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SIP Based IPTV Architecture for Heterogeneous Networks

Server Architecture

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Abstract

This dissertation presents an IP Television (IPTV) service architecture that applies the Session Initiation Protocol (SIP) for session and media control, while incorporating a design suitable for deployment in the context of an IP Multimedia Subsystem (IMS) architectural framework.
The main features of the architecture include flexible delivery of personalized multimedia streams, adaptability to the characteristics of the networking infrastructures and channels used for transmission, and a modular design to facilitate implementation of new functionalities and services.
In order to maximize the end users Quality Of Experience (QoE) independently of the access medium used, a Quality Of Service (QoS) adaptation method is proposed allowing dynamic realtime updates of session attributes. The developed solution is specifically designed for live multimedia streaming, such as broadcasted events, independently of the cast mode (unicast or multicast). Private Video Recorder (PVR) functions and Video On Demand (VoD) services are supported, their control is assured by standard SIP messages.
This dissertation is focused on the development of an IPTV Application Server (AS), that interacts with an IPTV Client developed under the same research project. The functionalities and scalability of the AS were tested on a live wireless 3G Code Division Multiple Access 2000 (CDMA2000) network and on a campus high speed network.

Keywords

Internet Protocol Television, Session Initiation Protocol, Next Generation Networks, IP Multimedia Subsystem, Streaming, Multimedia Multicast

Resumo

Esta dissertação apresenta uma arquitectura para serviços IPTV que utiliza o *Session Initiation Protocol* (SIP) para estabelecimento e controlo do serviço e de fluxos multimédia. A arquitectura tem como objectivo ser implementado num contexto de rede de próxima geração, através da arquitectura IMS.

As principais funcionalidades incluem mecanismos de entrega flexível de conteúdos de multimédia personalizados, um método de adaptar a transmissão multimédia às características do canal e infra-estruturas de rede e uma definição de arquitectura modular que facilita a implementação de novas funcionalidades e serviços.

Propõe-se um novo mecanismo de adaptação e monitorização da Qualidade de Serviço QoS, que permite actualizações da parametrização multimédia em tempo real, que tem como objectivo maximizar a Qualidade de Experiência QoE. A solução desenvolvida é especificamente focada em ambientes de transmissão multimédia em directo, tais como serviços de difusão em directo, para serviços em *unicast* e *multicast*. Funcionalidades de gravação vídeo privado PVR e de conteúdos *on-demand* VoD são também suportados, sendo que o seu controlo é assegurado pelo protocolo SIP.

Este trabalho é focado no desenvolvimento da arquitectura de um Servidor Aplicacional IPTV, que interage com um Cliente IPTV desenvolvido no contexto do mesmo projecto de investigação. As funcionalidades e escalabilidade do Servidor Aplicacional são também testados, recorrendo à utilização de uma rede móvel de 3ª geração CDMA2000 e numa rede universitária de alto débito.

Palavras Chave

Internet Protocol Television, Session Initiation Protocol, Redes de Próxima Geração, Subsistema Multimédia IP, Streaming, Multimedia Multicast

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List of Acronyms

3GPP 3rd Generation Partnership Project

3GPP2 3rd Generation Partnership Project 2

AAA Authentication, Authorization and Accounting

AAC Advanced Audio Coding

A-BGF Core Border Gateway Function

AMR Adaptive Multi-Rate Compression

API Application Programming Interface

AS Application Server

AVC Advanced Video Codec

BER Bit Error Rate

CABAC Context-adaptive binary arithmetic coding

CAVLC Context-adaptive variable-length coding

CDMA2000 Code Division Multiple Access 2000

CGF Charging Gateway Function

CNAME Canonical Name

CPU Central Processing Unit

CSCF Call Session Control Function

DCT Discrete Cosine Transform

DHCP Dynamic Host Configuration Protocol

DRM Digital Rights Management

DSL Digital subscriber line

DSS Darwin Streaming Server

DVB Digital Video Broadcasting

DVD Digital Video Disc

EPG Electronic Programming Guide

ES Elementary stream

ETSI European Telecommunications Standards Institute

FP7 Seventh Framework Programme

GOP Group Of Pictures

GUI Graphical User Interface

HD High Definition

HSS Home Subscriber Server

HTTP Hypertext Transfer Protocol

I-BCF Interconnect Border Control Function

I-BGF Interconnect Border Gateway Function

I-CSCF Interrogating Call Session Control Function

IETF Internet Engineering Task Force

IGMP Internet Group Management Protocol

IM Instant Messaging

IMS IP Multimedia Subsystem

IP Internet Protocol

IPsec IP Security

IPTV IP Television

IPv4 Internet Protocol Version 4

IPv6 Internet Protocol Version 6

ISO International Organization for Standardization

ISP Internet Service Provider

ITU International Telecommunication Union

ITU-T ITU Telecommunication Standardization Sector

JVT Joint Video Team

LAN Local Area Network

MCF Media Control Function

MDF Media Delivery Function

MIME Multipurpose Internet Mail Extensions

MoD Music On Demand

MP3 MPEG-1 Audio Layer 3

MPEG Moving Picture Experts Group

MPEG-1 first group of MPEG codecs

MPEG-2 second group of MPEG codecs

MPEG-4 fourth group of MPEG codecs

NAPT Network Address and Port Translation

NASS Network Attachment Subsystem

NGN Next Generation Network

P2P Peer-To-Peer

P-CSCF Proxy Call Session Control Function

PDF Policy Decision Function

PES Packetized Elementary Stream

PHB Per Hop Behavior

PIM Protocol Independent Multicast

POSIX Portable Operating System Interface for Unix

POTS Plain Old Telephone Service

PS Program Stream

PSNR Peak Signal-to-Noise Ratio

PTT Push-to-Talk

PVR Private Video Recorder

QoE Quality Of Experience

QoS Quality Of Service

QTSS QuickTime Streaming Server

RACF Resource Access Control Function

RACS Resource and Admission Control Subsystem

RDP Real Networks Transport Protocol

RGB Red Green Blue

RLP Radio Link Protocol

RSVP Resource Reservation Protocol

RTCP RTP Control Protocol

RTP Realtime Transport Protocol

RTSP Realtime Streaming Protocol

RTT Round Trip Time

S/MIME Secure / Multipurpose Internet Mail Extensions

SCF Service Control Functions

S-CSCF Serving Call Session Control Function

SD Standard Definition

SDF Service Discovery Functions

SDP Session Description Protocol

SI Switching Slice Intra Frame

SIP Session Initiation Protocol

SLA System Load Average

SP Switching Slice Predicted Frame

SSF Service Selection Function

SVC Scalable Video Coding

TCP Transport Control Protocol

THIG Topology Hiding Inter-Network Gateway

TISPAN Telecoms & Internet converged Services & Protocols for Advanced Networks

TLS Transport Layer Security

TS Transport Stream

TV Television

UDP User Datagram Protocol

UE User Equipment

UI User Interface

UMTS Universal Mobile Telephone System

UPSF User Profile Server Function

URI Uniform Resource Identifier

VCEG Video Coding Expert Group

VLC Video Lan Media Player

VoD Video On Demand

VoIP Voice Over IP

WMV8 Windows Media Video 8

WMV9 Windows Media Video 9

1

Introduction

Contents

1.1 Motivation and Objectives

The telecommunications world has dramatically evolved over the last 20 years. Until recently telephone network operators were focused mainly on telephony and data services, cable network operators focused on television distribution and mobile network operators primarily offered mobile telephony.

This market and service segmentation vanished as different types of operators started providing services out of their previously core business. As a result, competition has emerged between markets that were previously unrelated. Nowadays a telecommunication operator can offer the so called Triple-Play packages; essentially made as a bundle of three basic services: voice (telephony), video (television) and data. One common technology allows these services to co-exist on one network infrastructure, the Internet Protocol (IP) [4]. The implementation of IP technology over many access technologies, such as Digital subscriber line (DSL), 3G Mobile networks and Cable Networks (DOCSIS) [5], has allowed a convergence of data services on all fronts. Currently It is even possible to provide television services on a mobile device or over a phone line (with DSL technology), something that was considered unfeasible a mere decade ago.

Providing these multimedia services requires guarantees of stability and reliability in order to fulfill user expectations and sustain satisfactory levels of quality of experience. To respect these requirements many procedures need to be applied for maximizing the efficiency of the transmission channel. Techniques such as video and audio compression, transport, control and signaling protocols and streaming architectures [6] allow a service to minimize its transmission footprint and to subsist along with various other services in the same infrastructure.

Next Generation Networks are introducing new concepts to integrate services on a common platform. When fully implemented, these services will not be limited to a single access network, but rather available from a single source to any IP enabled device on any type of access network. This integrated diversity poses a huge challenge for a service like television, as current deployments are highly customized to the network they are offered on.

This dissertation presents a novel architecture for providing converged SIP based IPTV server in an IMS environment and describes the experience of the development and evaluation of the IPTV Server prototype for that architecture.

1.2 Contributions

The presented architecture was developed as part of the European project My eDirector 2010 [7] and the implemented architectural prototype was deployed and tested on a third generation CDMA2000 mobile network operator in Portugal. This dissertation is focused on the application server part of the project while another project developed the client application. The

described architecture also resulted in an article [8] that has been accepted for the Workshop on IPTV Technologies and Multidisciplinary Applications of the 10th International Conference on Telecommunications 2009 [9].

1.2.1 My eDirector 2012

Under the European Seventh Framework Programme (FP7) [10], the My eDirector project [7] aims to provide an innovative interactive television service to viewers. The interactivity is available through an automated intelligent director which will provide a means for end-users to identify, select and watch certain points of interest within live broadcasted scenes.

Its deployment should be feasible for large scale broadcasting events, where several smaller events take place over a span of multiple venues with multiple actors in rapid changing scenes. Such an event is the 2012 Olympic games, where the system will be showcased.

This thesis aims to provide a initial study into the adaptation of live content to heterogeneous devices, such as common **STB!** (**STB!**) and smartphones which may be used in the My eDirector project. The content adaptation mechanism described in this document describes a terminal centric approach which allows the end device to request content that it can support. As such, it may provide the My eDirector project with the ability to be deployed as a generic service that can be suitable to provide service to a wide variety of end user technologies.

1.3 Dissertation Research Topics

For the development of an IPTV Application Server architecture several research topics were explored:

- IPTV architectures. What are the various technologies that are the basis of the IPTV service? How are they currently deployed?
- Video Technology. How is multimedia content delivered over digital networks? What are the requirements from a network perspective?
- IPTV integration in Next Generation Network (NGN). How can NGN be leveraged to deploy an IPTV service? What are the benefits that arise from this integration?
- Adapting to Heterogeneous Networks. What mechanisms can be defined to allow a dynamic adaptation of the IPTV service to network conditions and characteristics?

These topics are discussed throughout the dissertation and are answered in the final chapter.

1.4 Dissertation Layout

This dissertation is composed of 7 chapters that share a common structure. They are logically separated in three main sections: a general introduction, the body of the chapter (where the subject matter is discussed), followed by a succinct summary and conclusion of the discussed topics.

Chapter 2 gives a introduction to the state-of-the-art technologies and methods used to provide television service over IP enabled networks as well as the related standards. It is important to follow these standards as to provide the Application Server with the necessary characteristics for interaction with user equipment. The main technologies used in the provision of IPTV are discussed, including: multimedia encoding algorithms, signaling protocols, architectures and modes of transmission. Some commercial and open source application servers available are also studied and compared.

In chapter 3 the NGN concept is introduced and the IMS architecture is described. If integrated with IPTV, the networks can leverage existing, previously independent services, to create new composite applications to the user. The European Telecommunications Standards Institute provides a framework for this integration, based on the IMS architecture. Providing an IPTV service on the IMS architecture presents various requirements that the service and network must abide to, these are also detailed within the chapter. Finally, benefits of the IPTV and IMS consolidation are analyzed.

Chapter 4 describes the SIP based IPTV architecture developed with the focus on the Application Server. It starts with the requirements that the Application Server must conform to, followed design of the architecture and signaling protocol the SIP for all session and media signaling. This protocol is used in IP-based telephony, Instant Messaging (IM), Push-to-Talk (PTT) and other services, it can be easily expanded for IPTV services with methods also described in this chapter. To provide PVR functionalities in addition to the traditional television service several scalability (in terms of concurrent client sessions) and functionality considerations are made. As such, the AS implements two modes of operation, a per client mode and a general purpose mode. Finally, the connection of the AS in the IMS architecture is analyzed and final conclusions are made.

Chapter 5 describes the prototype implementation process of the AS. The features and limitations of the libraries used are also presented. These libraries implement streaming functionalities and session protocol used in the prototype. The AS architecture is modular and consists of main blocks containing several modules each. Each module and block are explained and their limitations discussed.

Chapter 6 describes the AS evaluation tests. The two operation modes that the server allows are studied and the overall media functionalities are validated. The proposed IPTV signaling protocol is studied and compared with most commonly used protocols for IPTV. The tests were

executed on a mobile CDMA2000 network and on a Local Area Network (LAN) in order to validate the adaptation methods on heterogeneous networks.

Chapter 7 summarizes the contributions of the dissertation, and also presents the architectural shortcomings and suggests areas for future work.

2

State-of-the-Art

Contents

2.1 Introduction

This chapter starts with an overview of the models that allow streaming of multimedia content along with the platforms and architectures for video streaming, followed by the protocols for signaling and the multimedia codecs used on network elements and client devices. The chapter finishes with an overview of the online services available for both mobile and fixed streaming applications.

2.2 IPTV and Internet TV

With the definition of multiple protocols that allow video and audio streams to be sent over IP networks, we have witnessed two main IP network based television concepts arise, the IPTV and the Internet Video. These two concepts are based on the same basic multimedia encoding techniques and transport protocols, but similarities end there [11].

IPTV can be described as the delivery of televised multimedia services over IP based networks. These networks must be managed to provide the required level of quality of service and experience, security, interactivity and reliability [12]. The network is usually an operators private network, where it can duplicate or exceed services offered by traditional broadcast television.

2.2.1 IPTV Services

There are two main services offered in IPTV, the continuous stream transmission and the VoD content. Continuous stream transmission is analogous to traditional television distribution, where a live channel is broadcasted to the viewer. The viewer has to join a ongoing stream in order to watch its contents. Video on Demand, as the name suggests, allows the viewer to request a server to stream pre-stored content. Additionally, IPTV can offer interactive sessions over these two services to create a variety of derived services. This is possible due to the underlying bidirectional communication that these services provide, allowing viewers to manipulate content and participate actively during the session.

By providing an interactive and unique platform for multimedia distribution, IPTV enables the creation of innovative services that may be used as building blocks for future IPTV's killer application [1]. These services are:

- *Broadcast Television* - Traditional television, multiple–view television and subscription services (pay-per-view), in Standard Definition (SD) or High Definition (HD) and Electronic Programming Guide (EPG);

[1] Killer Application - a feature, function, or application of a new technology or product that is presented as virtually indispensable or much superior to rival products.

- *On request services* - Video and music on demand, respectively VoD and Music On Demand (MoD);
- *Public interest* - Emergency broadcasts, information and news feeds;
- *Advertising* - Segmented and interactive advertising, on request advertising (such as movie trailers);
- *Interactive services* - Communication (such as social networking, IM and e-mail), general information (such as weather forecasts and forums), commerce (e-Commerce and Online Banking) and entertainment (such as games and blogs);

2.3 Content Transmission

In IPTV, content has to be sent from a serving entity to one or more receiving clients. In IP networks, data packets (e.g., multimedia packets) are usually sent from a single source to a single client. Protocols that support this one-to-one transmission are known as unicast protocols. But there are other cast modes that may provide network conditions for distributing content from the same source to multiple clients without overloading the network. These are the multicast protocols.

2.3.1 Unicast

By using unicast protocols it is possible to emulate the traditional television service by creating multiple copies of one stream of multimedia, one for each client. This form of transmission is known as unicast [13] and is depicted in figure 2.1.

Considering N clients receiving the same content via this unicast solution, at least N unique copies of each packet will need to be sent from the source host. If each stream has a constant bit-rate of 300 kbps, the server host will need to continuously transmit $300 \times N kbps$. As N grows, the amount of available network bandwidth steadily decreases. As such, deploying a large scale distribution services via unicast can present severe scalability issues. Additionally, the server has to process a much higher amount of information to cope with the various simultaneous streams increasing hardware requirements [13].

2.3.2 Network Multicast

Another mechanism to distribute a channel to multiple clients is the network multicast. Unlike unicast, network multicast requires adequate support at the network layer. In network multicast, only one stream of the same content is transmitted from the source. When a network router receives a multicast packet, it determines to which outgoing port the packet should be sent, making

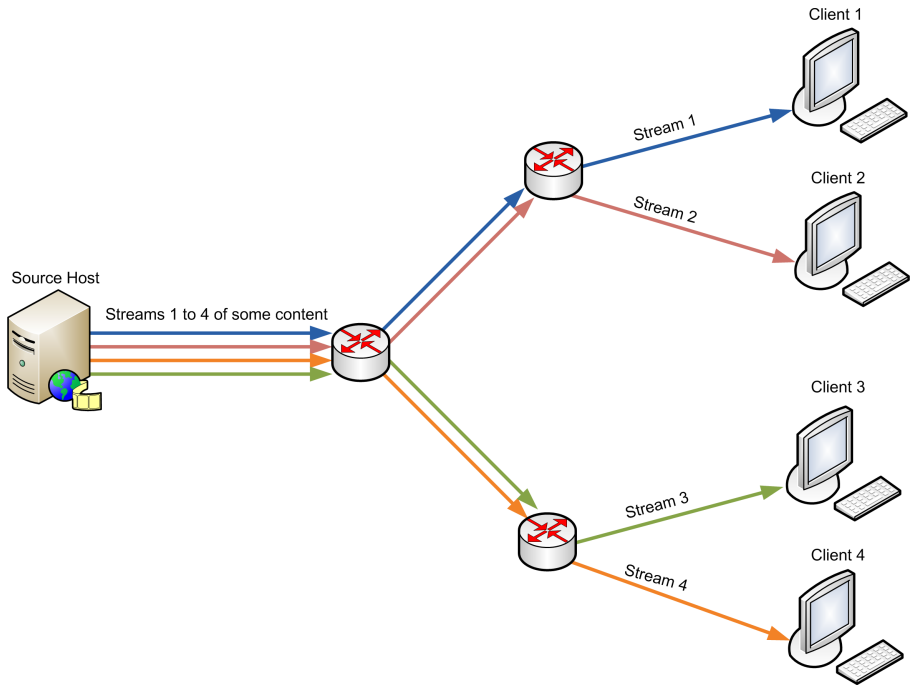


Figure 2.1: Unicast transmission of content to four clients

multiple copies if required. This solution depicted in figure 2.2, guarantees that only one copy of a packet will cross each link until reaching the recipients, increasing the system overall scalability [13].

While unicast packets specify the destination of a packet, multicast packets identify a group of recipients that belong to a multicast group. The group is identified by a specific IP ”group address”. Multicast mode is implemented with Internet Engineering Task Force (IETF) Internet Group Management Protocol (IGMP), that creates the mechanisms for group and membership management. Multicast enabled routers allow IP hosts to join and leave multicast groups by using IGMP messages. IGMP is also used by routers to query multicast groups and to evaluate if specific hosts are part of multicast groups.

After an IP host has sent a IGMP ”Join” to its serving router, the multicast router relies on Protocol Independent Multicast (PIM) [14] to signal neighboring routers to join the multicast group by means of a PIM ”Join” message. This creates a multicast distribution tree, where new branches are created when a client joins a group.

From the serving host perspective, the transmission of the multimedia data is analogous to a single client transmission (only one stream transmitted).

The main benefits arising from network multicast are scalability, increased network efficiency and reduced server load. However, deploying a multicast aware network requires routers that support the feature. And deployment of multicast is usually done at the Autonomous System of an Internet Service Provider (ISP). Global inter-domain multicast involving various Autonomous

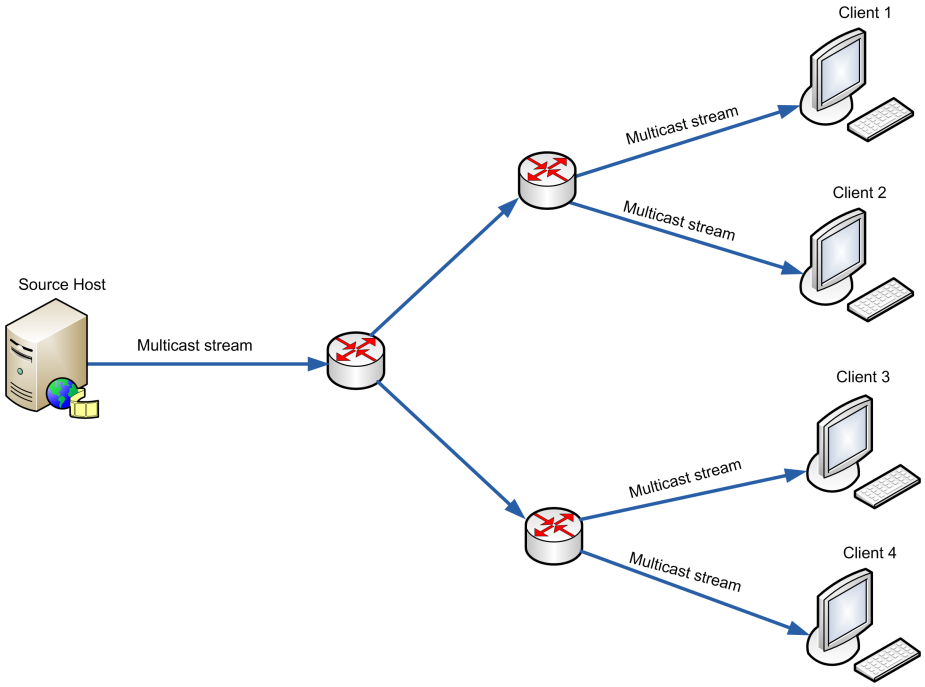


Figure 2.2: Multicast transmission of content to four clients

Systems, although technically possible is difficult to implement due to economic and commercial interconnect issues [15].

