rate : 1 005 Kbps
Video
Format : AVC
Format/Info : Advanced Video Codec
Format profile : High@L5.1
Format settings, CABAC : Yes
Format settings, ReFrames : 2 frames
Codec ID : H264
Duration : 1mn 0s
Bit rate : 864 Kbps
Width : 1 280 pixels
Height : 720 pixels

Display aspect ratio : 16/9
Frame rate : 23.976 fps
Resolution : 24 bits
Colorimetry : 4:2:0
Scan type : Progressive
Bits/(Pixel*Frame) : 0.039
Stream size : 6.21 MiB (86%)
Writing library : x264 core 59
Encoding settings : cabac=1 / ref=1 / deblock=1:0:0 / analyse=0x3:0x113 / me=hex / subme=5 / brdo=0 / mixed_ref=0 / me_range=16 / chroma_me=1 / trellis=0 / 8x8dct=1 / cqm=0 / deadzone=21.11 / chroma_qp_offset=0 / threads=6 / nr=0 / decimate=1 / mbaff=0 / bframes=2 / b_pyramid=0 / b_adapt=1 / b_bias=0 / direct=1 / wpredb=1 / bime=0 / keyint=250 / keyint_min=25 / sce-necut=40(pre) / rc=crf / crf=26.0 / rceq='blurCplx^(1-qComp)' / qcomp=1.00 / qpmin=10 / qpmax=51 / qpstep=4 / ip_ratio=1.40 / pb_ratio=1.30 / aq=2:1.00
Audio
Format : MPEG Audio
Format version : Version 1
Format profile : Layer 3
Codec ID : 55
Codec ID/Hint : MP3
Bit rate mode : Constant
Bit rate : 128 Kbps
Channel(s) : 2 channels
Sampling rate : 48.0 KHz
Resolution : 16 bits
Stream size : 942 KiB (13%)
Alignment : Split accross interleaves
Interleave, duration : 24 ms (0.58 video frame)

C.1.2 casshern_sins_ep2_action_h264_sd.avi

Format : AVI
Format/Info : Audio Video Interleave

of IPTV services. The use of H.264 encoded video should ensure most client's requiring HD and SD content will not require trans-coding to a different encoding format, since many of the devices that the HD and SD versions target are likely to support the H.264 standard. It may be necessary however to have both wide-screen and standard versions of the content to ensure that trans-coding of the HD and SD versions between wide-screen and standard aspect ratios is not required. This will of course lead to an increase in the storage space required by the scheme.

The mobile domain is the only real area where the adaptation of HD and SD sequences may be truly needed. Although many devices in the mobile domain are sequences may be truly needed. Although many devices in the mobile domain are mobile phones, mobile access technology has started to become very popular with notebook and laptop mobile computer users. It may be the case that a particular user may access the IPTV service from high-end mobile device with a large display and thus requiring the higher definition versions o f the content. The mobility of such a user through areas with varying signal quality could cause large fluctuations in the link bandwidth and would require dynamic alternations in the video. In such a case it will be better to switch between precoded versions of the content instead of trans-coding the HD and SD versions so as to ensure scalability of IPTV services.

5.7 Chapter Summary

This chapter began with a look at the tools and video sequences used for testing the hybrid content adaptation scheme in the IMS-IPTV test-bed. This was followed by a discussion on the file storage requirements of the precoded content and a look at trans-coding of content between H.264 and the older MPEG-4 ASP formats. After which the trans-coding between different precoded versions was analysed and discussed before a further analysis of trans-coding HD and SD content was given.

An analysis of the file sizes of the three precoded versions of the content showed that the use of precoding in the hybrid content adaptation scheme lead to almost double the amount of storage space being required, when compared with schemes that only use a single HD encoded video stream. The increase in required storage space is however offset by the performance gains achieved when trans-coding the lower definition versions of the streams and the results from the trans-coding of each precoded version of the test sequences showed this.

The use of H.264 was justified by showing that although it required more processing power to trans-code to H.264, users would prefer H.264 content as it provides better video quality than previous coding standards such as MPEG-4 ASP. In addition to this the widespread adoption of the standard by industry means it is highly likely that a video capable user end-terminal will support the standard.