2.3.3 End-to-end Transmission

In order to assess the feasibility of a transmission of multimedia data from server to client there are three metrics that can influence the End User's experience: the start-up delay, the end-to-end delay and the playback continuity. The start-up delay represents the delay between the request for the media and its subsequent display on the client. The end-to-end delay is the playback delay between the source and the destination of the media. And the playback continuity (measured by the percentage of received packets) in practice represents the ”hiccups” that a user witnesses when watching streamed content [16]. Multimedia data transmission may be done in three modes, streaming, downloading and hybrid progressive downloading.

The Streaming mode allows a client to view compressed video and audio without having to download the entire content. This is accomplished by using a buffer to store codec frames. With the use of buffers, the client is not required to store any persistent data. With this mode, random access functionalities such as PVR functions - are feasible during playback and start-up delays are typically short. However playback continuity is dependent on the status of the buffer as whenever data in the buffer is low the continuity can be broken [17].

The Downloading mode implies receiving the whole multimedia content from the server in the form of a file then reproducing it locally on the client. Typically, IPTV does not support this

form of transmission due to large start-up delays with large content, however, once the file is stored, playback continuity is guaranteed. The disadvantage of this form of transmission is that traditional television service cannot be offered, because broadcast television channels are a set of continuous streams of video and audio (for each channel). On the other hand content provided by downloading mode, corresponds to discrete content elements [11]. The advantages are: no time sensitive encoding requirements and simplicity of deployment over any type of network regardless of bandwidth and latency. The typical service that is offered in this mode is VoD of finite length videos.

The Progressive Downloading mode is a hybrid approach that allows a client to download part of the file and visualize its content as it arrives. Just as Downloading mode, this streaming mode uses the persistent storage in the client to save the files. This method does not allow full random access features as is the case with the Streaming mode, but bears similar start-up delays and better playback continuity.

2.4 Quality of Service in IPTV Systems

In real-time based systems, such as streaming applications, QoS is an important factor in the deployment of a commercial IPTV system. QoS can be described as an network operator's commitment with its customers to provide and maintain values of service parameters and characteristics that are acceptable and expected [18]. For realtime based applications, such as IPTV, the goal of QoS is to offer a minimum level of control over the current "best-effort" service provided by typical IP networks [19].

However, QoS management for digital video transmission over the Internet is an overwhelming task, due to the lack of guarantees of data transfer speeds and delivery times. The difficulties are increased if mobile wireless networks are considered, as available resources such as bandwidth and capacity are scarce and vary immensely over time [18] [20]. Additionally, as more end users connect to a service, degradation of service quality is seen in direct result of diminished network capacity [19].

QoS is not limited to the physical aspects of transmission as the overall quality of the session is greatly influenced by the encoding parameters used and the error and congestion control. The variety of end user equipment also bring particular QoS requirements in terms of latency, encoding parameters, processing power and bandwidth.

In order to provide video services with end-to-end QoS support there are two main approaches: QoS provisioning from a network perspective and QoS control from the end-system's perspective [20].

Network centric QoS provisioning mechanisms have been standardized by the IETF and are the Integrated Services (IntServ) [21] and Differentiated Services (DiffServ) [22] mechanisms.

DiffServ sets a marker in a data packet to select the next Per Hop Behavior (PHB) at a network element level. The PHB specifies how the scheduler should queue and transmit the packet, allowing packets to be prioritized according to their Class of Service. This provides a network QoS mechanism that is a improvement over the ”Best-effort” mechanisms present in IP networks [22]. IntServ on the other hand integrates services by means of a Resource Reservation Protocol (RSVP) that signals the network elements in a path to create an end-to-end reserved channel. The reserved channel can act on resource reservation or on admission control of data in the network. Both standards provide QoS on a network level however DiffServ provides a simpler mechanism that does not require explicit resource reservation and network signaling, needing just pre-programmed router parameters and marked packets, making it the most used QoS mechanism in best-effort networks [19].

End-system QoS control mechanisms provide end-to-end QoS support using various techniques, such as error, congestion and power control as well as multimedia encoding algorithms (such as the MPEG-4 Part 10 codec [1]). The goal of these techniques are to maximize QoS while not requiring any core network QoS mechanisms. This minimizes change requirements in the network but requires highly efficient control mechanisms.

2.5 Architectures

There are two main architectures for IP based television. Both architectures have their advantages and disadvantages and can be applied concurrently by an operator or service provider. They are the Server-Client architecture and the Peer-To-Peer architecture.

2.5.1 Server-Client Architecture

The first and simplest architecture is client-server based depicted in figure 2.3, where the client downloads or plays streams of multimedia content from a server. Typically the multimedia files are stored in a server or transcoded from a headend and sent to the client. This simplicity has allowed for a rapid deployment of infrastructures offering this type of service, such as the popular Youtube [23] for downloadable media or the Justin.tv [24] for live streaming channels.

2.5.2 Peer-to-Peer Architecture

With the ever increasing bit-rates that end users connect nowadays to the Internet, a more recent architecture for IP television has surfaced. It is based on a distributed Peer-To-Peer (P2P) model; in the P2P model each client receives video and audio chunks from other participants on the network and displays the output as it arrives, as shown in figure 2.4. The client then acts as a serving host and can forward the chunks to other requesting participants (see figure 2.4).

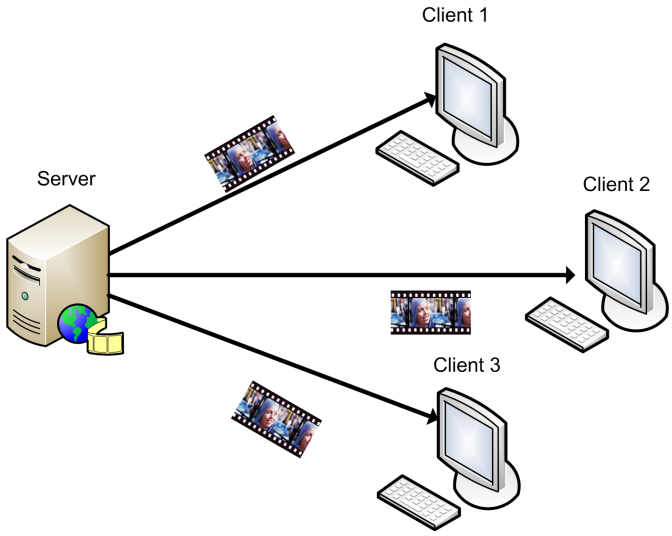


Figure 2.3: Server-client IP based television architecture

The benefits of this architecture are the scalability that P2P systems offer and the affordability of maintaining such a system. However, the nature of this solution makes other pressing issues difficult, such as digital rights management, longer start times compared to other solutions and the inherent instability due to the lack of well known serving hosts [16].

2.6 Transport, Control and Signaling Protocols

Just like traditional telephony, IPTV requires a signaling protocol in order to initiate, modify and terminate sessions under an IP environment. Control protocols are responsible for controlling the multimedia streams, introducing networked features that were previously restricted to video recorders, such as play, pause, rewind and forward. Finally, transport protocols are responsible for transporting the multimedia streams to the end device, be it a set-top-box or mobile device.

2.6.1 Realtime Transport Protocol

The Realtime Transport Protocol (RTP) is an application-layer protocol responsible for realtime delivery of, typically, multimedia data between two or more endpoints [25]. The primary services that RTP offers are:

1. *Payload Type* - The type of data transported in a RTP packet. There are formats specified for each type of data. As example formats 96 and 97, correspond to video and audio;

2. *Sequence Numbering* - A number present in the RTP packet header, that increments by one for each RTP packet sent. The initial number sent is random to increase security (in order to

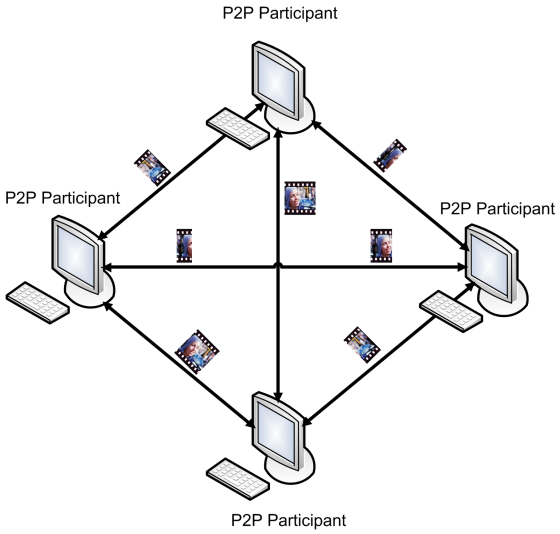


Figure 2.4: Peer-2-Peer IP based television architecture

make known-plaintext attacks more difficult). This numbering scheme allows the destination to detect packet loss and out-of-sequence packets;

3. *Timestamping* - In real-time sensitive applications, it is important to keep synchronization between sender and receiver. RTP implements a timestamping feature in its header, reflecting the sampling instant of the first octet of the data packet. The timestamp also allows statistical measurements, such as the calculation of end-to-end jitter;

4. *Delivery monitoring* - By using the RTP Control Protocol (RTCP) it is possible to monitor data distribution [25]. This is accomplished by a periodic transmission of control packets by all participants in the session. Services performed by RTCP are:

   (a) Feedback of quality of received data. It is related to flow and congestion control functions of lower level protocols;

   (b) RTCP carries a transport-level identifier for an RTP source, the Canonical Name (CNAME). This allows receivers to identify participants in one or more related RTP sessions, as well as multiple streams (such as video and audio multiplexed streams);

   (c) In an environment of multiple participants, control packets can flood the network with session reports. For this, RTCP has a self controlling mechanism to prevent such a situation;

2.6.2 Real Time Streaming Protocol

The RTSP [26] unlike the it suggests does not typically stream contents (although it can) or define the media to the end client. It is an application level protocol that establishes and controls media streams (e.g., Setup, Teardown, Play and Pause).

The definition of the streams to control are not specified by the protocol, this is left to a presentation description protocol such as Session Description Protocol (SDP) [27], the only requirement is the inclusion of at least one RTSP Uniform Resource Identifier (URI) in the media description. The content is to be streamed by a transport protocol such as RTP [25]. Additionally, RTSP does not infer a notion of connection, but of session identifiers that label each client-server session. This allows RTSP to be flexible to transport protocols (such as Transport Control Protocol (TCP) and User Datagram Protocol (UDP)) and allow clients to open and close connections to the server, by maintaining just the session identifier.

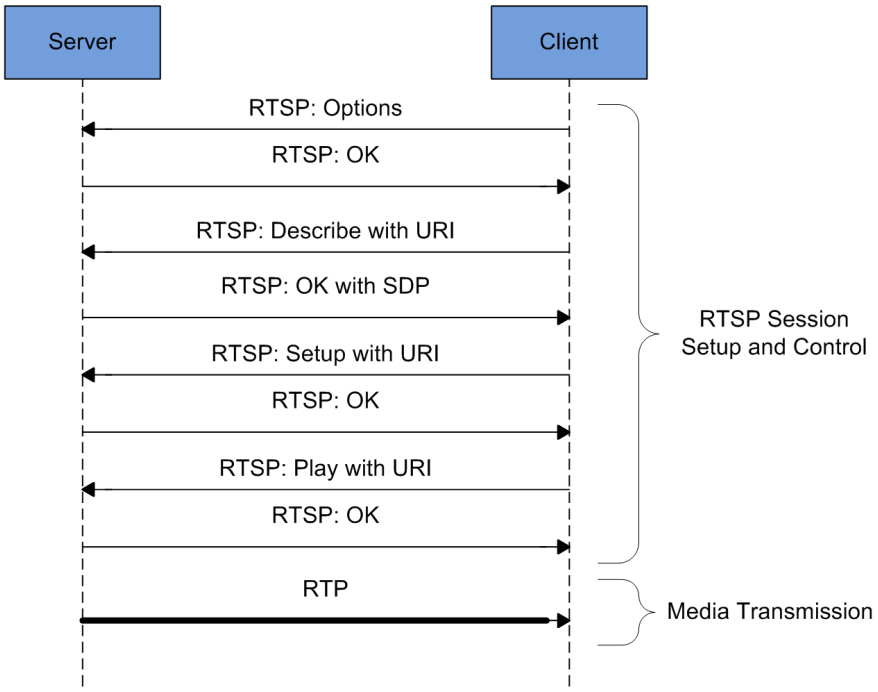


Figure 2.5: A RTSP session setup and control

RTSP was built in manner of operation and syntax similar to the version 1.1 of the Hypertext Transfer Protocol (HTTP) [28] in order to provide interoperability and extensibility of features. Despite the sharing of syntax and operation, RTSP and HTTP differ significantly:

- In HTTP, data is sent via the body section of the message, with the use of Multipurpose Internet Mail Extensions (MIME) [29]. Although possible, RTSP does not typically encapsulate stream data in its packets - but refers to a transport protocol (such as RTP) instead;
- RTSP defines method requests that are not available in HTTP, such as Play and Pause;
- Unlike HTTP, the RTSP protocol supports both client and server requests.
- HTTP requires an always open connection due to the use of TCP as a transport protocol, RTSP supports connectionless protocols such as UDP.

The example of a RTSP session setup and control message sequence is shown in figure 2.5. From a security perspective, besides the network and transport layer security implementations, RTSP can share the same authentication mechanisms provided by HTTP, such as basic and digest authentication.

The protocol, as specified, is able to share multi-client session, allowing several control streams to manipulate a media stream (through the use of a single session identifier). However, the protocol does not clarify how the multi-client scenario negotiation phase is set up. As the server does not specify transport means, it fully supports both Unicast and Multicast modes of operation.

2.6.3 Session Initiation Protocol

The SIP [30] was defined by the IETF to provide a general purpose, simple, agile and robust signaling protocol. SIP is a text based application-layer control protocol and is independent from lower level transport protocols allowing any form of session to be established; its main functions are to create, modify and terminate sessions between two or more peers (called user agents). SIP applications range from telephone and videophone calls to instant messaging. Along with extensions, SIP is the ”de-facto” protocol for signaling and session control in the next generation network IMS, as defined by the 3GPP[2] and 3GPP2[3] [31] [32].

Sessions are created by using SIP invitations that allow multiple user agents to specify sessions parameters that are compatible with all of the sessions peers. Because user agents can reside on a variety of networks, SIP makes use of a network of hosts called proxy servers to allow user agents to use several features, such as location registration. This location service enables user agents to send and receive invitations to sessions, as well as modifications to ongoing sessions. This also allows user agents to have mobility between networks. This mobility is accomplished by the definition of a universal identifier assigned to a user agent. In order to register a external identity the user agent contacts and registers itself in a Registrar server. This Registrar server serves as a directory for user agents to locate each other.

SIP multimedia sessions are established through five elements, considering the originating user agent (the calling party) and the terminating user agent (the called party), these are:

1. User location: location of the called user agent or end system to be used during the multimedia session;

2. User availability: availability of the called party to establish a multimedia session with the calling party;

3. User capabilities: determination of multimedia parameters to be used throughout the session, dependent on both called and calling party;

[2]3GPP - 3rd Generation Partnership Project

[3]3GPP2 - 3rd Generation Partnership Project 2

4. Session setup: Establishment of session between both called and calling parties. In telephony this is the ”ringing” stage, dictating when both ends of the user agents are ready to receive the multimedia stream;

5. Session management: this facet of SIP manages the modification of session parameters, invocation of services and termination of the session.

From a security point of view, SIP offers some security mechanisms. It allows replay protection and one-way authentication. Confidentiality issues are not easily addressed because proxy servers need to have access to the SIP headers in order to forward the SIP requests to their destinations. Secure / Multipurpose Internet Mail Extensions (S/MIME) [33] can be used to secure the body of the SIP messages, but it still allows an attacker to see the header of the packet [32]. At this level, content security or data encryption can be provided by using network level security (such as the IP Security (IPsec) [34] - suite of protocols) or Transport Layer Security (TLS) [32]. Authentication is provided by mechanisms offered by HTTP, by placing digital signatures in the header and non-repudiation and integrity measures.

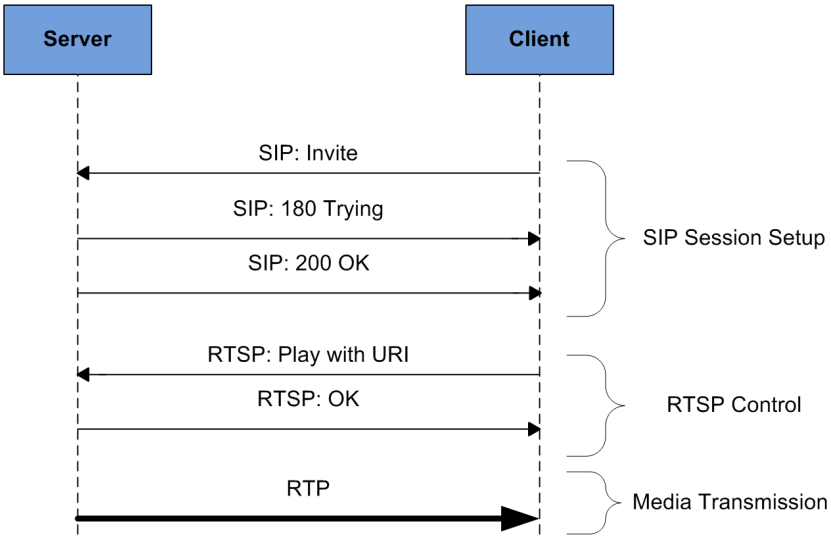


Figure 2.6: A hybrid SIP-RTSP approach

SIP is designed to be used with other protocols to form a multimedia system. For instance, in Voice Over IP (VoIP) systems, SIP is used as a signaling protocol and RTP [25] is used to stream the audio data to and from each endpoint. As referred in 2.6.2, RTSP provides session and control management, but for some applications, it may be desirable to use SIP for session control [35]. In this case, an internet draft by the IETF is in specification where SIP and RTSP can be used together [35]. SIP would provide session setup, modification and teardown and RTSP would provide media control mechanisms, such as play and pause. An example of the hybrid SIP-RTSP message sequence is shown in figure 2.6.

This approach allows a new composite media control protocol, other than RTSP, that can provide additional commands for multimedia streaming content. The requirements for such a media control protocol are as follows [35]:

1. The media control protocol (in this case RTSP) must support the following trick functions: play, pause, rewind, forward, fast forward, slow forward, rewind, fast rewind and slow rewind;
2. Negotiation of the media control protocol must be supported;
3. It must be possible to specify which media streams can be controlled by a given media control protocol;
4. The media control protocol must support asynchronous media notifications, such as end-of-stream;
5. TCP should be supported as a transport protocol;
6. A client can control a media stream independently of its location (on a home network or any other network);
7. If needed, the media control protocol should implement additional commands that are not available in RTSP to control the multimedia stream.

2.7 Multimedia Codecs

Streaming of multimedia content is quite demanding in terms of network capacity namely because bandwidth is a shared and scarce resource. As such, bandwidth toggling when streaming video is an important feature in order to provide scalability and reliability of an IP based video solution [36]. The easiest way to limit network usage is to consume as little bandwidth as possible. Because video and audio data are by far the most transferred in a multimedia session, it is important to compress the data before it is sent over the network. Compression and decompression of streams is achieved using encoders and decoders (codecs). Codecs have been constantly developed, and as such there are many specialized versions available. Examples are the ITUs [4] H.26x, focused on telecommunication applications such as video-telephony and MPEGs [5] standards, focused on rich multimedia applications such as high definition videos.

In order to achieve compression rates, video codecs rely on mathematical techniques, such as frame compression and adaptive perceptual quantization [37]. A video feed is comprised of many individual luminance and chrominance frames, these frames are then compressed by reducing

[4] ITU-T - The body of the ITU that coordinates standards for telecommunications.

[5] MPEG - Moving Pictures Experts Group, formed by the International Organization for Standardization (ISO) to set standards for video and audio compression.

spatial and temporal redundancy, using entropy encoding and motion compensation. Generally, codecs use three main types of frames (see figure 2.7):

I-Frames called Intra-frames, rely on Discrete Cosine Transform (DCT) (in the case of the MPEG family of codecs [1]) and entropy encoding (used by individual digital images), basically becoming compressed images of the original frames;

P-Frames (predicted frames) use similar encoding algorithms as the Intra-frames but adds motion compensation to previous Intra or Predicted frames allowing temporal redundancy to be explored, achieving a higher compression factor;

B-Frames (Bi-directional frames) just like Predicted frames, use the same spatial compression techniques, but can rely on previous and next Intra and Predicted frames. Bi-directional frames tend to achieve the highest compression factors.

A grouping of inter-dependent Intra, Predicted and Bi-directional frames are called a Group Of Pictures (GOP).