The trans-coding of HD and SD versions of the test sequences requires significant processing power and it was shown that with just four clients the high-end MDF machine was at full CPU load for the HD versions, and near full load for the SD versions. It is however envisaged that many devices will support the H.264 standard and thus the number of clients requiring trans-coding for the HD and SD versions to a different encoding format will be minimal. It was suggested that wide-screen and standard versions of the content be kept in order to ensure trans-coding of content between different aspect ratios is not required and that switching between the different versions of the content be used to adapt HD and SD versions of the video in mobile environments.

File size : 3.98 MiB
Duration : 1mn 0s
Overall bit rate : 554 Kbps
Video
Format : AVC
Format/Info : Advanced Video Codec
Format profile : High@L5.1
Format settings, CABAC : Yes
Format settings, ReFrames : 2 frames
Codec ID : H264
Duration : 1mn 0s
Bit rate : 412 Kbps
Width : 800 pixels
Height : 480 pixels
Display aspect ratio : 1.667
Frame rate : 23.976 fps
Resolution : 24 bits
Colorimetry : 4:2:0
Scan type : Progressive
Bits/(Pixel*Frame) : 0.045
Stream size : 2.96 MiB (74%)
Writing library : x264 core 59
Encoding settings : cabac=1 / ref=1 / deblock=1:0:0 / analyse=0x3:0x113 / me=hex / subme=5 / brdo=0 / mixed_ref=0 / me_range=16 / chroma_me=1 / trellis=0 / 8x8dct=1 / cqm=0 / deadzone=21,11 / chroma_qp_offset=0 / threads=6 / nr=0 / decimate=1 / mbaff=0 / bframes=2 / b_pyramid=0 / b_adapt=1 / b_bias=0 / direct=1 / wpredb=1 / bime=0 / keyint=250 / keyint_min=25 / scenecut=40(pre) / rc=crf / crf=26.0 / rceq='blurCplx^(1-qComp)' / qcomp=1.00 / qpmin=10 / qpmax=51 / qpstep=4 / ip_ratio=1.40 / pb_ratio=1.30 / aq=2:1.00
Audio
Format : MPEG Audio
Format version : Version 1
Format profile : Layer 3

Codec ID : 55
Codec ID/Hint : MP3
Bit rate mode : Constant
Bit rate : 128 Kbps
Channel(s) : 2 channels
Sampling rate : 48.0 KHz
Resolution : 16 bits
Stream size : 942 KiB (23%)
Alignment : Split accross interleaves
Interleave, duration : 24 ms (0.58 video frame)

C.1.3 casshern_sins_ep2_action_h264_ld.avi

Format : AVI
Format/Info : Audio Video Interleave
File size : 1.92 MiB
Duration : 1mn 0s
Overall bit rate : 267 Kbps
Video
Format : AVC
Format/Info : Advanced Video Codec
Format profile : High@L5.1
Format settings, CABAC : Yes
Format settings, ReFrames : 2 frames
Codec ID : H264
Duration : 1mn 0s
Bit rate : 129 Kbps
Width : 320 pixels
Height : 240 pixels
Display aspect ratio : 4/3
Frame rate : 23.976 fps
Resolution : 24 bits
Colorimetry : 4:2:0
Scan type : Progressive

Bits/(Pixel*Frame) : 0.070
Stream size : 953 KiB (48%)
Writing library : x264 core 59
Encoding settings : cabac=1 / ref=1 / deblock=1:0:0 / analyse=0x3:0x113 / me=hex / subme=5 / brdo=0 / mixed_ref=0 / me_range=16 / chroma_me=1 / trellis=0 / 8x8dct=1 / cqm=0 / deadzone=21.11 / chroma_qp_offset=0 / threads=6 / nr=0 / decimate=1 / mbaff=0 / bframes=2 / b_pyramid=0 / b_adapt=1 / b_bias=0 / direct=1 / wpredb=1 / bime=0 / keyint=250 / keyint_min=25 / scenecut=40(pre) / rc=crf / crf=26.0 / rceq='blurCplx^(1-qComp)' / qcomp=1.00 / qpmin=10 / qpmax=51 / qpstep=4 / ip_ratio=1.40 / pb_ratio=1.30 / aq=2:1.00
Audio
Format : MPEG Audio
Format version : Version 1
Format profile : Layer 3
Codec ID : 55
Codec ID/Hint : MP3
Duration : 1mn 0s
Bit rate mode : Constant
Bit rate : 128 Kbps
Channel(s) : 2 channels
Sampling rate : 48.0 KHz
Resolution : 16 bits
Stream size : 938 KiB (48%)
Alignment : Split accross interleaves
Interleave, duration : 42 ms (1.01 video frame)
General