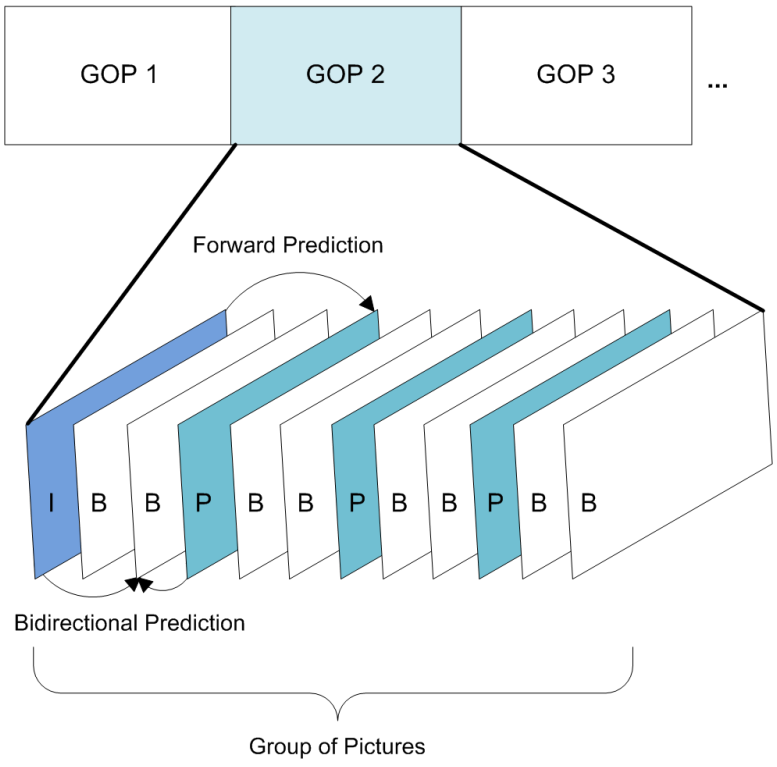


Figure 2.7: A 12 frame Group of Pictures

Quantization takes advantage of the human perception of lower and higher frequencies in images. In adaptive perceptual quantization, the images are split into a fixed number of frequency ranges. Less visually important frequencies have their DCT factor set to zero, while other frequencies are applied to predetermined coefficients, allowing only the visually important frequencies to

be compressed and transmitted [38].

2.7.1 MPEG-2 Coding

Until recently, the most successful codec used has been the Moving Picture Experts Group (MPEG) second standard in compression algorithms second group of MPEG codecs (MPEG-2) [39]. It's predecessor, first group of MPEG codecs (MPEG-1) [40], introduced the bidirectional dependent frames (B-Frames) but lacked overall quality at target bit rates [41].

MPEG-2 is interesting for streaming because it supports transport multiplexing of different MPEG-2 streams. Each stream, be it video or audio, is an Elementary stream (ES) that after packetizing composes a Packetized Elementary Stream (PES). After creating the PES stream, it is possible to multiplex the data in two formats:

MPEG Program Stream By tightly coupling various PES synchronized to a common time base. The usage of the MPEG Program Stream (PS) is better suited for low error transport mediums (such as Digital Video Disc (DVD)'s);

MPEG Transport Stream PES packets are split and packetized into the transport stream Transport Stream (TS). The TS possesses error correction mechanisms that make bit errors more recoverable, and so TS streams are more suited for higher Bit Error Rate (BER) media (such as the Digital Video Broadcasting (DVB) family of technologies).

Due to the widespread deployment of the MPEG-2 codec, many hardware and software based solutions have been designed, making coding and decoding of the multimedia fast enough to be compressed and decompressed on the fly, making it highly suitable for Live Television (TV) streams. The typical compression factors while keeping good video quality for MPEG-2 video are around 30:1 [41]. However, this value is highly dependent on the nature of the multimedia to compress, as the main factors for high compression values are low levels of motion (videoconferencing and video surveillance) and low level of detail.

Streaming a standard definition television quality channel using MPEG-2 can be roughly calculated, considering a Red Green Blue (RGB) 24 bit color pallet for each pixel and a resolution of 704 pixels wide and 576 pixels height at a compression factor of 30:1, each video frame will have a size of:

$$\text{Bits per frame} = \frac{24 \times (704 \times 576)}{30} = 324403.2\text{bits} = 0.3\text{Mbits} \tag{2.1}$$

At a constant 25 frames per second, even with the 30:1 compression factor, the average output will be roughly 8 Mbps per channel not including any audio channels and, in the case of delivery over IP, not including the overhead from headers of the various protocols discussed.

2.7.2 MPEG-4 - Advanced Video Coding

A more recent video coding standard is the H.264 Advanced Video Codec (AVC) [42]. This standard started being developed in 1998 when the Video Coding Expert Group (VCEG) as part of the ITU Telecommunication Standardization Sector (ITU-T), created a project called H.26L. The target for this project was to double the coding efficiency, meaning that for the same quality offered by MPEG-2, the new standard that resulted from H.26L would use half of the bit rate. For the example above and the same conditions, the coded bit-rate should be roughly 4 Mbps. After a call for proposals for the project, the MPEG formed a Joint Video Team (JVT) with the ITU-T and submitted for approval the H.264 AVC in 2003 [43].

As in previous projects, only the decoder is standardized allowing any video coding implementation to work with the decoder as long as it is in conformity with the restrictions specified by the standard. This freedom of implementation, allows optimization and customization of the coding block for different scenarios, be it over a low BER medium, like DVD, or highly error prone medium like satellite transmission, making it interesting for implementation on streaming solutions over mobile networks [1].

Several enhancements over MPEG-2 were implemented in order to improve compression and provide better resilience to errors:

Enhanced entropy encoding H.264 included a advanced arithmetic coding scheme known as Context-adaptive binary arithmetic coding (CABAC) and a context adaptive coding scheme known as Context-adaptive variable-length coding (CAVLC). Both coding schemes allow a higher compression performance than previous video standards.

Motion compensation Unlike MPEG-2, the size and shape of blocks referenced by motion vectors are variable, the motion vectors have quarter-pixel resolution (in MPEG-2 the resolution is of half-pixel), multiple reference pictures can be used for motion compensation and motion vectors may now point to outer picture areas;

Artifact reduction H.264 allows an adaptive filter to reduce the blocking artifacts found in block based video coding (such as MPEG-2 and fourth group of MPEG codecs (MPEG-4)). The filter is tightly coupled to the motion compensation prediction loop and so, all improvements found in the Intra frames will be propagated to the other derived Predicted and Bi-directional frames. This filter is known as In-the-loop deblocking filtering. Artifact reduction is accomplished by improvement in the spatial redundancy calculations using smaller block sizes (4x4) and an integer transform instead of the DCT.

Greater error resilience A new feature in H.264 consists in new picture types (known as Switching Slice Predicted Frame (SP) and Switching Slice Intra Frame (SI) pictures) that allow the recovery of synchronization of parts of images without sending whole, bit heavy, Intra Frames.

In addition to the SP/SI pictures, H.264 allows the encoder to include within the video stream redundant low-quality sections of pictures. Sections of coded pictures may be sent and decoded with relative independence of the rest of the coded pictures. This design provides robustness on networks that have an out-of-order delivery behavior.

In order to address the various applications environments, three main usage profiles have been defined. They are the Baseline, Main and Extended profiles. The best quality is usually achieved by the usage of the Main profile; the drawback is the higher complexity due to the usage of CABAC and predictive tools. With higher complexion comes a higher delay, and for realtime requirements the Baseline profile is more adequate. The Extended profile is a superset of the Baseline profile, with the exception of a few tools; this allows baseline encoded streams to be decoded by the Extended profile [1]. Figure 2.8 illustrates MPEG-4 H.264 profiles and tools.

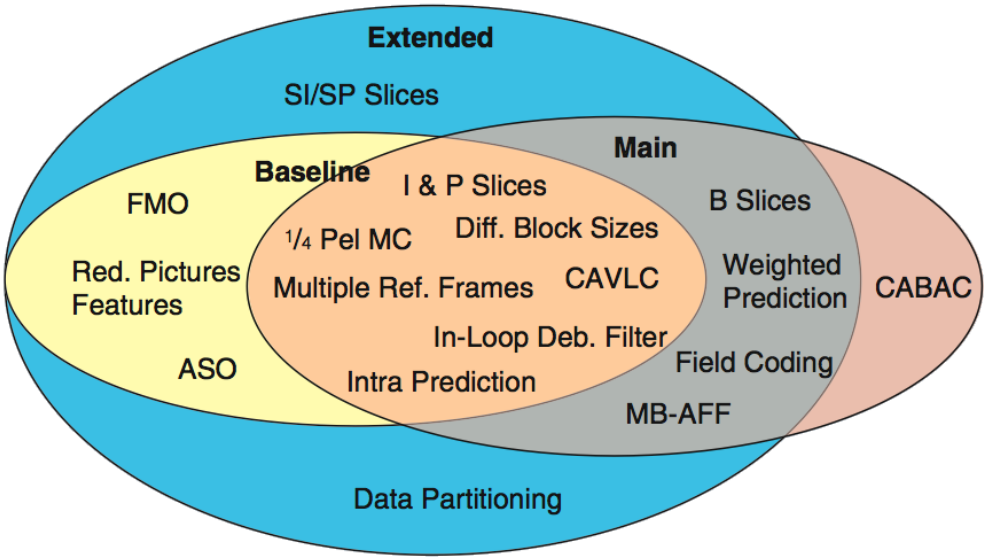


Figure 2.8: MPEG-4 H-264 profiles and tools [1]

Baseline Profile As a simple implementation for the H.264 codec, it includes all of the above mentioned tools, with exception to the B and SP/SI slices, CABAC coding, interlaced coding and other predictive tools used by the other profiles. The main targets are low complexity applications with low delay requirements;

Main Profile Sharing the same tools as the baseline profile, Main doesn't include the redundant pictures. However, it does include the CABAC entropy coding scheme, interlaced picture coding and most of the predictive tools excluded in the Baseline profile;

Extended Profile This profile includes all tools present in the H.264 specification with the exception of CABAC.

With the standard defined, proliferation of the H.264 AVC codec has been possible, being currently used in numerous devices, from IPTV set-top-boxes to cellular telephones, and has also been deployed in numerous IPTV architectures and commercial solutions. Several studies have shown that H.264 provides real and effective improvements on video transmission over IP networks [44].

2.7.3 VC-1 Coding

This coding scheme was developed and implemented by Microsoft for their proprietary video format, Windows Media Video 9 (WMV9). The VC-1 is a video coding format standardized by the SMPTE [6]. It provides higher compression factors and quality improvements than those found in Microsoft's Windows Media Video 8 (WMV8) codec. The VC-1 represents an alternative to ITU-Ts H.264 AVC, showing substantial benefits when compared to MPEG-2 [45].

VC-1's main features include the ones already discussed in paragraph 2.7.2 MPEG-4, with little variation between the two codecs . Tests have shown that both video coding standards have similar subjective quality and compression results. Complexity comparisons between the two codecs have shown favorable results for the VC-1 codec. While it is difficult to evaluate quality of experience, Peak Signal-to-Noise Ratio (PSNR) comparisons have shown a VC-1 image quality to be slightly superior to that of H.264 [46].

2.7.4 AAC Coding

While video occupies most of the bandwidth in a multimedia session, audio quality is also an important component in the overall QoE while watching a video stream. The most promising codec for audio is the MPEG-4 Advanced Audio Coding (AAC) [47], based on the previous standard MPEG-2 AAC. The MPEG-2 version of AACs main goal is to provide an indistinguishable quality encoder when compared to the original uncompressed audio signal.

The AAC encoder is a time/frequency (T/F) based codec, which means, audio is encoded using spectral representations (in the frequency domain) of the input audio. This type of encoder has numerous benefits for audio compression: it allows filtering of irrelevant audio input and explores redundancy removal techniques. With these techniques, it is possible to compress audio while keeping the perceptual quality unaltered. Due to the outstanding quality and performance achieved at low bit-rates (from as low as 16 kbps), MPEG-2 AAC was chosen as the main audio encoder for the MPEG-4 format as MPEG-2 AAC already provides indistinguishable audio encoding from bit-rates at 96 to 128 kbps for stereo audio and 256 to 320 kbps for 5 channel audio signals.

[6] SMPTE - Society of Motion Picture Engineers

The main additions of the MPEG-4 version of AAC to MPEG-2 AAC are related to streaming applications such as low delay encoding, error resilience and robustness. As mentioned before, error robustness at the encoding level is an important feature to reduce transmission protocol dependence and delay in any realtime system.

2.7.5 Codec Comparison

Although the MPEG-2 video coding and compression algorithm is still the most supported standard, the H.264 and the VC-1 encoders are becoming the replacement of choice due to their higher compression factors. In fact, the two codecs have a compression efficiency that approximately doubles that of the MPEG-2 video codec. However, both H.264 and VC-1s complex algorithms make real time encoding very computationally demanding.

A moderated video standard included in the MPEG-4 group of standards is the MPEG-4 Part 2 [48], also known as MPEG-4 Visual. This video codec shares many of the features found in MPEG-2 and the MPEG-4 Part 10 (H.264) codec. As such, it achieves a compression rate better than that of MPEG-2 while being less complex than H.264 and VC-1. Table 2.1 summarizes the main differences between the four codecs.

Various services appeared based on the H.264 codec, from IPTV to DVB, taking advantage of the high compression factors and better error resilience. A few examples of services using these codecs are the MEO family of products by Portugal Telecom [49], that offers IPTV and Satellite broadcasting as well as the terrestrial DVB service currently being implemented in Portugal [50]. Additionally, in Portugal the ISP Sonaecom [51] currently offers its IPTV platform using MPEG-2 Visual codec instead of the MPEG-4 family of codecs.

2.8 Streaming Servers

As video streaming interest increases, many servers capable of streaming multimedia content over IP connections are being developed. The main solutions available nowadays can supply pre-configured systems providing advanced streaming options on a User Interface (UI) that is both powerful and easy to manipulate. These solutions are the Apple Quicktime Streaming Server, the Helix Video Server and the VideoLan Media Server.

2.8.1 Quicktime Streaming Server

Created by Apple, the QuickTime Streaming Server (QTSS) offers a multimedia streaming server based on RTSP and RTP protocols [52]. Codecs supported are the MPEG-4 family and MPEG-1 Audio Layer 3 (MP3). Management is facilitated by means of a simple Graphical User

	MPEG-2	MPEG-4 Visual	AVC/H.264	VC-1
Profiles	4	19	4	3
Input source	Interlaced Progressive	Interlaced Progressive	Interlaced Progressive	Interlaced Progressive
Chroma formats	4:2:0, 4:2:2	4:2:0, 4:2:2, 4:4:4	4:2:0, 4:2:2, 4:4:4	4:2:0
Complexity	Low	Moderate	High	Moderate
Streaming tools	-	Scalable Video Coding (SVC)	Switching Slices	-
Encoding structure	Hierarchical Profiles@Levels	Hierarchical Profiles@Levels	Hierarchical Profiles@Levels	Hierarchical Profiles@Levels
Picture type	I, P, B	I, P, B	I, P, B, SP, SI	I, P, B, BI
Average SD throughput	4 to 8 Mbit/s	1,5 to 3 Mbit/s	1 to 2 Mbit/s	1 to 2 Mbit/s
Motion estimation precision	Up to ½ pixel	Up to ¼ pixel	Up to ¼ pixel	Up to ¼ pixel
Loop filter for blockiness reduction	Not implemented	Not implemented	Implemented	Implemented
Entropy encoding	VLC	VLC	CAVLC CABAC	VLC
Licensing	Open-source available	Open-source available for some profiles	Open-source available	Proprietary

Table 2.1: Comparison between codecs (adapted from [3])

Interface (GUI) and web based configuration page. QTSS supports multicast and unicast streaming of video content seamlessly. Services supported include:

1. **Live TV** - By using a third party program for capturing and encoding, such as QuickTime Broadcaster, QTSS can receive MPEG-4 encoded RTP streams that are announced by the SDP protocol;

2. **On Demand** - QTSS supports unicast and multicast streaming of pre-recorded multimedia as long as encoded using the aforementioned codecs with hinted tracks. Hinted tracks index the positions within the MPEG files allowing random access and synchronization between video and audio;

3. **Media Relaying** - By relaying multimedia streams to other video servers, QTSS allows load balancing of incoming connections, turning it into a fairly scalable solution.

The system is not tailored for cross-platform compatibility, being limited to Apple manufactured hardware. A derived open-source based project, known as Darwin Streaming Server (DSS) [53], allows the server to be run on various platforms, such as Windows and Linux. The only withdrawn

functionality is the system GUI. Other issues on both versions are the format of the videos, which have to be encoded on Quicktime derived products. On-the-fly transcoding can only be done using a separate streaming application and clients are limited to MPEG-4 compatible devices.

2.8.2 Helix Video Server

Helix Video Server is at time of written at version 12 and is produced by Real Networks software company [54]. It supports RTSP masked as HTTP for signaling. Codecs that are fully supported are the proprietary RealVideo and RealAudio, the MPEG-4 video, the H.264, H.263, AAC and the Adaptive Multi-Rate Compression (AMR). Two transport protocols are supported: RTP and Real Networks Transport Protocol (RDP) [54]. Services supported are identical to Apple's solution with the following additional services:

1. **Fast-channel switching** - Helix supports a web based feature that allows seamless changing of video or audio channels. The client starts by requesting a new channel via HTTP, the server proceeds to stop transmitting the previous channel and immediately starts the new channel with the most recent available intra frame [55];

2. **Web portal** - Helix allows third party web portals to act as interface for users to select multimedia streams. Streams are selected using a normal HTTP Get command;

3. **Authentication, Authorization and Accounting (AAA) support** - AAA functions are supported by the Helix streaming server.

Being a proprietary server implementation, Helix comes with increased cost. But an open-source project known as Helix DNA allows a free solution to be used. It also has a licensed mobile version specifically tailored for mobile devices. Although using, by default, proprietary protocols and encoders, users are required to install Real Networks client media player to use most of its functions. Additionally stored content cannot be transcoded on-the-fly.

2.8.3 VideoLan Media Player

Starting off as a engineering project for a university in France, Video Lan Media Player (VLC) has evolved to a highly customizable open-source multimedia streaming application [56]. The media player is the result of media player and media server integration into one simple application. It currently supports most codecs and offers various open source implementations of coders and encoders [57], providing transcoding mechanisms from acquired content.

Unlike previous streaming servers, it is highly portable, ranging from UNIX-based enterprise systems to handheld devices. Non-proprietary signaling and transport protocols (such as HTTP and RTP) are also supported.

It supports various interfaces such as web-based, telnet, command line and even infrared control (with certain operating systems). It is extensible by providing an Application Programming Interface (API), that allows third party applications to include the VLC media player and server functionalities.

2.8.4 Server Comparison

The Quicktime and Helix servers, although presenting an open API and open-source versions still have limitations due to proprietary functions. The Quicktime server only supports content that is coded on quicktime platforms (such as quicktime broadcaster and quicktime professional player). The Helix Video Server uses some codecs that are proprietary to Real Networks, as such, the user equipment must use the real player software to use the service.

	Quicktime Streaming Server	Helix Video Server	VideoLan Media Player
Suported platforms	Mac OS, Windows*, Linux*	Windows, Linux, Solaris	Mac OS, Windows, Linux, Mobiles
Open API	Yes	Yes	Yes
Suported Protocols/Transports	RTP, RTSP, SDP, UDP, TCP	RTP, RDP, RTSP, HTTP, SDP, UDP, TCP	RTP, RTSP, MMS, HTTP, UDP, TCP
Supports Live Streaming	Yes	Yes	Yes
Input Codecs	Quicktime Encoded MPEG-4, H.263	MPEG-4 Visual, H.263, H.264, RealAudio & Video	MPEG-2, MPEG-4 Visual, H.261, H.263, H.264, VC-1, RealAudio & Video, among others
Transcoding	No	No	Yes
Output Codecs	-	-	MPEG-2, MPEG-4 Visual, H.261, H.263, H.264, VC-1***, RealAudio & Video ***, among others
User Interface	Application, Web-based*	Web-based	Application, Telnet, Web-based, CLI
Licensing	Proprietary and open-source versions*	Proprietary and open-source versions**	Open-source

Table 2.2: Comparison between streaming servers

Although the three servers are very similar in modes of operation, as shown in Table 2.2, only the VLC supports real time transcoding. VLC is not only a server, but also a client which allow almost any source of video and audio to be used (such as DVD, RTSP streams, etc.). VLC also supports a much wider range of input and output (encoding) codecs than the other servers. VLC is also present on more platforms than the other solutions, including client versions for mobile devices.

2.9 Conclusion

The goals in this chapter were to study and analyze the state-of-the-art technologies in the dissemination of multimedia contents over IP networks and to clarify the differences between the often confused IPTV and Internet television concepts.

The benefits and drawbacks of the unicast and multicast multimedia transmission mechanisms were discussed for a better understanding of the scenarios where they can be applied. Therefore it was shown that unicast is more suited for personalized streaming, such as video on demand services, whilst the benefits of multicasts inherent scalability makes it more appropriate for broadcasting of multimedia to several users.

Also discussed was the importance of assuring quality of service and how it affects the user experience.

The server to client architecture was also presented and the more recent peer-to-peer architecture was introduced. Each of the architectures has benefits, but the greater ease in management of the server to client architecture reflects its widespread usage.

Signaling protocols that make the IPTV service possible are also discussed. Specifically the RTP which is responsible for transporting the audio and video streams to the end user and provides mechanisms to assure the correct visualization of the content. Followed by the RTSP and SIP protocols which allow Trick Functions and session control.

Finally a brief introduction to the state-of-the-art in multimedia codification and compression standards was discussed as well as the most common open-source streaming servers.

3

IPTV in Next Generation Network environments

Contents

3.1 Introduction

This chapter starts with brief overview of the IMS architecture and the IPTV service model to bridge the converged model of IPTV in IMS and its requirements.

3.2 Overview of the IMS Architecture

In terms of convergence, IMS originally designed by 3rd Generation Partnership Project (3GPP) and later updated by 3GPP, 3rd Generation Partnership Project 2 (3GPP2) and Telecoms & Internet converged Services & Protocols for Advanced Networks (TISPAN) [58] is considered the ”de facto” architecture for many within the telecommunications industry [59] and presents itself as the framework that allows the convergence of the Internet, wireless and wireline networks and the global platform for the delivery of IP multimedia applications in NGN. This All-IP network architecture is access medium independent and is based on packet based networks [31].

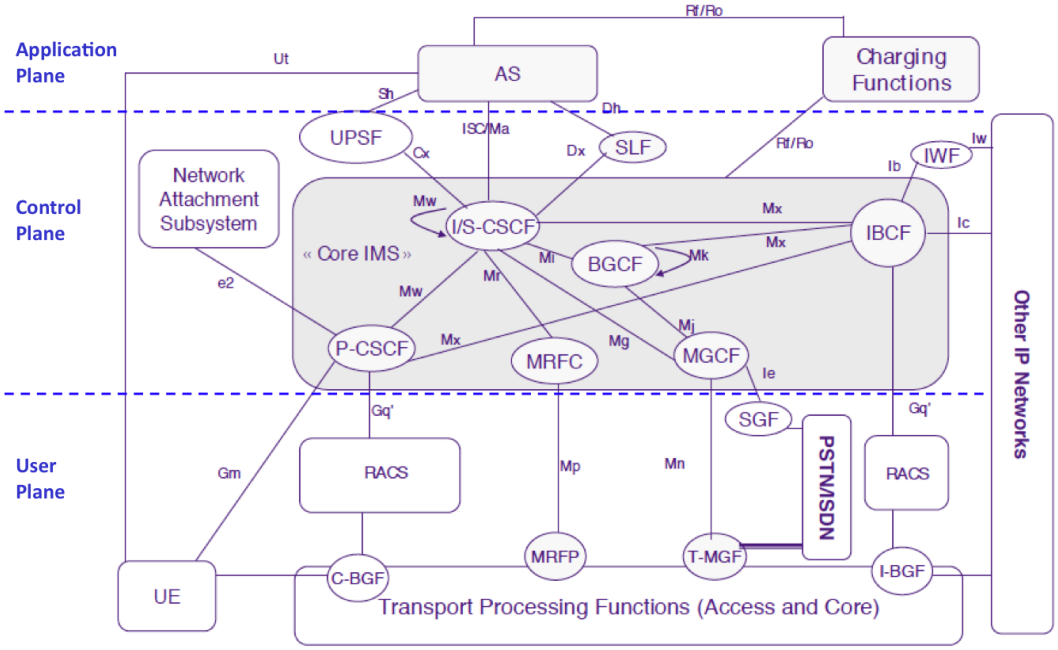


Figure 3.1: ETSIs IMS Functional Entities and interfaces [2]

The IMS architecture defines how services or applications interact with the underlying heterogeneous networks and how they communicate with the network provider’s back-end systems [60]. This architecture defines Functional Entities (3.1) and their roles organized in layers and planes (Figure 3.2):

- Service Plane (Application Layer)
- Session Control Plane (Session and Database Layer)

- Media Control Plane (Media Control and Gateway Layer)
- User Plane (Access and Transport Layer)

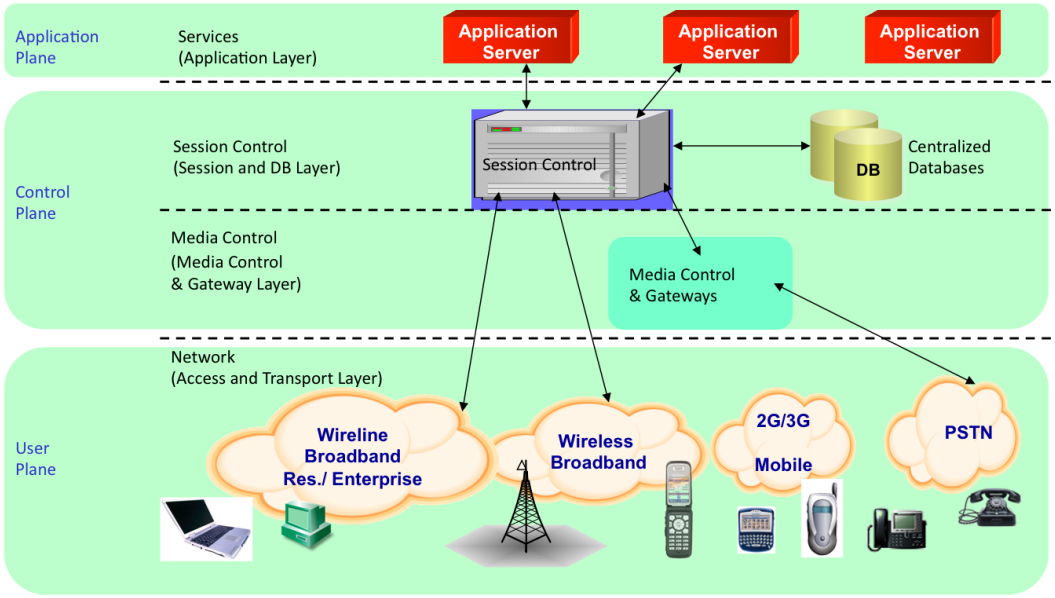


Figure 3.2: IMS architectural Planes and Layers [2]

3.2.1 The IMS User Plane

The IMS user plane provides a QoS-enabled IP core network with access from a User Equipment (UE) over various access technologies. It is designed to provide a wide range of IP multimedia server-based and P2P services. The access to the core network is through Border Gateways (such as the Interconnect Border Gateway Function (I-BGF) and Core Border Gateway Function (A-BGF)). These gateways enforce policies provided by the IMS core, controlling the flow of traffic between access and core networks.

Within the IMS User Plane the Resource and Admission Control Subsystem (RACS) controls the resources required for a session, these resources are defined by Interconnect Border Control Function (I-BCF) present in the Control Plane. The I-BGF and A-BGF provide media relay for hiding endpoint addresses with the use of Network Address and Port Translation (NAPT) functions.

3.2.2 The IMS Control Plane

The control plane is located between the Service and User Planes. It routes call signaling (Session Control), informs the User Plane of traffic flow policies (Media Control) and generates billing information for the use of the network. At the core of this is the Call Session Control Function (CSCF), which is composed of the following functions:

- The Proxy Call Session Control Function (P-CSCF) is the first point of contact for the users with the IMS. The P-CSCF is responsible for the security of messages transferred between the users and the network, it also allocates the necessary resources for media flows.

- The Interrogating Call Session Control Function (I-CSCF) is the first point of contact with other IMS networks, it is responsible for querying the Home Subscriber Server (HSS) the Serving-CSCF for a user and may use a Topology Hiding Inter-Network Gateway (THIG) to hide the network topology from other peer networks.

- The Serving Call Session Control Function (S-CSCF) controls the call sessions. It is responsible for processing user and call registrations to record each user location, user authentication and call processing. The S-CSCFs operation is controlled by policies stored by the HSS.

The Control Plane includes the User Profile Server Function (UPSF), it is responsible for maintaining the service policies for each user in a central location. It includes services such as: roaming information, telephony service information (such as call forwarding information), IM information (such as buddy lists), etc. This centralization simplifies user management and insures a consistent view of subscribers across all services.

The IMS Control Plane also controls the traffic within the User Plane through the RACS. This control consists of two functions:

- The Policy Decision Function (PDF) implements the local policy on resource usage.

- The Resource Access Control Function (RACF) controls the QoS within the access network.

3.2.3 The IMS Application Plane

The IMS Application Plane provides an infrastructure that allows the management and provision of services. It defines standard interfaces to IMS functions which include:

- The UPSF which provides configuration storage, location and presence of users and identity management.

- The Charging Gateway Function (CGF) that provides the billing functions.

- The control of multimedia calls and messaging which is provided by the Control Plane.

The various Application Servers (AS) are implemented in the Application Plane, they provide end-user service logic. AS have been developed for services such as telephony, PTT and IM.

3.3 Overview of the IPTV Service Model

The ITU has defined a reference model for the IPTV NGN architecture. The architecture considers four main roles [12] that may pertain to separate entities, for example, the content provider may be a media organization, the service and network provider may be a operator and together serve the customer with IPTV services, as depicted in Figure 3.3. A description of the roles are as follows:

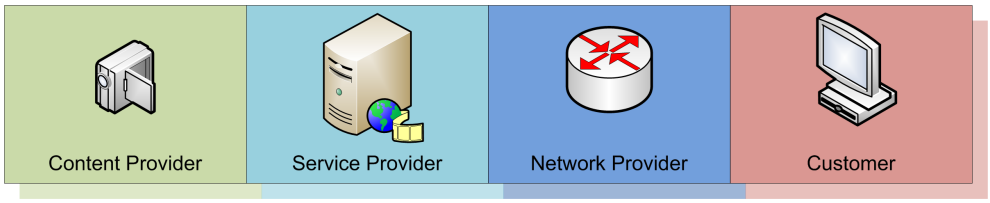


Figure 3.3: ITU reference model for IPTV in next generation networks

Content Provider This entity owns or sells the content to be streamed to the Customer.

Service Provider The IPTV service is provided by the Service Provider, the content is licensed or acquired from the Content Provider. The Customer buys the service which is a package that the Service Provider creates from available content.

Network Provider The connection between the Service Provider and the Customer is assured by the Network Provider.

Customer This entity purchases and consumes the IPTV service.

3.4 The ETSI View of IPTV in IMS

As the IPTV service becomes widespread the standardization of its architecture is fundamental. The ETSI[1] TISPAN[2] standards defines an architecture and set of procedures for the IPTV service within an IMS framework [2], depicted in figure 3.4.

The IPTV architecture relies on the IMS architecture, separating the control and user planes from the IPTV service [61]. The CSCF process incoming session signaling from the user equipment and calls the User Preference Serving Function to locate the user profile and preferences, as well as reserving network resources by contacting the RACS. After the network provisioning phase, the user may select the wanted service by contacting the Service Discovery Function The

[1] ETSI - European Telecommunications Standards Institute, an European standardization organization in the telecommunications industry

[2] TISPAN - Telecoms and Internet converged Services and Protocols for Advanced Networks, standardization body of ETSI

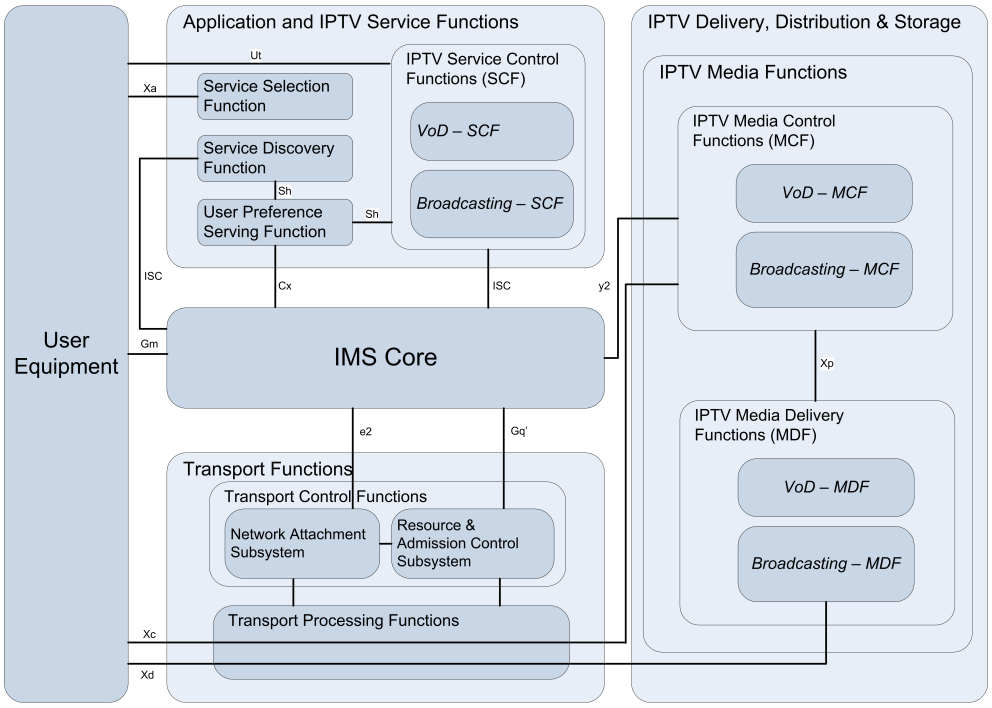


Figure 3.4: The ETSI proposed IPTV architecture in the IMS framework [2]

service is selected by obtaining the pertinent Service Provider server IP address and the connection is established. The user equipment may from this point onwards contact the Application Server to obtain session information such as available media parameters.

By contacting the service control functions, the service can perform the necessary authorization functions and select the IPTV media functions relevant to the user profiles and requests. The Service Control Functions acts as common IMS Application Servers by communicating with the core IMS Entities via standardized interfaces.

The ETSI architecture provides media control capabilities externally to the IMS core as to minimize the delay overhead introduced by the IMS functions. Therefore, the Media Control Function (MCF) acts directly with the user equipment via RTSP for Trick Functions and the IGMP for broadcast services. The MCFs role is to control the flow of multimedia content to the user equipment with the actual delivery of the content performed by the Media Delivery Function (MDF).

The Network Attachment Subsystem (NASS), is responsible for the provisioning of the configuration parameters of the UE (such as IP address using Dynamic Host Configuration Protocol (DHCP)). It is also in control of the user authentication prior or during the allocation of the UE addresses, network access and configuration.

3.5 IPTV in IMS Requirements

The 3GPP efforts to standardize the IMS as a NGN architecture also provides a general overview of the functionalities that the IPTV architecture should support. The requirements are separated into three main groups: User, Network and Services [4]:

- User Perspective: The customer and user expects IPTV to provide the traditional television service in addition to: VoD, e-learning, personalized content and interactivity. Being all-IP based services, they may be used as building blocks for new services such as instant multimedia messaging, online games and conferencing.
- Network Perspective: As for IMS, the IPTV architecture should support and seamlessly integrate various access networks, such as wireless and fixed networks. In order to support scalable video distribution services, the network should support multicast and broadcast.
- Services Perspective: The service must support billing functions, such as subscriber based online and offline charging, pay per view mechanisms and content and service charging. The IPTV architecture should also provide setup and modification of services according to users preferences and capabilities, as well as adapting and monitoring sessions. Finally, the NGN architecture must provide service discovery and it consumed, the consequent resource reservation.

The ITU's NGN architecture has also presented requirements for the deployment of IPTV services and are separated in six areas [12], they are:

1. IPTV network architecture
2. QoS and performance
3. Security and content protection
4. Network and control
5. End and middleware systems
6. Public service

3.5.1 IPTV Network Architectural Aspects

IPTV services shall be integrated into existing services (such as Plain Old Telephone Service (POTS), VoIP and video conferencing) allowing all services to be available under one unified platform. This implies that an interface between administrative domains should exist.

In order to supply the IPTV service over heterogeneous environments, dynamic network reaction to changes should be implemented. This is especially relevant for wireless networks due to their dynamic characteristics.

As IPTV systems can amount to a large number of network devices, the ability for remote network management of service elements should also be available.

Finally, the IPTV system should allow private entities (such as end users) to act as content-providers, in order to allow content sharing.

3.5.2 QoS and Performance Aspects

The architecture should provide mechanisms to guarantee IPTV service as defined by the Service Level Agreement (SLA) provided by the operator. In order to provide satisfactory end user experience, this may range from a given minimum supported bitrate to a certain service availability.

The network must provide the capability to monitor the QoS and may support dynamic adjustment of QoS/QoE parameters. To manage these parameters the network operators may accommodate a QoS management system that integrates with other NGN services.

The IPTV architecture shall provide service differentiation mechanisms by assigning packet priority based on the provided service. Such methods include identification and marking, policing and conditioning as well as scheduling and discarding the packets. Dynamic traffic load balancing should also be supported as to adjust to instant network conditions at any given time.

3.5.3 Security and Content Protection Aspects

To prevent piracy of content, Digital Rights Management (DRM) on content and secure storage shall be provided by the IPTV architectures security mechanisms. Authorization and authentication of both user and content shall also be provided by the system before service delivery, in order to guarantee access to subscribed users only and to authenticate the source of the content.

3.5.4 Network and Control Aspects

From the network point of view, service providers shall be able to use unicast and multicast transmission and bi-directional communications for their services. The network should also provide PVR functions by using either the Service Providers networks or end-user's devices. It shall also support Internet Protocol Version 4 (IPv4) and Internet Protocol Version 6 (IPv6) protocols through static and dynamic address allocation schemes.

3.5.5 End Systems and Middleware Aspects

The IPTV service mediation system (middleware) shall be able to communicate with service providers for media control functions, with DRM systems and be able to provide mechanisms to allow end systems to negotiate service sessions.

The IPTV architecture should also support the capability for end-users to access different IPTV content on distinct terminal devices. These terminal devices must have the ability to easily identify television channels and its progress position.

3.5.6 Public Interest Aspects

In order to make usability facilitated and the IPTV service available to a broader population of users, some universal design principles should be followed when designing applications and equipment. For people with disabilities, closed captioning mechanisms should be provided by the architecture.

Regulatory information services and notifications should also be made available by the system (e.g., EPG guides). Finally, customers should be able to select IPTV network providers, service providers and content providers through mechanisms available in the IPTV architecture.

3.6 Benefits of a IMS IPTV Architecture

The service oriented aspects of the IMS architecture allow IPTV to leverage on many aspects to improve its services and overall quality.

IMS provides many of the underlying platforms needed to deploy a commercial IPTV service, such as authentication, security and network assurance mechanisms. This allows future IMS enabled IPTV applications to focus solely on providing television over IP. Additionally, as IMS is agnostic to network topology and technology, operators can deploy services over a wide array of independent operating domains and access technologies.

With the integration of the service within the IMS core, combining IPTV with other multimedia services such as VoIP and IM is simplified. All these services can share user preferences and properties as well as unified billing, delivering true triple or quadruple-play offers. Associated to the agnostic nature of the architecture, these unified services are no longer limited to a single technology.

3.7 Conclusion

IMS is, by design, a point-to-point architecture, while IPTV services are typically point-to-multipoint in nature. So, by a design perspective, broadcast service should prove to be a greater

challenge than VoD service in a IMS environment [59]. Existing multimedia signaling protocols and IMS signaling protocols need also to be adapted to provide interoperability and support for the various functions within the IMS architecture. Finally, the traditional client and server streaming architecture has to be adjusted to the more complex structure of several Entity Functions and logical components that make up the IMS architecture.

Despite these integration issues, the benefits that the architecture provides are enough to encourage the development of a IMS based implementation of the IPTV service. The adaptability to the various access technologies and the possibility to provide seamless integration of legacy and new services, enables network operators to leverage existing networks and services to new and existing clients. Additionally, the user personalization of services that IMS allows can truly change the way users make use of television services.

4

Architecture

Contents

4.1 Introduction

This chapter describes the architecture of the SIP based IPTV system with focus on the IPTV Application Server developed within the scope of the European project My eDirector 2012 [7] and respecting the requirements specified by a Portuguese 3G CDMA2000 mobile network operator. The first part details the functional and non-functional requirements that the AS design must follow. The following part details the two main modes of operation for the AS, each mode providing a separate service (VoD and Live Broadcasting). The third part describes the proposed AS SIP signaling and control structure and some of the limitations of the design. The Final part describes the connection of the AS in IMS environments.

4.2 Functional and Non-functional Requirements

The features and requirements for the AS prototype are the following:

1. **SIP Signaling**: the signaling used between AS and client must be accomplished with the SIP, while maintaining the standard SIP messages and attributes;

2. **Network agnostic**: the AS shall provide service to clients on several access technologies, such as Ethernet LAN, DSL, CDMA2000, Universal Mobile Telephone System (UMTS) or WiFi;

3. **Client agnostic**: through the use of encoding standards and parameters, the multimedia content must be adaptable to various UE, ranging from desktop computers to cellular mobile devices;

4. **Seamless quality transition**: when signaled, the AS must change streamed multimedia parameters (such as bitrate, codecs, among others) without the end user being aware that a change has occurred;

5. **MPEG-4 AVC**: The default supported video encoding standard shall be the MPEG-4 AVC standard, commonly known as H.264. Other codecs should also be supported to maximize the scope of compatibility with older and less powerful UE;

6. **IMS compatibility**: the AS architecture shall be modular in order to support connection to an IMS environment with functions mappable to corresponding parts and Functional Entities of the IMS architecture;

7. **Scalable and Reliable**: the AS should use hardware and network resources efficiently while performing its required functions under a variety of conditions.