C.2 Die Hard - Action Sequences

C.2.1 die_hard_action_h264_hd.avi

Format : AVI
File size : 16.7 MiB

Duration : 59s 393ms

Overall bit rate : 2 358 Kbps

Video

Format : AVC

Format profile : High@L5.1

Format settings, CABAC : Yes

Format settings, ReFrames : 2 frames

Codec ID : H264

Duration : 59s 393ms

Bit rate : 2 216 Kbps

Width : 1 280 pixels

Height : 544 pixels

Display aspect ratio : 2.35

Frame rate : 23.976 fps

Resolution : 24 bits

Colorimetry : 4:2:0

Scan type : Progressive

Bits/(Pixel*Frame) : 0.133

Stream size : 15.7 MiB (94%)

Writing library : x264 core 59

Encoding settings : cabac=1 / ref=1 / deblock=1:0:0 / analyse=0x3:0x113 / me=hex / subme=5 / brdo=0 / mixed_ref=0 / me_range=16 / chroma_me=1 / trellis=0 / 8x8dct=1 / cqm=0 / deadzone=21.11 / chroma_qp_offset=0 / threads=6 / nr=0 / decimate=1 / mbaff=0 / bframes=2 / b_pyramid=0 / b_adapt=1 / b_bias=0 / direct=1 / wpredb=1 / bime=0 / keyint=250 / keyint_min=25 / scenecut=40(pre) / rc=crf / crf=26.0 / rceq='blurCplx^(1-qComp)' / qcomp=1.00 / qpmin=10 / qpmax=51 / qpstep=4 / ip_ratio=1.40 / pb_ratio=1.30 / aq=2:1.00

Audio

Format : MPEG Audio

Format profile : Layer 3

Codec ID : 55

Codec ID/Hint : MP3

Bit rate mode : Constant

Bit rate : 128 Kbps
Channel(s) : 2 channels
Sampling rate : 48.0 KHz
Resolution : 16 bits
Stream size : 928 KiB (5%)
Alignment : Split accross interleaves
Interleave, duration : 24 ms (0.58 video frame)

C.2.2 die_hard_action_h264_sd.avi

Format : AVI
Format/Info : Audio Video Interleave
File size : 9.82 MiB
Duration : 59s 393ms
Overall bit rate : 1 387 Kbps
Video
Format : AVC
Format/Info : Advanced Video Codec
Format profile : High@L5.1
Format settings, CABAC : Yes
Format settings, ReFrames : 2 frames
Codec ID : H264
Duration : 59s 393ms
Bit rate : 1 245 Kbps
Width : 800 pixels
Height : 480 pixels
Display aspect ratio : 1.667
Frame rate : 23.976 fps
Resolution : 24 bits
Colorimetry : 4:2:0
Scan type : Progressive
Bits/(Pixel*Frame) : 0.135
Stream size : 8.82 MiB (90%)
Writing library : x264 core 59

Encoding settings : cabac=1 / ref=1 / deblock=1:0:0 / analyse=0x3:0x113 / me=hex / subme=5 / brdo=0 / mixed_ref=0 / me_range=16 / chroma_me=1 / trellis=0 / 8x8dct=1 / cqm=0 / deadzone=21.11 / chroma_qp_offset=0 / threads=6 / nr=0 / decimate=1 / mbaff=0 / bframes=2 / b_pyramid=0 / b_adapt=1 / b_bias=0 / direct=1 / wpredb=1 / bime=0 / keyint=250 / keyint_min=25 / scenecut=40(pre) / rc=crf / crf=26.0 / rceq='blurCplx^(1-qComp)' / qcomp=1.00 / qpmin=10 / qpmax=51 / qpstep=4 / ip_ratio=1.40 / pb_ratio=1.30 / aq=2:1.00
Audio
Format : MPEG Audio
Format version : Version 1
Format profile : Layer 3
Codec ID : 55
Codec ID/Hint : MP3
Bit rate mode : Constant
Bit rate : 128 Kbps
Channel(s) : 2 channels
Sampling rate : 48.0 KHz
Resolution : 16 bits
Stream size : 928 KiB (9%)
Alignment : Split accross interleaves
Interleave, duration : 24 ms (0.58 video frame)