The design, implementation and testing of the AS prototype relies on the design and implementation of the corresponding IPTV Client, developed in a parallel project [62]. The IPTV client project objectives are, among others, the design of an end client that can interact with the IPTV AS, analyze network conditions and provide proof-of-concept of the requirements.

4.3 Architectural Design

The goals for the IPTV AS design were to surpass the functional requirements specified. In order to accomplish this the IPTV AS main functionalities were focused on providing service establishment, control and termination. The architecture follows a modular design that facilitates the implementation of new functionalities. An example of such future functionalities might be the addition of a context-aware module to explicitly change video streams based on content or Customer preferences. The IPTV AS design also caters for scalability in terms of both concurrent client load and multimedia encoding process distribution.

The architecture that was developed for the IPTV AS includes some extra functions required to simulate the Control Plane functions of the IMS environment. This option was necessary in order to test the system without a real IMS environment.

4.3.1 Overview

As earlier discussed, the IPTV Application Server is aimed to be deployable in a IMS network core. As such, streamed content is supplied by Content Providers and its transmission to the Customer is assured by Network Providers.

The scalability requirement for this AS prototype consists on providing service for several clients, each with different service parameters. The AS tackles for load distribution by spawning a process for each IPTV client that establishes a session, as exemplified in the diagram of figure 4.1. Each process handles all session control between server and client, and terminates after the client-server session is closed. This method of process spawning allows the implementation of clusters, by distributing processes over several machines, dramatically increasing scalability [63].

The multimedia encoding of video streams is another scalability issue as it corresponds to resource heavy processes. As such, the encoding process distribution of the AS is done in two scenarios, a Per-client encoding stream and a General Purpose encoding stream.

4.3.1.A Per-client Stream

The per-client stream provides high adaptability to client requests but is a very high resource consuming process. For N clients connected to the AS, there are N encoding processes running. This solution allows a wide set of combinations for service attributes and client control,

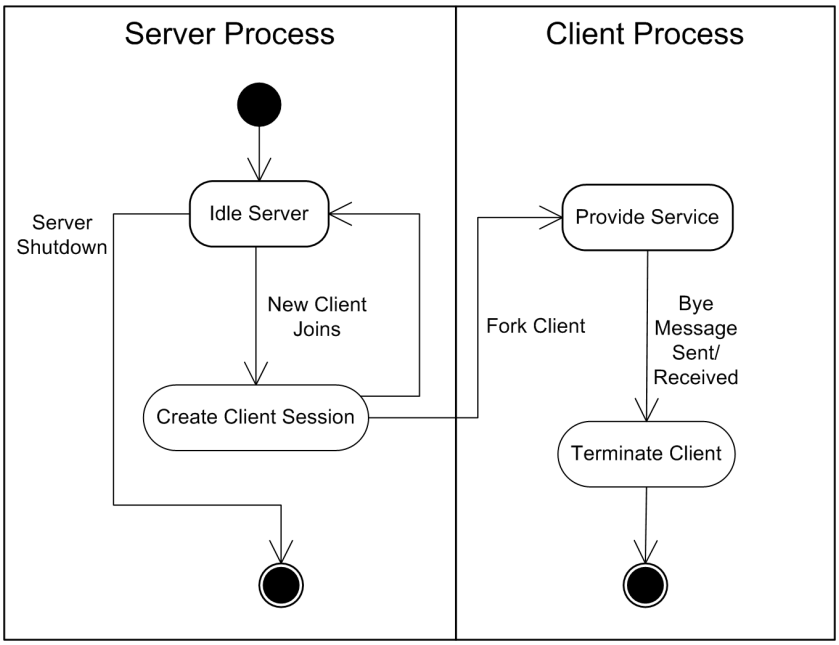


Figure 4.1: IPTV Application Server spawning a new client instance

freely defined by each client process (such as multimedia codecs, frame rates, group-of-picture sizes, etc.). Figure 4.2 illustrates the Per-client stream mode. This Per-client encoding stream is

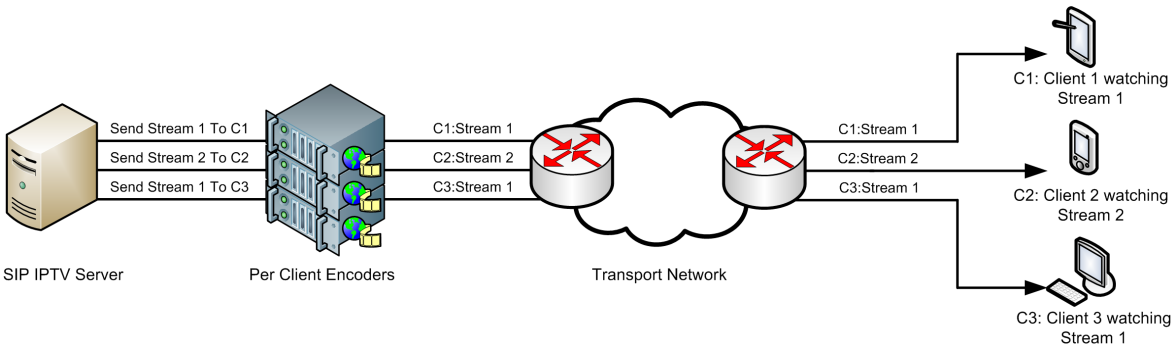


Figure 4.2: Per-client stream interaction with clients

appropriate to VoD content transmission.

4.3.1.B General Purpose Stream

The General Purpose approach provides high scalability but at a cost of a smaller set of available options for service attributes and client control. With this fixed amount of encoding processes are running at any given time and clients have to select one of the encoding streams from the available pool. Figure 4.3 illustrates the General Purpose stream mode. This scenario is well suited for live content transmission as each stream is being watched simultaneously by one or more clients. From a implementation perspective each stream must be predetermined on the

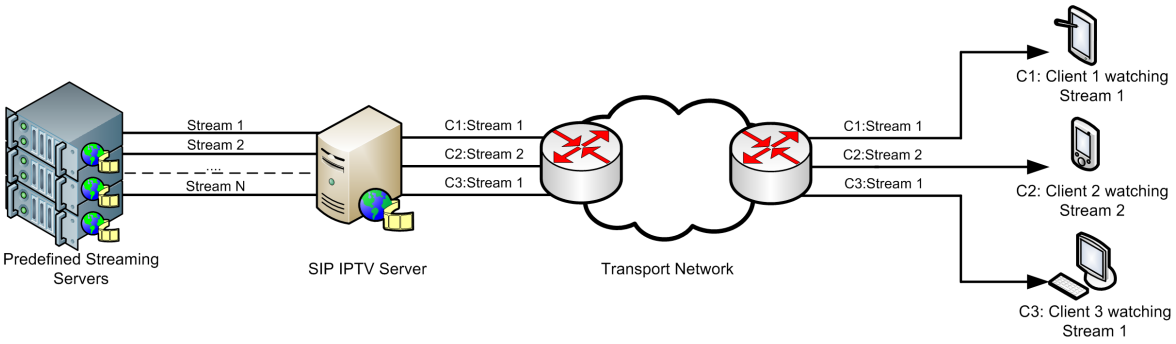


Figure 4.3: General purpose stream with dedicated encoding machines

AS prior to any client request.

4.3.2 IPTV AS Modules

The main components of the IPTV AS are the Server Block and the Session Block as illustrated in Figure 4.4.

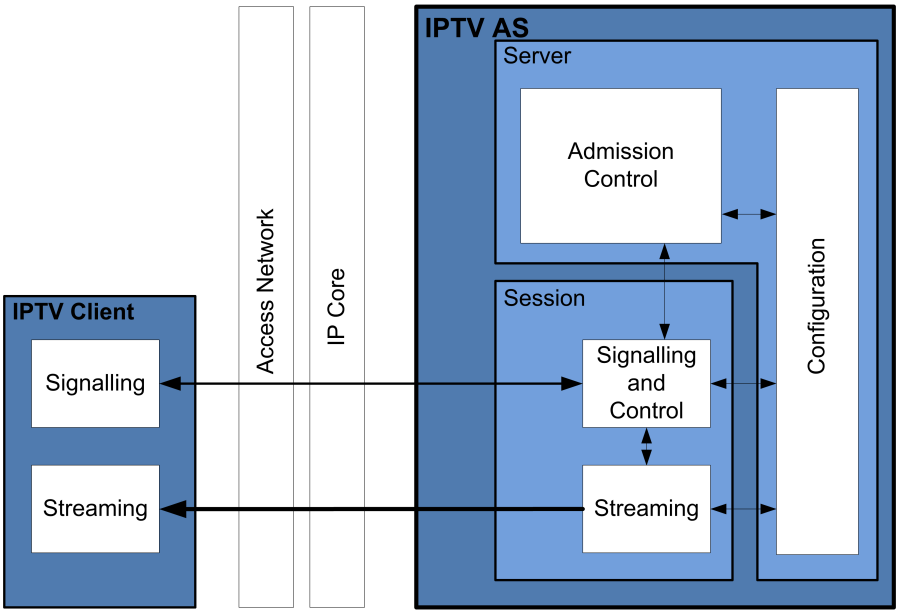


Figure 4.4: IPTV Application Server architectural blocks, modules and interaction with the client and network

4.3.2.A Server Block

The server block provides server configuration and control and acts as primary anchor for connecting clients. This block is composed of the following sub-blocks:

Configuration Block This sub-block is responsible for the configuration of server properties. For the General Purpose stream, this block provides clients with the attributes of the available streams (see 4.3.1.B). For the Per-client stream, this block provides available streaming attributes for each client processes.

Admission Control Block This sub-block is responsible for the creation of a new client session whenever a new client connection request arrives. These new sessions are then handled by the Session Block.

4.3.2.B Session Block

The Session Block, is a Per-client instance that provides service to the clients. This block is composed of the following sub-blocks:

Signaling and Control Block This sub-block is responsible for the session setup, control and subsequent tear-down. This sub-block provides the signaling and control for all IPTV Trick Functions. Additionally, the multimedia adaptation requests from clients are processed in this block, which in turn, signals the Streaming Block to perform the updates.

Streaming Block This sub-block provides the streaming functionalities necessary for the acquisition and transmission of multimedia content and implementation of the Trick Functions. If a Per-client encoding process is needed, it encodes and streams the content according to specifications defined by the session parameters. Client request updates are processed in this block (seamless transitions between quality levels or content stream changes).

4.4 Signaling Structure

The service signaling structure of the IPTV system prototype is based only on SIP for both the session and media control.

4.4.1 Overview

Within an IMS based IPTV environment, this model for service control plays a major role on advanced service capabilities by introducing the set of Trick Functions to the user.

There is a clear advantage on this model when compared to hybrid models of SIP in combination with the RTSP for both the signaling and media control: the simplicity of integration of the IPTV services with other session oriented services based on SIP, for example, VoIP.

Another advantage is the reuse of functions and attributes already available on SIP and SDP to implement the media control functions, avoiding the design of extensions to SIP. For that purpose the SIP UPDATE [64] message with different attribute values was used.

As table 4.1 shows, the SIP protocol also has fewer stages than the RTSP protocol as well as the hybrid SIP-RTSP protocol. This optimization is especially important for environments where packet loss rates are high and bandwidth is limited, as it minimizes the need of retransmission of packets and number of signaling stages.

	RTSP		SIP	
	Messages Transfered	Message Size (KB)	Messages Transfered	Message Size (KB)
Session Establishment	10	3	4	1,7
Channel Change	12	3,6	3	1,3
Quality Change	12	3,6	3	1,3
Pause & Play	2	0,4	3	1,2
Session Termination	2	0,5	2	0,8

Table 4.1: Comparison between RTSP and SIP messages

Figure 4.5 shows the SIP Signaling Structure used on the IPTV architecture.

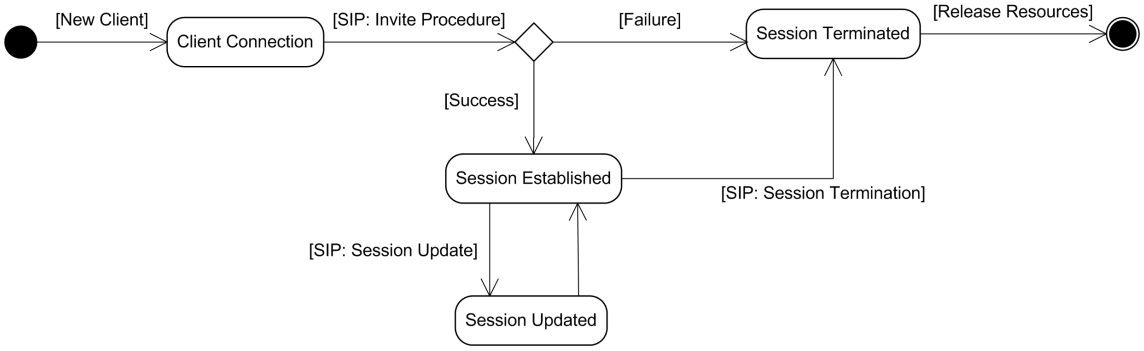


Figure 4.5: SIP signaling overview

The main sections in the signaling structure are: session establishment, session control, session quality and session tear-down (see annex A). The various signaling messages can be seen in figure 4.6

4.4.2 Session Establishment

As the name implies, this section is responsible for session setup. The general procedure of the protocol is similar to the one used in VoIP applications. The signaling messages transmitted during session setup are depicted in figure 4.7.

The IPTV Client sends an SIP INVITE message to the IPTV service, which is forwarded by the core IMS CSCF to the IPTV AS (see Figure 4.7). This SIP INVITE message contains the SDP

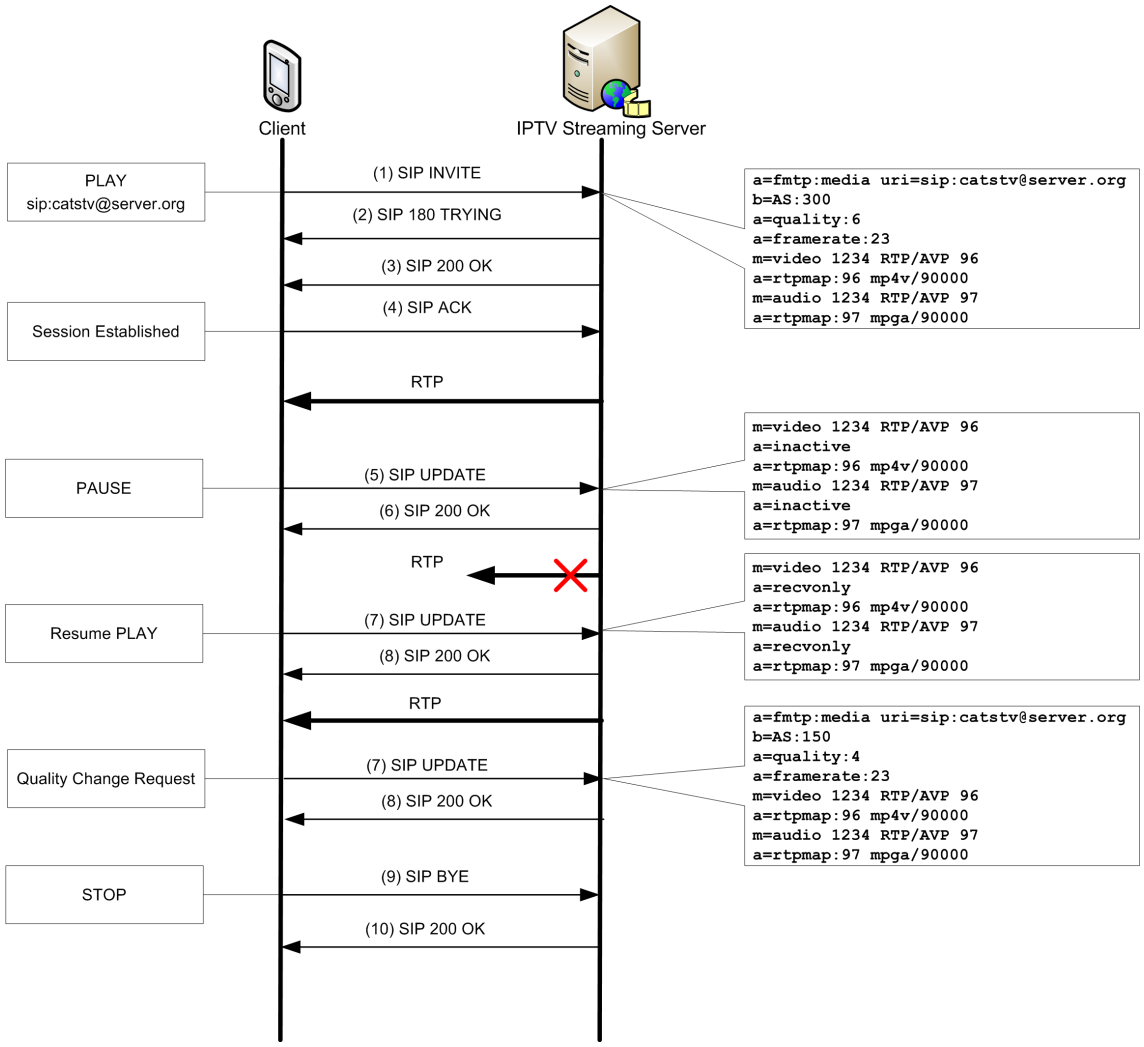


Figure 4.6: Server and client SIP message protocol

initial session parameters, such as frame rates and codecs to be used, as well as the URI of the selected content. This URI corresponds to a multimedia content, such as a file located in a video repository or a live multimedia transmission. The IPTV AS replies with a SIP 180 Trying response and checks whether the resource is available and if session parameters can be sustained. If so, the IPTV AS proceeds with the reservation of the resources and sends a SIP 200 OK message to inform the IPTV Client that the session is established and the RTP flow is started. In case the resource is unavailable (due to a wrong resource identifier, deleted content or unsupported codec), the server sends one of the SIP 400 failure responses and the session is finished with a SIP BYE message.

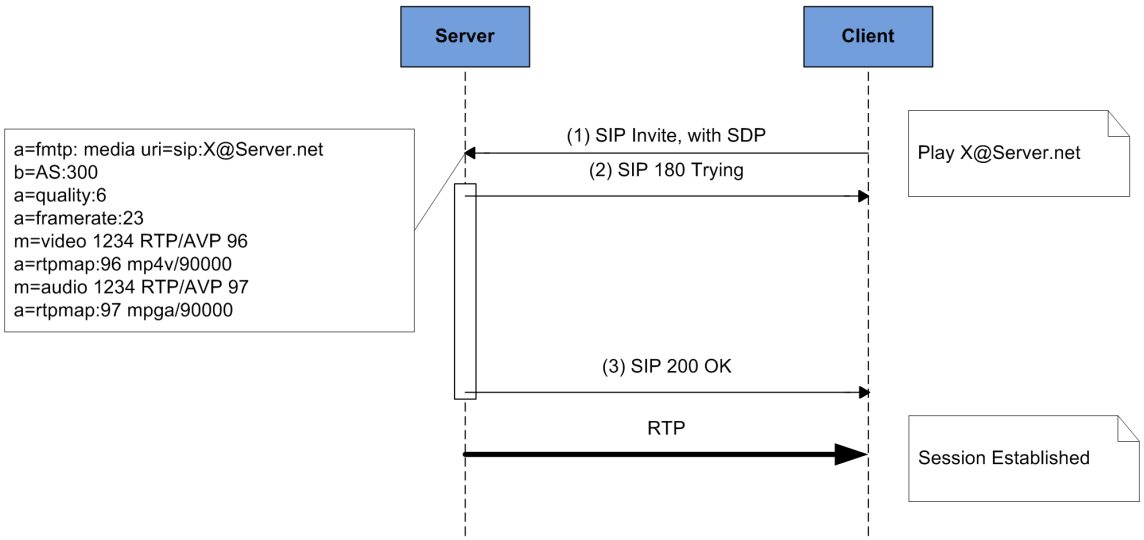


Figure 4.7: SIP messages exchanged during session setup

4.4.3 Session Control

Session control allows the server to change session attributes without impacting the state of the SIP dialog between the client and server. This is accomplished with SIP UPDATE messages. When issued from the server side, an UPDATE allows the callee (client) to update session attributes (such as available encoding options). From the client side it implements Trick Functions to update the state of the media being received (such as playing or paused). The SIP Update should contain a SDP message [64]. The content of the session Control UPDATE message is shown in Figure 4.8.

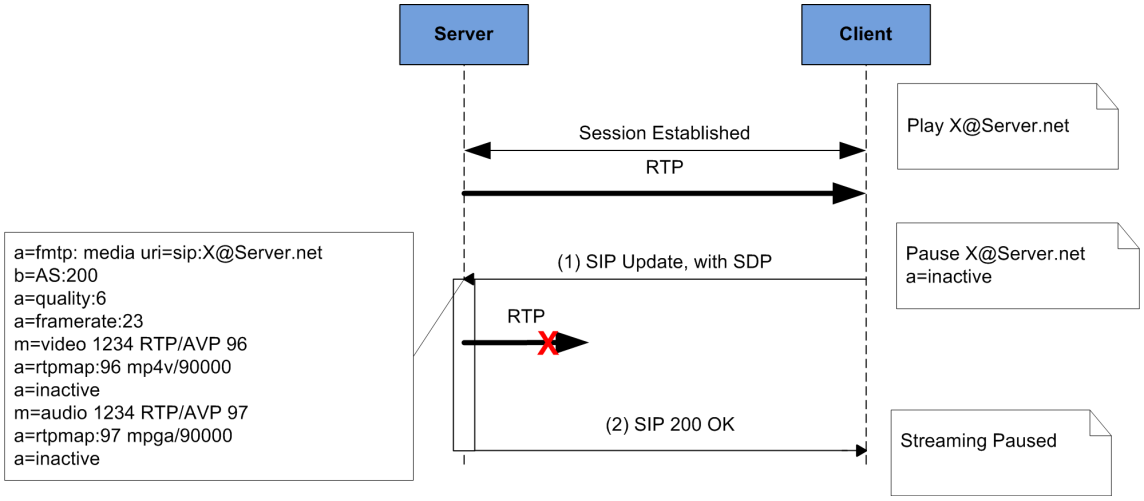


Figure 4.8: SIP messages transferred during a session pause

Session Control can be accomplished during or after the session establishment phase. For a PAUSE the client sends a SIP UPDATE message with the attribute a=inactive for the RTP/AVP

streams. The server attempts to apply the requested updates to the session parameters and, if successful, responds to the client with a SIP 200 OK message. If the parameters cannot be applied, the server replies with a non-200 SIP message, such as 415 Unsupported Media Type or 420 Bad Extension messages. To resume the session the client sends another SIP UPDATE message with the attribute ”a=recvonly” for the RTP/AVP streams.

4.4.4 Session Quality Update

The IPTV AS does not monitor the session quality, instead this function is delegated to the IPTV Client. The dynamic QoS adaptation is signaled by the IPTV Client using the standard SIP UPDATE messages. Figure 4.9 depicts a quality change during a session.

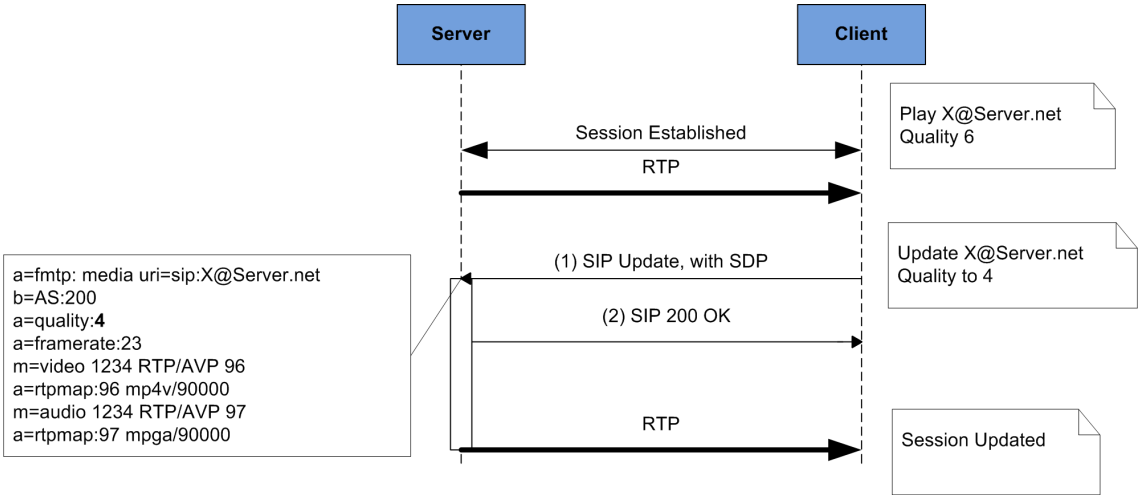


Figure 4.9: SIP messages transferred during a session quality update

The Quality UPDATE message is sent from the IPTV Client with a SDP quality attribute ”a=quality:xx”, the quality values range from 0 to 10. The IPTV AS interprets the SIP message and proceeds to adapts the multimedia stream by reducing the target bit-rate, without interrupting the ongoing session. There is a visual impact when the stream is updated, but the session is not compromised. This method was based on the QoS negotiation for IPTV using SIP as described in [65].

4.4.5 Session Teardown

The session tear-down is a two stage procedure. The requesting party, which can be either the client or server, sends a SIP BYE message that is accepted or rejected by the other party. If the receiving party decides to end the session, all of its resources dedicated to the session should be terminated. The content of the session Tear-down BYE message is shown in Figure 4.10.

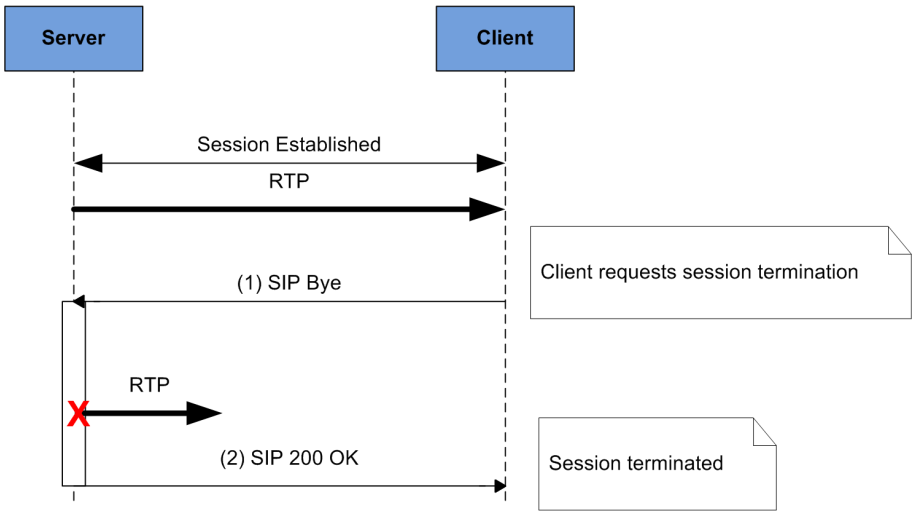


Figure 4.10: SIP session termination procedure on clients request

On the requesting side, after receiving the BYE response, all resources dedicated to the session should also terminate. If the BYE response is rejected, the originating party should retry to end the session by sending another BYE message.

4.5 Integration in IMS

As previously stated the IPTV AS is aimed to connect to an IMS core. Most of the requirements presented in section 3.5 were addressed in the IPTV AS prototype architecture. However, because a real IMS environment was not available, the developed AS prototype includes functionalities that simulate the Control Plane functions of the IMS environment.

Figure 4.11 demonstrates the relation between the developed IPTV AS components and the Functional Entities of the IPTV architecture proposed by European Telecommunications Standards Institute (ETSI) [2].

By controlling and delegating each session to individual processes, the IPTV AS Admission Control block is able to implement a session routing mechanism and provides rudimentary user authentication and charging functions. This block also shares the IPTV service location functions provided by the Service Discovery Functions (SDF) and some of the accounting and authentication functions present in the Service Control Functions (SCF). Some of the core IMS functions are also implemented in this AS block as it is responsible for the SIP message redirection to the session processes.

By providing available configuration parameters, the AS configuration block shares some functionality with the ETSI Service Selection Function (SSF) and the UPSF. The SSF and UPSF provide the end user system with a list of available streams based on client capabilities and preferences. These user preferences and capabilities are taken into account when they are specified

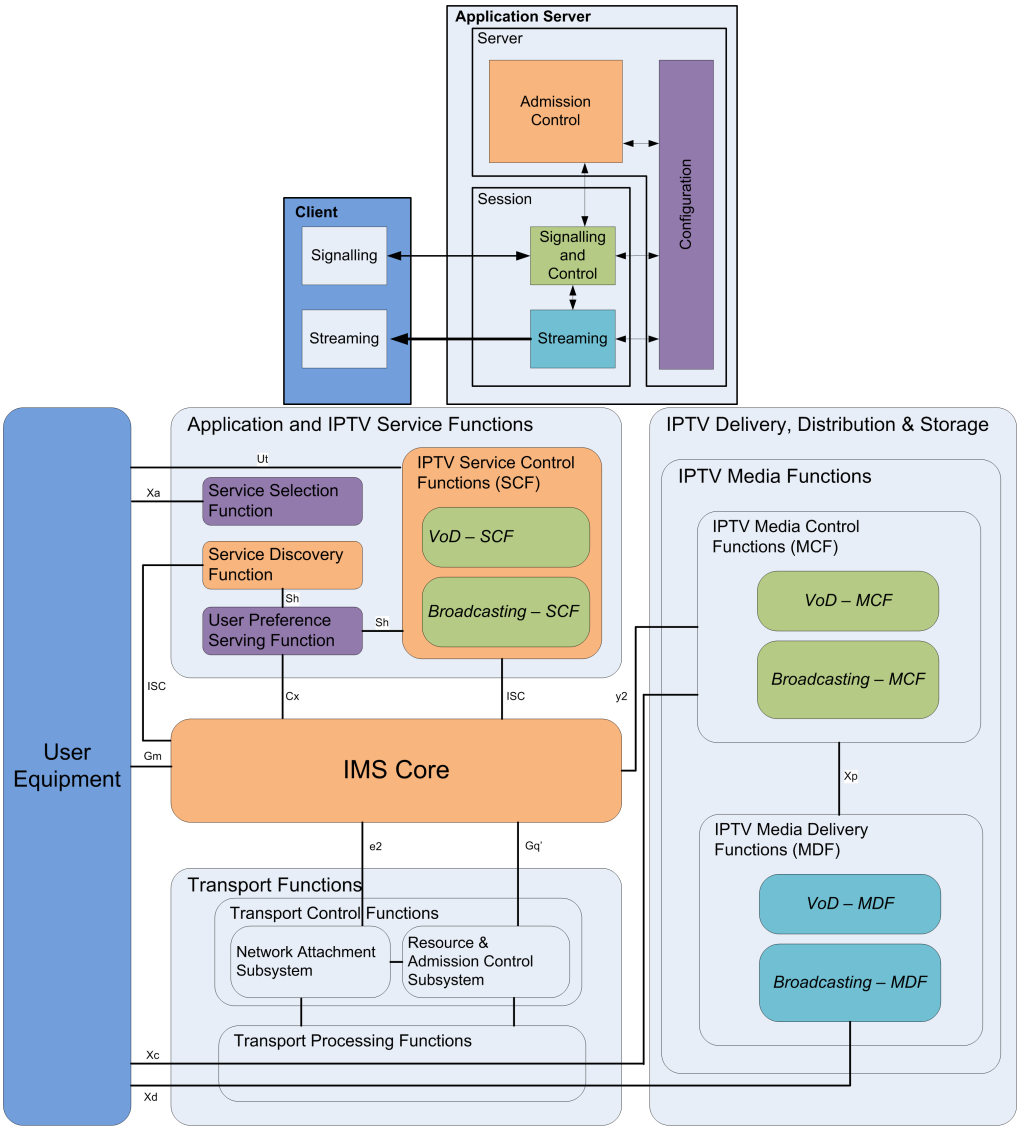


Figure 4.11: Application Server integration in the IP Multimedia Subsystem architectural framework

in the SIP Invite and Update messages. For implementation simplicity, the Configuration block does not implement an explicit Xa interface with the client, delegating the function to the signaling and control block.

As previously discussed, all session logic is present in the Signaling and Control block, and so most of the IMS functions are here implemented. Service session setup, control, update, teardown and media control (play, pause, stop, etc.) functionalities are also implemented in this block, they correspond to the IMS IPTV service control functions and the IPTV media control functions. Particularly, each application mode - Per-client and General Purpose - implement the two main IPTV functions, VoD and Broadcasting respectively. Finally, the multimedia streaming functions of the AS streaming block directly implements the IPTV media delivery functions as specified by the

ETSI IMS architecture.

4.5.1 Architecture Limitations

The architecture of the prototype provides a rich set of functionalities, but also a few shortcomings that may be overcome with future enhancements. The main disadvantage of the described architecture is the centralization of the IPTV Application Server, even with clustering, all content streams for clients are sent from a single cluster of machines. This is due to the Streaming block being attached to the Session block. This can be partially solved by delegating the Streaming blocks to independent media servers, but implies that the Signaling server must support a communication protocol between the two applications. Although not described, load balancing by means of additional IPTV Application Servers can be accomplished by redirecting clients upon the initial signaling. But this requires additional components at the network level for service virtualization. The IPTV AS prototype architecture was specifically tailored for live streaming. For VoD content streaming and multiple network characteristics, the real-time encoding of the Per-client process is not scalable. An alternative would be the use of Scalable Video Coding (SVC) [66].

4.6 Conclusion

The architecture of the prototype solution allows a hybrid approach to the two main services that IPTV provides, Live Broadcasting and Video on Demand. Using the Per-client approach it is possible to provide a fully customized set of attributes that allow an end user to tailor the stream to the device capabilities and user preferences. Using the General Purpose mode it is possible to deliver Live Broadcasting that seamlessly adapts to disparate transmission conditions and different types of user devices.

The modular design of the AS provides independence between each module and an architecture that allow its connection to the IMS framework. The modular design also allows the addition of several enhancements to the application server functionalities.

The proposed SIP signaling structure is not that different from other IMS enabled multimedia services, such as VoIP. It is important to retain that all the messages types used in the signaling structure are defined in the SIP and SDP standards.

5

Implementation

Contents

5.1 Introduction

This chapter describes the various stages that were followed during the development process for the implementation of the server prototype. A general overview of the programming language and libraries used to build the various modules is also detailed. The chapter concludes with consideration about the implementation limitations and suggests some possible solutions.

5.2 Development Process

The main objectives defined for the prototype developed, were to support streaming multimedia content over an IP network using SIP as a signaling and control protocol. The streaming requirements were those typical for Live video streaming and Video-on-Demand. Additionally, seamless transition between qualities of a stream was also required.

The development process can be separated into five stages, the initial requirement phase, the research phase, the architecture definition phase, the implementation phase and finally the validation and analyses phase.

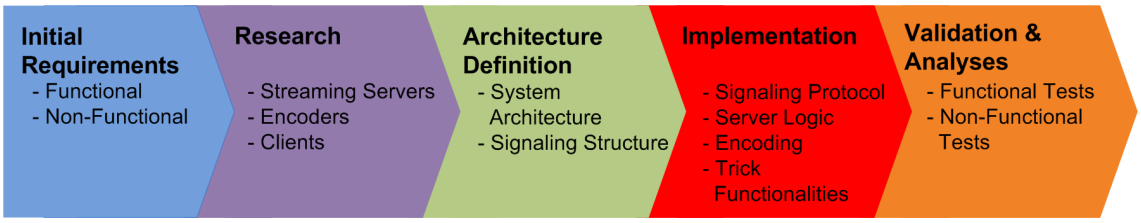


Figure 5.1: Five stage development process

The first phase corresponded to the specification of functional and non-functional requirements and the general planning of what was to be accomplished. The main functional requirements were video and audio streaming in live and VoD scenarios, integration in an IMS architecture and seamless transition between streaming parameters. The primary non-functional requirement was to provide a scalable and highly available architectural solution, capable of being deployed in a wireless operators network.

With the requirements defined, a research phase started, where various streaming servers and client implementations were tested, such as the Darwin Streaming Server, the VLC Server and the Helix DNA server. Quicktime and VLC Clients were studied for client applications. In this phase various encoding and decoding libraries were also analyzed, such as the FFMPEG project libraries and the x264 project.

The third phase of the project corresponded to the selection of the tools to be used and the definition of the prototype architecture. This architecture was defined in close collaboration with the IPTV client development project. In this stage the SIP signaling structure was defined as well

as the general architecture of the system.

After selecting the tools and defining the architecture, the next phase consisted on building the prototype for the Application Server. This was a gradual process that began with the SIP signaling structure and the base logic of the prototype without any content transmission, followed by the integration with the streaming and encoding functionalities and the application to a scenario where multiple clients could interact with the service. The final step of the implementation phase was the inclusion of features such as providing central configuration of server parameters and also optimization of the code.

The last phase corresponded to the validation of the prototype in a live network for conformity to requirements and to the testing and analyses of its capabilities and features.

5.3 Software Libraries

The application development environment used was the Ubuntu distribution of the Linux operating system [67], currently available for a large set of different devices, ranging from enterprise servers to small embedded systems. The programming language used was the C language, compliant with the Portable Operating System Interface for Unix (POSIX) standard[1]. The choice of this language was due to the portability, stability, reliability and speed factors that it provides, especially for the development of network enabled applications. Additionally, in order to maximize portability of the application to various platforms, only open-source and platform independent libraries were used.

The two main libraries used were LiboSIP [68] for SIP and SDP message generation and parsing and the libVLC library [69] for media streaming and encoding.

5.3.1 SIP Stack Implementation - liboSIP

LiboSIP is an open-source implementation of a SIP stack library, it provides a API that allows simplified SIP message generation and parsing [68]. Although liboSIP provides a platform for SIP parsing, it does not implement the SIP logic. Therefore the SIP signaling structure is part of the prototype logic and not handled by the oSIP library. This allows full control of message transmission throughout the signaling session.

The library also provides SDP message generation and parsing. This is used throughout the project to describe the multimedia session parameters. It is of the upmost importance to generate valid SDP messages because most of the Trick functions and quality adaptations are described using this protocol.

[1] POSIX - IEEE defined family of standards that specify a common application programming interface for UNIX systems

5.3.2 Multimedia Streaming and Encoding Library - libVLC

Based on the VLC application [70], libVLC supports interaction between external applications and the VLC functionalities [69]. Its usage in the prototype is primarily as a encoding and streaming platform. The libVLC library provides and API and so third party applications may use the features found in VLC. The available functions in the library are:

- VLC instantiation: the most powerful functionality, it allows instantiation of VLC processes with all the input options that the command line interface allows, such as transcoding options, streaming parameters and subtitle overlaying;
- Playlist manipulation: the basic Trick Functions are supported, such as play, pause and stop;
- Stream control and information: these functions allow multimedia information and control, i.e., getting input stream position and setting the position of the current playing item. These simple commands allow rewind and forward capabilities to be implemented in the application.

With these API functions, the server is capable of opening a wide array of media sources and encoding schemes. External RTSP streams, DVD sources, camera capturing and local files can be used as a source of video. The transcoding options also allow any codec supported by VLC to be used in the prototype and changed during runtime (for example, H.264 and MPEG-2 video coding).

Similarly, the streaming features that are used are those made available by the VLC application and can be also changed during runtime, for switching streaming sources and destinations or for IPv4 to IPv6 network addressing.

The prototype design aimed at providing a means to rapidly and easily add new features and this was achieved by using a modular design and by defining interfaces between each module.

5.4 Modules

To simplify the usage of the prototype, some support modules were added. An add-on module was implemented to allow additions of new features with minimum change to the overall architecture. Generic to all modules is a logging and output module to permit analyses of the program functions. The modules are depicted in Figure 5.2

5.4.1 Server Runtime

The server runtime implements all functionalities that are related to server setup and client management. When the server initiates, this module is run and follows the steps depicted in Figure 5.3:

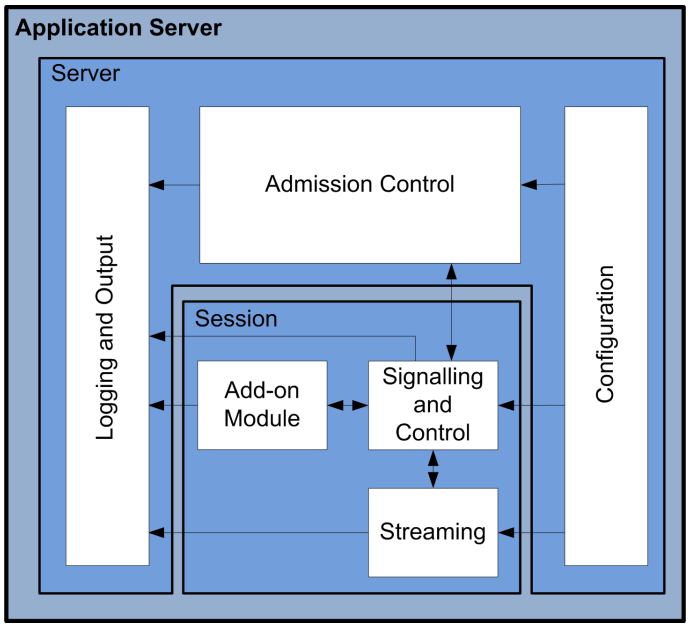


Figure 5.2: IPTV Application Server implementation modules

1. Configuration: The server proceeds to call the configuration module to get all server parameters;

2. Encoding setup: If the General Purpose mode is used, the server can create encoding processes that are used to receive content from various origins (RTSP, DVD, Files) and streams the content locally or to another prototype instance. If activated, the prototype also becomes a encoding server.

3. Listening point: The server then sets up a listening point that is configurable to wait for new client connections;

4. Client management: When a new client joins or when connected client leaves, the server runtime proceeds to the setup or tear-down for signaling sessions. After this step, the runtime returns to the listening point step to wait for new a client.