C.2.3 die_hard_action_h264_ld.avi

Format : AVI
Format/Info : Audio Video Interleave
File size : 3.30 MiB
Overall bit rate : 465 Kbps
Video
Format : AVC
Format/Info : Advanced Video Codec
Format profile : High@L5.1
Format settings, CABAC : Yes
Format settings, ReFrames : 2 frames

Codec ID : H264
Duration : 59s 393ms
Bit rate : 324 Kbps
Width : 320 pixels
Height : 240 pixels
Display aspect ratio : 4/3
Frame rate : 23.976 fps
Resolution : 24 bits
Colorimetry : 4:2:0
Scan type : Progressive
Bits/(Pixel*Frame) : 0.176
Stream size : 2.30 MiB (70%)
Writing library : x264 core 59
Encoding settings : cabac=1 / ref=1 / deblock=1:0:0 / analyse=0x3:0x113 / me=hex / subme=5 / brdo=0 / mixed_ref=0 / me_range=16 / chroma_me=1 / trellis=0 / 8x8dct=1 / cqm=0 / deadzone=21.11 / chroma_qp_offset=0 / threads=6 / nr=0 / decimate=1 / mbaff=0 / bframes=2 / b_pyramid=0 / b_adapt=1 / b_bias=0 / direct=1 / wpredb=1 / bime=0 / keyint=250 / keyint_min=25 / scenecut=40(pre) / rc=crf / crf=26.0 / rceq='blurCplx^(1-qComp)' / qcomp=1.00 / qpmin=10 / qpmax=51 / qpstep=4 / ip_ratio=1.40 / pb_ratio=1.30 / aq=2:1.00
Audio
Format : MPEG Audio
Format version : Version 1
Format profile : Layer 3
Codec ID : 55
Codec ID/Hint : MP3
Bit rate mode : Constant
Bit rate : 128 Kbps
Channel(s) : 2 channels
Sampling rate : 48.0 KHz
Resolution : 16 bits
Stream size : 922 KiB (27%)
Alignment : Split accross interleaves

Interleave. duration : 24 ms (0.58 video frame)

C.3 Gladiator - Dialogue Sequences

C.3.1 Complete name : gladiator_h264_hd.avi

Format : AVI

Format/Info : Audio Video Interleave

File size : 3.46 MiB

Duration : 1mn 0s

Overall bit rate : 479 Kbps

Video

Format : AVC

Format/Info : Advanced Video Codec

Format profile : High@L5.1

Format settings, CABAC : Yes

Format settings, ReFrames : 2 frames

Codec ID : H264

Duration : 1mn 0s

Bit rate : 337 Kbps

Width : 1 280 pixels

Height : 720 pixels

Display aspect ratio : 16/9

Frame rate : 23.976 fps

Resolution : 24 bits

Colorimetry : 4:2:0

Scan type : Progressive

Bits/(Pixel*Frame) : 0.015

Stream size : 2.43 MiB (70%)

Encoding settings : cabac=1 / ref=1 / deblock=1:0:0 / analyse=0x3:0x113 / me=hex / subme=5 / brdo=0 / mixed_ref=0 / me_range=16 / chroma_me=1 / trellis=0 / 8x8dct=1 / cqm=0 / deadzone=21.11 / chroma_qp_offset=0 / threads=6 / nr=0 / decimate=1 / mbaff=0 / bframes=2 / b_pyramid=0 / b_adapt=1

/ b_bias=0 / direct=1 / wpredb=1 / bime=0 / keyint=250 / keyint_min=25 / scenecut=40(pre) / rc=crf / crf=26.0 / rceq='blurCplx^(1-qComp)' / qcomp=1.00 / qpmin=10 / qpmax=51 / qpstep=4 / ip_ratio=1.40 / pb_ratio=1.30 / aq=2:1.00
Audio
Format : MPEG Audio
Format version : Version 1
Format profile : Layer 3
Codec ID : 55
Codec ID/Hint : MP3
Bit rate mode : Constant
Bit rate : 128 Kbps
Channel(s) : 2 channels
Resolution : 16 bits
Stream size : 945 KiB (27%)
Alignment : Split accross interleaves
Interleave, duration : 24 ms (0.58 video frame)