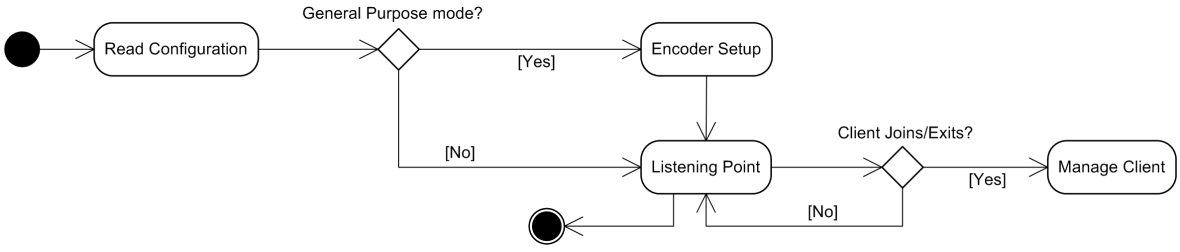


Figure 5.3: Server runtime process

5.4.2 Configuration Module

The server prototype allows a configuration file to be used, read upon server initiation. The configuration file allows changes to the servers main parameters without the need to recompile the server or edit any of the source code. The configuration file allows the following parameters to be set:

1. Listening point: the server port has a default value, but can be changed in the configuration file;

2. Video and audio codecs: If in the General Purpose mode the default video and audio codecs for encoding can be defined in the configuration file;

3. VoD content source: the location of the VoD content must be specified;

4. Channel definitions: the prototype allows aliases to be given to each channel provided by the General Purpose mode. Each alias is accompanied by a given quality value, and these two attributes define a stream to be sent to the client.

5.4.3 Signaling and Control Module

The signaling and control module is responsible for all interaction with clients. As such, the LiboSIP library is the only one used within this module. Figure 5.4 shows the signaling and control process.

In addition to session control and signaling the multimedia functionalities provided by the streaming module, the trick functionalities such as play and pause are also controlled from this module. The various SIP Update messages for that purpose are directly mapped to the streaming modules corresponding functions. For example, a SIP Update message with both multimedia attributes marked as ”inactive” triggers the pause streaming function, whereas when marked as ”recvonly” triggers the playback function to resume the streaming.

5.4.4 Streaming Module

The main objective of the streaming module is to provide a common interface for the multimedia API (the libVLC library) and other prototype modules, namely the Signaling and Control Module. This allows any new multimedia application or library to be used without changing the prototype structure.

The module provides the following set of features:

1. Multimedia instance management: this allows the streaming or encoding instance to be setup or destroyed. This is called during session setup, teardown and quality transitions;

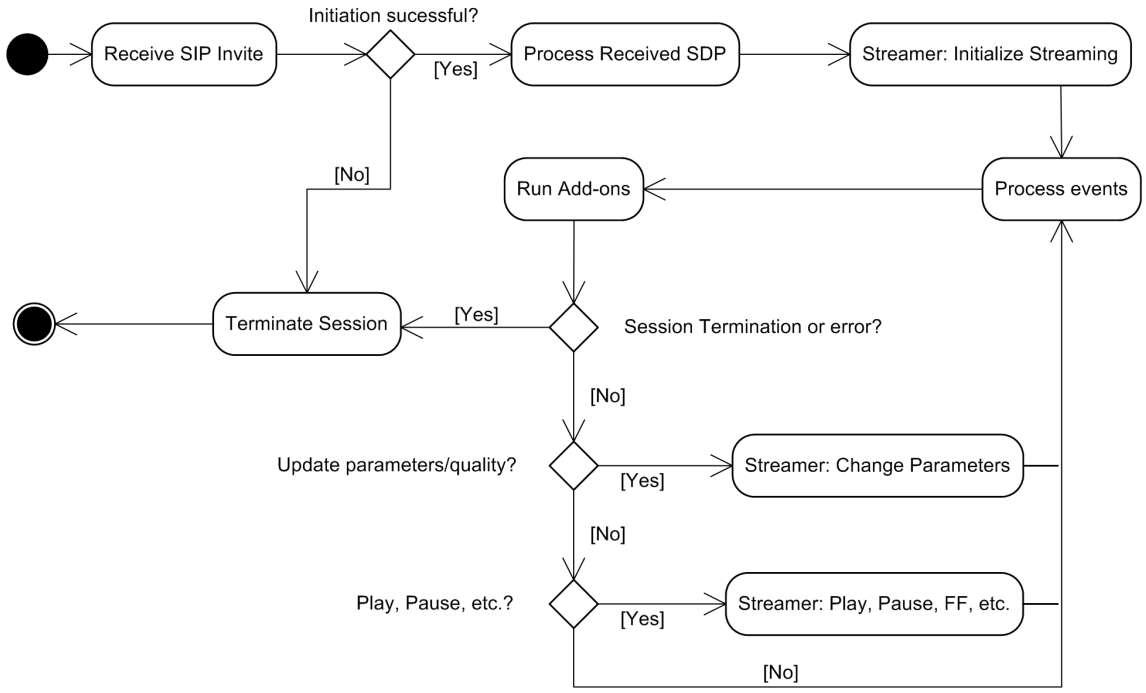


Figure 5.4: Signaling and control process

2. Multimedia parameter definition: the parameters to be sent to libVLC are handled through this section of functions allowing realtime changes to the streamed content;

3. Trick Functions: the play, pause, stop, seek and set position functions provided by libVLC are handled through a set of internal wrappers. These wrappers provide the required transparency between the prototype and libVLC.

5.4.5 Logging and Output Module

This module provides a central set of interfaces that allow the remaining modules to record server events. These records can be sent to the terminal output or written to log files.

The event attributes that can be logged are:

- Process ID: the identifier of the process that generated the log of the event;
- Timestamp: the date and time that the event occurred;
- Type of event: the type of event that was logged: SIP signaling related, streaming related or server related;
- Message: the provided message describes the event.

5.4.6 Add-on Module

The add-on module allows new functionalities to be implemented without the need to change the prototype internal code. This module is run after a specific timeout (typically 100ms) or when any event occurs. It provides access to a structure that defines the particular session. This structure contains the streaming configuration, client address, client port, SIP session variables and other session specific parameters.

5.5 Implementation Limitations

There were two main limitations found with the implementation of the prototype related to the transmission of the signaling messages and the transition between content qualities.

5.5.1 SIP Signaling

The liboSIP library solely provides the parsing and creation of SIP and SDP messages, and as aforementioned, the protocol itself is implemented by the application. Because the application server design goals were on a basis of proof-of-concept, only the SIP signaling procedures for core functionalities were implemented. As such, session redirection, authentication and security were not implemented. All content is transmitted without encryption or any similar security mechanism.

5.5.2 Transition Between Qualities

When a client requires a quality change in the Per-client mode, the application server is required to switch between the current stream and a new stream. In order for this to work seamlessly, the first content frame of the new stream should be transmitted immediately after the end of transmission of the old stream. LibVLC has a limitation in the seeking mechanism whereby to seek within a file, the file has to be already playing. Due to this, in the Per-client mode, the application server needs to open and play the video and then advance to the position of the video that is being watched. In the current implementation, these steps are somehow still noticeable in the switching process as the video returns to the initial frame then moves to the correct playing position within the stream.

The General Purpose mode also has issues with the quality change mechanism. To achieve a seamless transition with live content, the application server starts sending the new stream - Quality 2 - while still briefly transmitting the old stream - Quality 1 - as shown in figure 5.5. This means that both streams are typically being transmitted during half of the Round Trip Time (RTT). When the client receives Quality 2 it immediately starts playing, while at the same time ignoring

Figure 5.5: Transition between qualities

the Quality 1 stream. In order for this mechanism to work properly, the transmission of streams has to be carefully synchronized. As to minimize visual artifacts at the client, the transition should be carried out with the use of an Intra frame. Due to the limited functionality of the libVLC API, this detailed control of frames during transmission is not achievable meaning that it is not possible to specify at what frame the transition shall occur. With this limitation the content has to be encoded with a small GOP size, i.e. the number of frames between two Intra frames must be small. This solution increases the bitrate of the streamed content as smaller GOP sizes translate to higher encoded bitrates.

5.6 Conclusion

The successful implementation of this prototype was highly dependent on the correct definition of the overall architecture. In this chapter the development process was discussed, from the research to the evaluation phase, with each step described and detailed.

The libraries used to implement the application server prototype were also described. LiboSIP was used for parsing and generation of SIP and SDP messages and the libVLC API was integrated to implement the transcoding and streaming functions.

Finally, the various core and support modules of the IPTV Application Server were described, focused on the logical architecture. The chapter concludes with the limitations that affect the server functionalities.

6

Evaluation Tests

Contents

6.1 Introduction

This chapter describes the evaluation tests of the IPTV AS. The objectives of these tests are detailed, followed by a brief description of the test environment and metrics used. The test results are then described and studied and final conclusions are made.

6.2 Evaluation Test Objectives

The goals while testing the prototype system were to evaluate the overall scalability of the solution as well as the functionalities it provides. The tests provide information on the quality of the IPTV Application Server as well as the service provided to the various clients. In order to assess the overall nature of the AS, the tests are separated into functional and system tests.

The functional tests evaluate the services that the AS provides to the clients. These tests analyze various AS functionalities, such as session setup, teardown, trick functions (play, pause and stop), quality and content transitions.

The system tests - or non-functional tests - were designed to evaluate the AS qualities, such as the reliability and scalability it provides. As the AS developed is only a prototype, security measures are not included in the tests. The system evaluation is separated in Per-client and General Purpose tests, as to study each AS mode. Each test implements both MPEG-4 video codecs (MPEG-4 AVC and MPEG-4 Visual) as to evaluate the overall response of the system.

6.3 Test Environment

A campus High-Speed LAN and a wireless CDMA2000 network were used throughout the tests (see Figure 6.1). The encoder was set up with 10 different quality levels for each stream, ranging from 16 Kbps to 292 Kbps. The functionalities under test included:

- Session establishment and tear-down;
- Trick Functions (e.g., play, pause);
- Dynamic QoS/QoE parameter adaptation for both Per-client and General Purpose streams.

The IPTV AS scalability was evaluated measuring CPU load and memory usage. In the general purpose mode, two of the client machines were used as transcoders to simulate the transmission of live content to the application server.

Scalability tests were conducted using more than one client instance on each client machine. On the server machine a single instance of the prototype was run with solely relevant operating system processes.

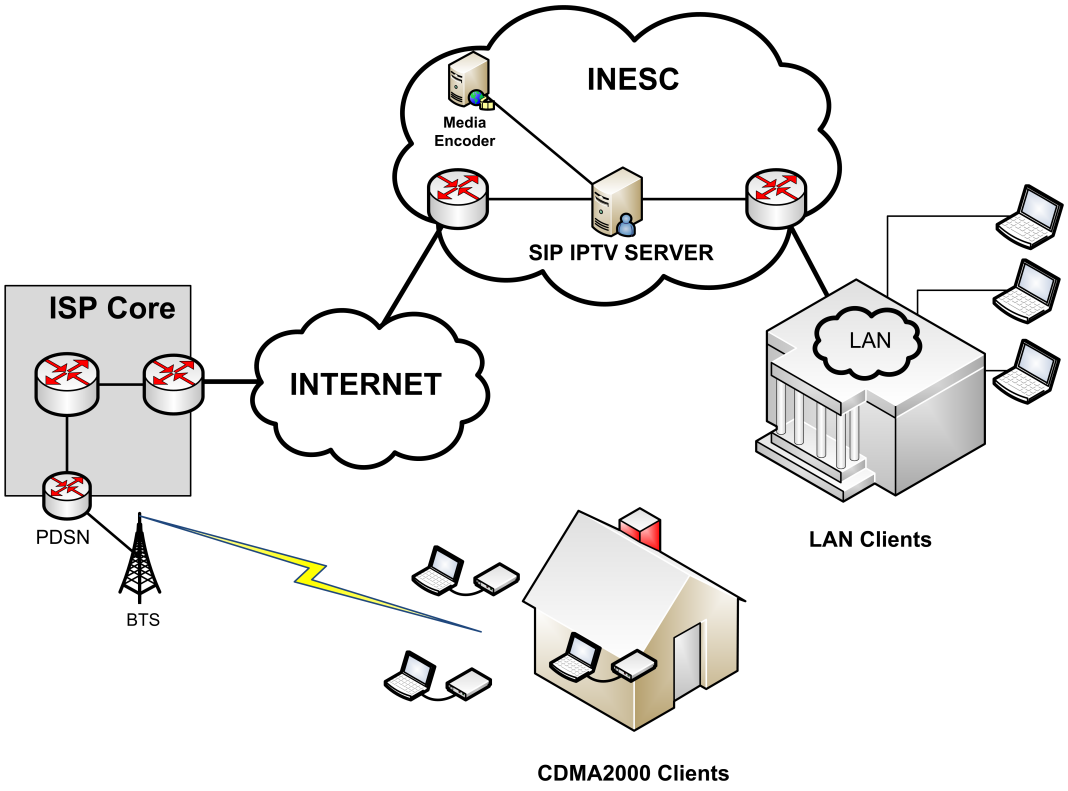


Figure 6.1: Application server test environment layout

6.3.1 Server Metrics

The collection of statistic data on the server was done using the tool CACTI [71]. CACTI provides a front-end to a complete set of data gathering and graphing tools, allowing the treatment of statistics of various machines on a network. The CACTI server was installed on one of the client computers and configured to poll the application server machine every ten seconds. The resulting values were then averaged over a span of three minutes and presented to the applications GUI.

6.4 Functional Tests

In order to run the Functional Testes the IPTV AS was set for both Per-client and General Purpose streams and only one IPTV Client was used.

The Functional Tests results are common for both Per Client and General Purpose stream modes. The following test analyze the transition between content qualities, focusing on the seamless switching of video streams.

6.4.1 Signaling Protocol

The campus network test results for signaling in Table 6.1 show that session setup, teardown, channel change, pause and quality change are nearly within the suggested maximum limits specified by the Broadband Forum triple-play QoE service requirement [72], as seen in table 6.1 meaning that the system adapts well to higher quality levels due to the low congestion and available bandwidth.

	Session Startup (s)	Session Teardown (s)	Channel Change (s)	Quality Change (s)	Pause Time (s)
Average	2,94	0,64	2,51	1,52	0,29
Deviation	0,49	0,39	0,48	0,41	0,16
Maximum	4,00	1,70	3,20	2,00	0,60
Minimum	2,22	0,20	2,00	0,90	0,10
TR-126 Limits	2,00		2,00		0,20

Table 6.1: Signaling execution times in a LAN

As Table 6.2 shows, the results on the CDMA2000 network were very satisfactory considering the characteristics of the radio link due to the spiky nature of the CDMA2000 Radio Link Protocol (RLP) processing. It is important to note that the suggested maximum limits were defined for a general broadband connection. Additionally the values could be reduced with the IPTV AS prototype relocation to the CDMA2000 core. This extra latency overhead negatively impacts the results as all signaling packets have to cross public networks between the campus LAN and the mobile network.

	Session Startup (s)	Session Teardown (s)	Channel Change (s)	Quality Change (s)	Pause Time (s)
Average	3,74	1,06	3,79	3,43	0,36
Deviation	0,77	0,82	0,71	0,77	0,21
Maximum	6,00	3,00	5,60	5,00	0,70
Minimum	3,10	0,40	3,00	2,10	0,10
TR-126 Limits	2,00		2,00		0,20

Table 6.2: Signaling execution times in a wireless CDMA2000 network

These tests show that the usage of SIP as a session setup and control protocol is feasible under very distinct network conditions. Although the signaling values are higher than the limits specified by the broadband forum, tests with users show that these times are acceptable, especially considering the characteristics of a wireless mobile network.

6.4.2 Transition of Content Quality

The transition of content quality during sessions is aimed to be as seamless as possible. To accomplish this, the AS uses a fast switch between two streams of content with different quality values. To achieve this result, a compromise between GOP size and quality had to be made. The smaller the GOP size the higher the amount of Intra frames (I-frames) present in the stream. This implies that all non-intra frames have to be coded with higher bitrate constraints, degrading its quality. Additionally, a smaller GOP size allows a more robust stream to be sent, as lost frames and error recovery times are shorter. The content used is encoded using a fixed 12 frame GOP size for MPEG-4 Visual and a maximum I-frame interval of 12 frames for H.264 (due to the dynamic GOP structure). Considering a static frame-rate of 24 frames per second, a I-frame will be transmitted at least once every 500 ms with these settings. In the developed prototype, only fixed size GOP was used in order to simplify implementation, however dynamic GOP size parameterization is available in the LibVLC library.

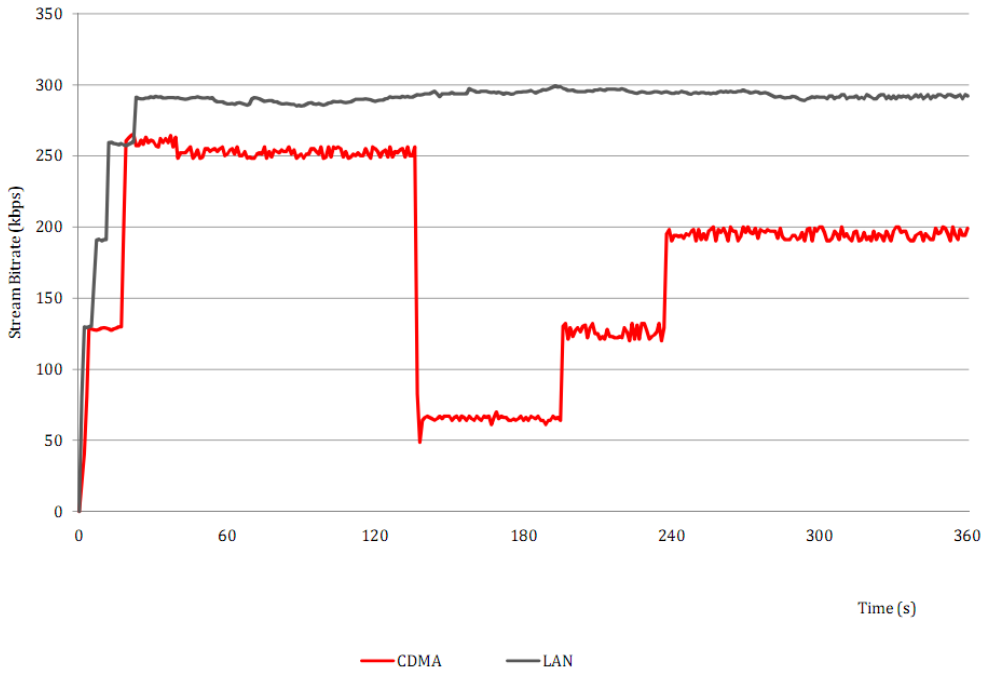


Table 6.3: Dynamic quality level adaptation to network conditions

In Table 6.3 the results of tests within the LAN network show, that content quality usually achieves and remains at the highest value (which was defined at 300 Kbps). This is consistent with the constant nature of the high bit rate, low latency Ethernet network that is the 100 Mbps campus LAN. On the other hand, the CDMA2000 network tests showed that the client requested more streaming session quality changes. The behavior of the system shows clearly that quality levels are dynamically adapted to low bandwidth and high latencies and losses. Although the channel availability values are higher than the aforementioned limit, the solution provides a reasonable

QoE for users with seamless channel changes and quality adaptation.

6.5 System Tests

The scalability of the architecture and prototype was assessed using several IPTV clients and a single IPTV AS prototype. These tests had a duration of thirty minutes each.

To provide a reference for a comparison of all measurements an initial test was run with the server in idle mode. This test consisted in running the server with no connecting clients.

6.5.1 Test Description

The system tests were run in the two modes allowed by the server. In the Per-client mode it was important to ascertain the load on the server by each client, as such, concurrent sessions were tested until the server achieved the maximum load. The tests were as follows:

1. One client using H.264: a test using the H.264 video codec with a single client session;
2. One client using MPEG-4 Visual: a test using the the MPEG-4 video codec with one client session;
3. Multiple clients: a series of tests ranging from three to ten clients using the Per-client mode and five to fifty clients. Due to high complexity of H.264 encoding and the server hardware used, only the MPEG-4 Visual codec was used in these tests.

The metrics analyzed for each test were:

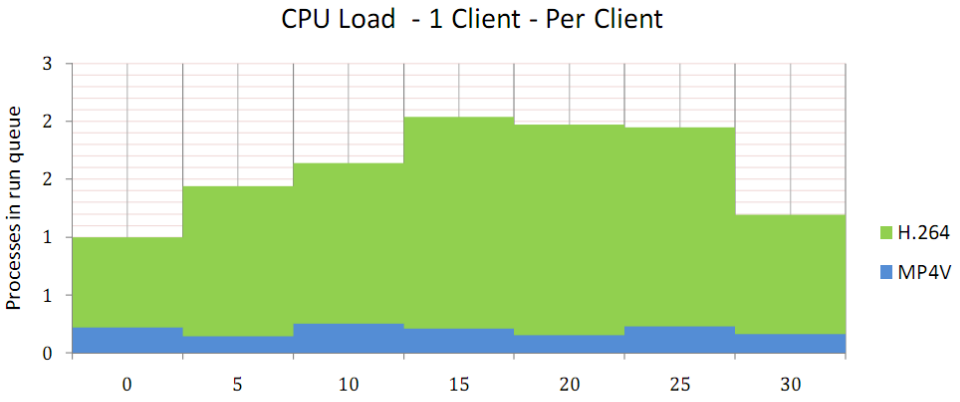
1. Percentage of CPU usage: the CPU measurements are separated into three categories: user, system and nice. The user category consists of CPU usage by user applications. The system and nice categories relate to system specific functions and applications being run. The relevant category studied in the tests is the user category for the IPTV AS runtime;
2. System Load Average: the System Load Average (SLA) in Unix based operating systems is represents the average amount of processes in the system run queue. This allows a general perspective on the system load.
3. Memory usage: the systems memory usage was monitored to detect memory leaks. Even though this problem is not directly related to the servers architecture it yields information about the prototype implementation and stability;
4. Network utilization: the network input and output of data was monitored on the active network interface. This allows assessing the amount of bandwidth required to sustain the service for the various clients.