C.3.2 gladiator_h264_sd.avi

Format : AVI
Format/Info : Audio Video Interleave
File size : 1.94 MiB
Duration : 1mn 0s
Overall bit rate : 269 Kbps
Video
Format : AVC
Format/Info : Advanced Video Codec
Format settings, CABAC : Yes
Codec ID : H264
Bit rate : 127 Kbps
Height : 480 pixels
Resolution : 24 bits
Colorimetry : 4:2:0
Scan type : Progressive

Bits/(Pixel*Frame) : 0.014
Stream size : 936 KiB (47%)
Writing library : x264 core 59
Encoding settings : cabac=1 / ref=1 / deblock=1:0:0 / analyse=0x3:0x113 / me=hex / subme=5 / brdo=0 / mixed_ref=0 / me_range=16 / chroma_me=1 / trellis=0 / 8x8dct=1 / cqm=0 / deadzone=21,11 / chroma_qp_offset=0 / threads=6 / nr=0 / decimate=1 / mbaff=0 / bframes=2 / b_pyramid=0 / b_adapt=1 / b_bias=0 / direct=1 / wpredb=1 / bime=0 / keyint=250 / keyint_min=25 / scenecut=40(pre) / rc=crf / crf=26.0 / rceq='blurCplx^(1-qComp)' / qcomp=1.00 / qpmin=10 / qpmax=51 / qpstep=4 / ip_ratio=1.40 / pb_ratio=1.30 / aq=2:1.00
Audio
Format : MPEG Audio
Format version : Version 1
Codec ID/Hint : MP3
Bit rate mode : Constant
Channel(s) : 2 channels
Sampling rate : 48.0 KHz
Resolution : 16 bits
Stream size : 945 KiB (48%)
Alignment : Split accross interleaves
Interleave, duration : 24 ms (0.58 video frame)

C.3.3 gladiator_h264_ld.avi

Format : AVI
Format/Info : Audio Video Interleave
File size : 1.23 MiB
Duration : 1mn 0s
Overall bit rate : 171 Kbps
Video
Format : AVC
Format/Info : Advanced Video Codec
Format profile : High@L5.1
Format settings, CABAC : Yes

Format settings, ReFrames : 2 frames
Codec ID : H264
Duration : 1mn 0s
Bit rate : 28.9 Kbps
Width : 320 pixels
Frame rate : 23.976 fps
Resolution : 24 bits
Colorimetry : 4:2:0
Scan type : Progressive
Bits/(Pixel*Frame) : 0.016
Stream size : 214 KiB (17%)
Writing library : x264 core 59

/ me=hex / subme=5 / brdo=0 / mixed_ref=0 / me_range=16 / chroma_me=1 / trellis=0 / 8x8dct=1 / cqm=0 / deadzone=21,11 / chroma_qp_offset=0 / threads=6 / nr=0 / decimate=1 / mbaff=0 / bframes=2 / b_pyramid=0 / b_adapt=1 / b_bias=0 / direct=1 / wpredb=1 / bime=0 / keyint=250 / keyint_min=25 / scenecut=40(pre) / rc=crf / crf=26.0 / rceq='blurCplx^(1-qComp)' / qcomp=1.00 / qpmin=10 / qpmax=51 / qpstep=4 / ip_ratio=1.40 / pb_ratio=1.30 / aq=2:1.00

Audio
Format : MPEG Audio
Format version : Version 1
Codec ID : 55
Codec ID/Hint : MP3
Bit rate mode : Constant
Bit rate : 128 Kbps
Channel(s) : 2 channels
Resolution : 16 bits
Stream size : 945 KiB (75%)
Alignment : Split accross interleaves
Interleave, duration : 24 ms (0.58 video frame)

Appendix D

Accompanying CD-ROM

The CD-ROM included with this thesis contains the following files and information:

- *Research Literature* - Electronic copies of the research papers and other literature used during the course of this research can be found in the directory labelled "Research Literature".
- *Software* - All the source code developed for the evaluation framework can be found in the directory labelled "Software".
- *Test Sequence Video Files* - The test sequences used for the testing section of this thesis can be found in the directory labelled "Test Sequence".
- *Thesis* - An electronic copy, in PDF format, of this document can be found in the directory labelled "Thesis".