6.5.2 Test Results

The scalability results vary significantly between the two AS streaming modes. The first group of tests uses the AS Per-client stream mode. The second group of tests uses the General Purpose stream mode.

6.5.2.A Per-client Stream Mode

The analysis of the results of the tests, as may be observed in Figures 6.2 and 6.3, show that using the described hardware, the H.264 codec does not scale well for live encoding when handling more than one client. With one client requesting a live stream the CPU usage is generally over 90%. On average there are 1.60 processes in the run queue, therefore the SLA raises to 160%. The scalability tests were performed both on a private LAN and on a CDMA2000 wireless network, streaming quality is always set to a pre-configured highest value, approximately 300 Kbps. With H.264, two client sessions were tested, but the multimedia streams exhibited high losses and out of synchronization frames, dramatically lowering the QoE. This limitation with H.264 is due to the higher processing power required by the codec. The MPEG-4 Visual codec

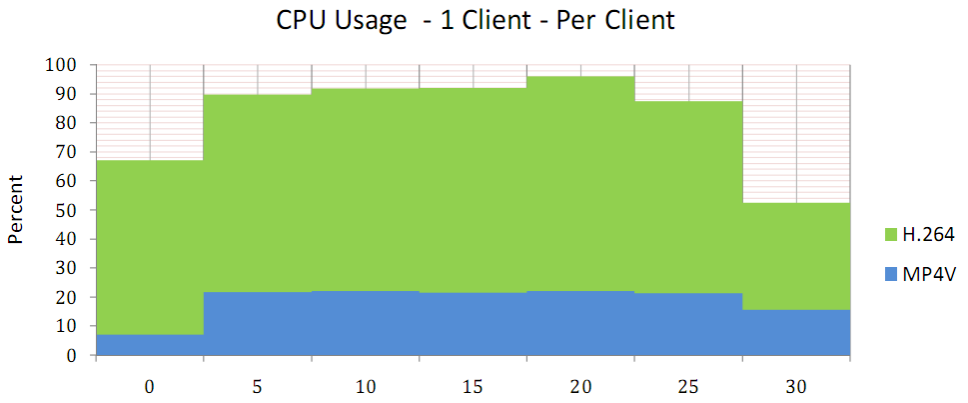


	Average	Minimum	Maximum
H.264	1.60	1.00	2.04
MP4V	0.19	0.14	0.25

Figure 6.2: CPU load comparison with the per-client mode

shows a CPU usage and load considerably reduced when compared to the H.264 codec. One client represents approximately 20% of system usage and 0.2 processes in the run queue for CPU load, i.e., SLA of 20%. With 3 clients the AS system usage rises to approximately 40% and CPU load increases to 0.4 processes in the run queue.

With the machine used for the AS the service can be supplied to a maximum of 10 concurrent clients with the CPU usage reaching sporadically 100%. Similarly, the average run queue length rises to 1.3 processes, which amounts to a SLA of 130%. Even though these values are high, the



	Average	Minimum	Maximum
H.264	82.4	52.5	96.0
MP4V	18.8	7.0	22.1

Figure 6.3: CPU usage comparison with the per-client mode

service on the client application is uninterrupted due to local stream buffering.

The traffic analysis, as observed in Figure 6.4 shows, as expected that for either codec, a client represents approximately 300 Kbps stream output in addition to a overhead introduced by signaling and media packet headers. By linear interpolation from the various tests, a 100 Mbps link allows approximately 150 concurrent clients.

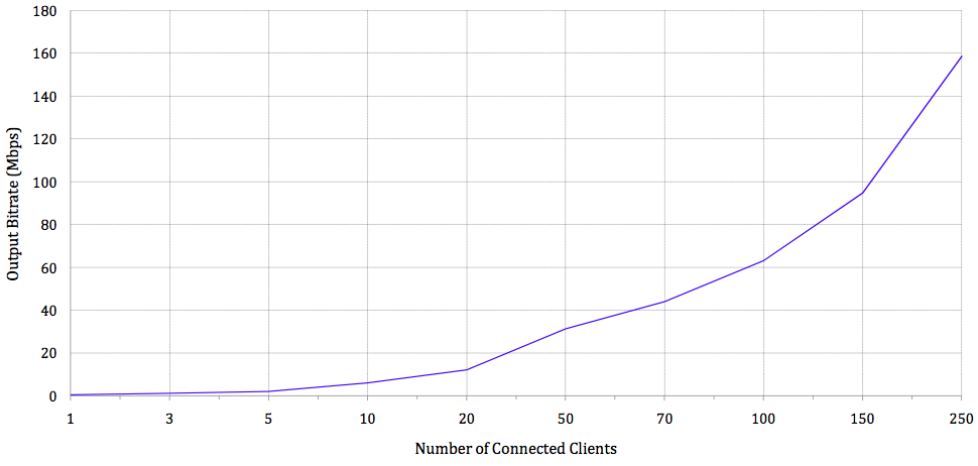


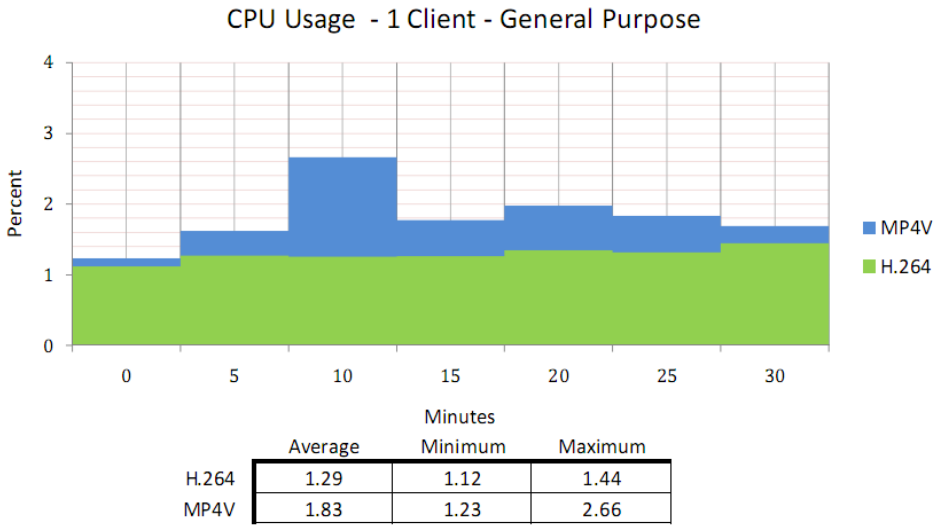
Figure 6.4: Interpolation of outbound traffic on Network Interface Card (NIC)

It is clear from the results of these tests that the choice of the codec has a direct impact on the performance and scalability of the AS. The H.264 codec requires a high percentage of CPU usage for real-time encoding, allowing only a single client session before reaching maximum system load. A possible solution would be to use dedicated hardware for the H.264 encoders. The MPEG-4 Visual codec allows a maximum of approximately 10 users with a single AS prototype

running on the described hardware.

6.5.2.B General Purpose Stream Mode

The IPTV AS in General Purpose stream mode solely receives and forwards the streams required by the clients, with no meaningful impact in behavior from the codec used. As such, tests were run using both codecs under identical conditions. For one client, with both MPEG-4 video and H.264 codecs, the server machine had a CPU usage of approximately 1% and the SLA between 1% and 5%. This clearly contrasts with the results observed in the Per-client mode and demonstrates that both codecs have the same impact on system load and usage.



	Average	Minimum	Maximum
H.264	1.29	1.12	1.44
MP4V	1.83	1.23	2.66

Figure 6.5: CPU usage comparison with the general purpose mode

From Figure 6.5, it is clear that the General Purpose mode offers a higher scalability than the Per-client mode as it shares resources among the connected clients. With 10 clients the CPU usage rises to 10% and the SLA is approximately 40%. With 50 concurrent clients the CPU usage rises to 22% with an average of 2.9 processes in the run queue during the tests. By a linear interpolation of this data it can be estimated that the application server can support up to 250 concurrent clients (considering service can be supported with 100% CPU usage and keeping the same hardware used for the AS machine), as seen in Figure 6.6.

The General Purpose stream mode has clear advantages but can only in the case of transmission of live content.

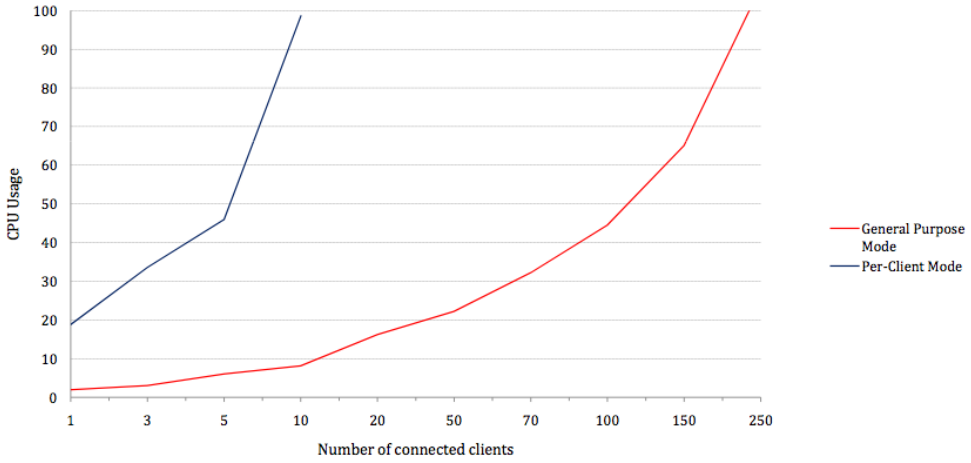


Figure 6.6: System load and usage with various clients for both server modes

6.6 Conclusion

This chapter described the functional and system tests that were carried out and analyzed.

An important test result relates the content adaptation technique that proved the expected behavior for seamless transition upon client requests. This feature allows a client to change streaming parameters on request with little or no influence on the Quality of Experience. In IMS environments, this is of utmost importance as heterogeneous and very distinct access networks are used.

Tests on both LAN and CDMA2000 wireless networks show that the IPTV service is sustainable on both networks. The delivery of the service in these networks prove the dynamic nature of the IPTV AS to adapt to varying network conditions.

The non-functional aspects also show relevant results, as the two AS stream modes have very distinct hardware requirements. These requirements have direct impact on the number of client sessions that are sustainable. The Per-client mode demonstrated very limited scalability mainly due to the encoding mechanisms used. On the other hand, the General Purpose mode offers a highly scalable solution and is not significantly influenced by the encoding used. For both stream modes, the AS scalability can be improved by delegating the encoding and streaming selection mechanisms to independent media servers.

7

Future Work and Final Conclusions

Contents

7.1 System Limitations and Future Work

The developed IPTV AS prototype presents various limitations that can be overcome with future work. The main focus would be on Access Control and Charging functions, content adaptation mechanisms and scalability of the AS for both Broadcasting and VoD services.

Although the AS implements some Access Control and Charging (accounting) mechanisms, these are quite rudimentary. A dedicated AAA module within the architecture could be deployed using the diameter protocol [73]. This protocol allows the AAA functions to be delegated to a dedicated AAA Diameter server. Among other features the protocol provides: reliable transport of AAA messages (important for accounting where message losses may translate into revenue loss), proxy agent support (which allows roaming mechanisms to be in place) and most importantly it is specified within the IMS architecture and can provide a single AAA platform. This will allow user profiles to be in stored in the network, thus freeing the need for the user equipment to maintain user preferences.

The current content adaptation mechanisms that the AS prototype provides are very simple, and while effective in most scenarios, they do not fully support the H.264 codec. When a seamless transition is needed, the AS needs to interrupt the current stream and switch to a new one. This transition can be accomplished without impacting the users visualization, but only if the content bitrate is low. This is primarily due to buffering in the switching mechanism. Future work in this area would be required to a seamless content adaptation improvement, eventually using a leaky-bucket type of transmission mechanism. Additionally, in order to provide a flexible means of increasing compression factors, a dynamic GOP functionality can be implemented using the current LibVLC API.

As seen in chapter 6, the AS performance is highly degraded when in the Per-client mode. This is due to the live encoding mechanisms that are needed for personalized streaming of broadcasting events (such as temporary defer of live content or replay mechanisms). A simple solution that can be easily implemented is to save the previously encoded content to a file on the video server. This means that, even though multiple copies of the stream are saved, the need for encoding is removed and thus processing power is greatly reduced.

In the General Purpose mode, scalability can be hindered by the amount of encoders in use. As the amount of channels increase and dissimilar users join the streams, more encoders are needed to supply different streams. To improve this mechanism an innovative encoding mechanism can be used, called SVC [74]. As part of the MPEG-4 group of codecs SVC allows multiple subsets of a stream to be encoded, thus allowing multiple qualities to be transcoded with a single encoding algorithm. This solution can also be applied to the content adaptation mechanism referred above. This codec, however still has limited applications due to the processing power required to encode and even decode the stream. Despite the hardware requirements, more pow-

erful future equipment may make feasible and common the applications of this codec.

7.2 Conclusion

This dissertation proposes a novel IPTV client-server architecture that can be used on heterogenous networks. The main goals of the dissertation were the study of the state-of-the-art technologies that make possible offering the IPTV services and, in a second phase, to develop an IPTV system, for broadcasting or for content on demand, independent of the end users terminal type or network access technology.

As a starting point, the major IPTV technologies and concepts were studied. These included the network architectures and protocols used for IPTV service delivery as well as the video and audio encoding algorithms for transmission of multimedia content over digital mediums. The integration of IPTV service in next generation networks was also studied as well as the requirements it entails and advantages of such an integration.

The second phase was dedicated to the design and implementation of the IPTV architecture, an Application Server capable of delivering IPTV service in an Next Generation Network environment with adaptation of content being transmitted to the network used to access it. Another design goal was to respect the conditions for integration in an IMS framework.

A new IPTV signaling structure is proposed based on the Session Initiation Protocol (SIP) widely used for other multimedia applications. The SIP signaling structure allows the IPTV service to be integrated with other SIP services supported in IMS environments, such as VoIP or IM. By leveraging SIP flexibility, Trick Functions are incorporated without the need to extend SIP with new message types. This is accomplished by receiving SIP Update messages from the client to dynamically change video, audio and transmission attributes to reflect the requested changes.

From a service perspective, two main IPTV stream modes of operation were identified: Video-on-Demand (VoD) and Broadcasting, that vary in requirements and applications. VoD delivers a personalized and unique stream to a end user client. The Broadcasting mode, as the name implies, is suited for the delivery of traditional television service, where multiple clients consume the same content. From a platform perspective two related stream modes were developed for the IPTV AS: a Per-client stream and a General Purpose stream, respectively suited for VoD and Broadcasting operations.

The architecture was tested from functional and non-functional perspectives. The functional tests showed that the IPTV AS successfully delivers the IPTV service using SIP as a signaling protocol, requiring less messages to be transmitted for setup, modify and tear-down sessions, when compared to RTSP or SIP+RTSP. The Trick Functions provided are able to be used within specified limits for broadband connections. The new QoS adaptation method implemented allows dynamic updates of session attributes

From a non-functional perspective, the prototype was able to sustain various clients. In the Per-client mode, a maximum of 10 clients were supported using the MPEG-4 Visual codec. With the H.264 codec a single instance was deployed for the Per-client stream mode as the encoding mechanism required 100% of CPU usage. In the General Purpose mode, independently of the encoding scheme or network parameters, 50 clients could be served using only 20% of the available system resources. The tests have proved that the overall performance of the system can gracefully adapt and scale to a large number of clients with different network conditions, always offering the highest possible QoE to the user, making the proposed architecture suitable for live multimedia streaming.

Future work will cope with further service developments, essentially related to Media Delivery Functions, such as, adaptive content processing, enhancements to minimize channel switching delays and to dynamically adjust QoS/QoE parameters

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Appendix A

A.1 Session Establishment, Modification and Teardown

Client originated messages are colored red and server responses are colored blue. Depicted in this series of messages is a session establishment with a initial quality level of 10, followed by a change of level to 8. The session is terminated by the clients request.

```
INVITE sip:video1@192.168.0.1 SIP/2.0
From: <sip:cliente-iptv@192.168.0.2>
To: <sip:server-iptv@192.168.0.1>
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length:   418

v=0
o=SIP Server/Proxy 1234567890 1234567890 IN IP4 192.168.0.1
i=SIP IPTV Client session description
c=IN IP4 192.168.0.1
t=0 0
m=application 9 TCP/SIP sip
b=AS:300
a=connection:new
a=setup:active
a=quality:10
a=framerate:23
a=fmtp:media uri=sip:video1@192.168.0.1
m=audio 1234 RTP/AVP 97
a=recvonly
a=rtcp:0
a=rtpmap:97 mpga/90000
m=video 1234 RTP/AVP 96
a=recvonly
a=rtcp:0
a=rtpmap:98 h264/90000

SIP/2.0 100 Trying
From: <sip:cliente-iptv@192.168.0.2>
To: <sip:server-iptv@192.168.0.1>
CSeq: 1 INVITE
```

```
Max-Forwards: 70
Content-Length: 0

SIP/2.0 200 OK
From: <sip:cliente-iptv@192.168.0.2>
To: <sip:server-iptv@192.168.0.1>
CSeq: 1 INVITE
Content-Type: application/sdp
Max-Forwards: 70
Content-Length:   360

v=0
o=SIP Server 1234567890 1234567890 IN IP4 192.168.0.1
i=SIP IPTV Client session description
c=IN IP4 192.168.0.1
t=0 0
m=application 1234 TCP/SIP sip
b=AS:300.000
a=connection:new
a=setup:active
a=quality:10
a=framerate:23.00
a=fmtp:media url=video1
m=audio 1234 RTP/AVP 97
a=recvonly
a=rtcp:0
m=video 1234 RTP/AVP 96
a=recvonly
a=rtcp:0

ACK sip:video1@192.168.0.1 SIP/2.0
From: <sip:cliente-iptv@192.168.0.2>
To: <sip:server-iptv@192.168.0.1>
CSeq: 2 ACK
Content-Length: 0

UPDATE sip:video1@192.168.0.1 SIP/2.0
```

```
From: <sip:cliente-iptv@192.168.0.2>
To: <sip:server-iptv@192.168.0.1>
CSeq: 3 UPDATE
Content-Type: application/sdp
Content-Length:   418

v=0
o=SIP Server/Proxy 1234567890 1234567890 IN IP4 192.168.0.1
i=SIP IPTV Client session description
c=IN IP4 192.168.0.1
t=0 0
m=application 9 TCP/SIP sip
b=AS:250.00
a=connection:new
a=setup:active
a=quality:8
a=framerate:23
a=fmtp:media uri=sip:video1@192.168.0.1
m=audio 1236 RTP/AVP 97
a=recvonly
a=rtcp:0
a=rtpmap:97 mpga/90000
m=video 1236 RTP/AVP 96
a=recvonly
a=rtcp:0
a=rtpmap:98 h264/90000

SIP/2.0 200 OK
From: <sip:cliente-iptv@192.168.0.2>
To: <sip:server-iptv@192.168.0.1>
CSeq: 3 UPDATE
Content-Type: application/sdp
Max-Forwards: 70
Content-Length:   374

v=0
o=SIP Server 1234567890 1234567890 IN IP4 192.168.0.1
i=SIP IPTV Client session description
```

```
c=IN IP4 192.168.0.1
t=0 0
m=application 1236 TCP/SIP sip
b=AS:300.000
a=connection:new
a=setup:active
a=quality:10
a=framerate:23.00
a=fmtp:media url=sip:video1@192.168.0.1
m=audio 1236 RTP/AVP 97
a=recvonly
a=rtcp:0
m=video 1236 RTP/AVP 96
a=recvonly
a=rtcp:0

ACK sip:video1@192.168.0.1 SIP/2.0
From: <sip:cliente-iptv@192.168.0.2>
To: <sip:server-iptv@192.168.0.1>
CSeq: 4 ACK
Content-Length: 0

BYE sip:video1@192.168.0.1 SIP/2.0
From: <sip:cliente-iptv@192.168.0.2>
To: <sip:server-iptv@192.168.0.1>
CSeq: 5 BYE
Content-Length: 0

SIP/2.0 200 OK
From: <sip:cliente-iptv@192.168.0.2>
To: <sip:server-iptv@192.168.0.1>
CSeq: 5 BYE
Max-Forwards: 70
Content-Length: 0
```

B

Appendix B

Contents

B.1 Libraries and OS

The application server was developed and tested on Ubuntu 8.04 and the following libraries were used:

- libvlc0 - VLC API library;
- libosip2 version 3.1.0 - The SIP stack library used with the AS.

To install on Ubuntu run the following commands:

```
# sudo apt-get install build-essential

# wget http://ftp.gnu.org/gnu/osip/libosip2-3.1.0.tar.gz
# tar -zxvf libosip2-3.1.0.tar.gz
# cd libosip2-3.1.0
# ./configure && sudo make && sudo make install

# sudo apt-get install libvlc0-dev vlc
```

B.2 Installation

To compile the server simply type:

```
# make
```

B.3 Configuration

To change the configuration of the server please edit the 'config.txt' present in the config folder. Instructions are in the file.

B.4 Execution

To run it simply type.

```
# ./server
```

Optionally you can specify a port to listen on in the configuration file or as an argument in the CLI.