HELSINKI UNIVERSITY OF TECHNOLOGY
Department of Computer Science and Engineering
Degree Program of Information Networks

INTERNET PROTOCOL DATACASTING

A Technology Overview

Master's Thesis

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ABSTRACT OF MASTER'S THESIS

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It is difficult to provide multimedia services to mobile users in large scale. Mobile users cannot, for instance, make use of the services provided over the terrestrial digital television network. The reception of such signals is processing-intensive and a receiver in motion is further exposed to different types of disturbances than a stationary receiver.

IP Datacasting, or IPDC, uses digital broadcasting technology adapted for mobile users to deliver data cheaply and efficiently. Yet, in order for the receiver to reach the sender, it must use a separate interaction channel. This combination of a unidirectional broadcast channel and a separate interaction channel sets challenges on the IPDC delivery architecture. Although some established protocols can be used in IPDC as such, others need to be modified and for some purposes, new protocols must be designed.

The objective of this thesis is therefore to review the protocols and standards related to IPDC and to evaluate how they take the constraints and requirements of IPDC into consideration. The thesis also presents a five-layer model of the IPDC delivery architecture and evaluates each layer separately. The session layer is found to contain the most unsolved questions.

Keywords: IPDC, IP Datacasting, architecture, evaluation, unidirectional delivery
Language: English

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Det är idag svårt att erbjuda multimediatjänster i stor skala till mobila användare. Mobila användare kan till exempel inte tillgodogöra sig det terrestriala digitala televisionsnätets utbud, eftersom mottagningen kräver kraftig processering och eftersom en mottagare i rörelse utsätts för andra slags störningar än stationära.

IP Datacasting, eller IPDC, använder sig av digital broadcastingteknologi som är anpassad för mobila användare. Broadcasting är ett effektivt och billigt sätt för sändaren att leverera data till användarna. Men om en mottagare behöver nå sändaren, måste han använda en separat interaktionskanal. Detta måste protokollarkitekturen för IPDC beakta. Vissa etablerade protokoll kan som sådana användas för IPDC, men andra måste modifieras och för en del ändamål finns det inga lämpliga etablerade protokoll.

Detta diplomarbete strävar till att ge en översikt av de protokoll och standarder som en IPDC leveransarkitektur kan byggas upp av, samt utvärdera hur dessa möter de krav IPDC ställer. För detta ändamål presenteras först en leveransarkitektur i fem nivåer. Denna nivåmodell används genom hela arbetet för att presentera och sedan evaluera IPDC-protokollen och -standarderna. Sessionsnivån visar sig innehålla de största frågetecknena.

Nyckelord:	IPDC, IP Datacasting, arkitektur, utvärdering, enkelriktad leverans
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Mobiileille käyttäjille on tällä hetkellä hankalaa tarjota multimediapalveluita isossa mittakaavassa. Mobiilit käyttäjät eivät voi esimerkiksi käyttää maanpäällisen digitaalisen televisioverkon palveluita hyväkseen, koska vastaanotto on prosessointi-intensiivistä ja vastaanotin kärsii erilaisista häiriöistä kuin paikallaan pysyvä vastaanotin.

IP Datacasting tai IPDC hyödyntää mobiilille käyttäjälle optimoitua digitaalista joukkolähetysteknologiaa. Joukkolähetyksellä voi tehokkaasti ja halvalla toimittaa dataa käyttäjille. Toisaalta joukkolähetys on yksisuuntaista, minkä takia vastaanottajan täytyy käyttää erillistä vuorovaikutuskanavaa, mikäli on tarve ottaa yhteyttä lähettäjään. IPDC:n lähetysarkkitehtuurin täytyy huomioida tämä, ja siksi IPDC:ssä voidaan sellaisenaan käyttää hyväksi vain osaa vakiintuneista protokollista. Toisia protokollia on muunneltava IPDC-käyttöä varten, ja osittain tarvitaan kokonaan uusia protokollia.

Tämä diplomityö pyrkii esittämään sellaiset protokollat ja standardit, joiden avulla voidaan toimittaa palveluita IPDC:n yli. Työ myös vertailee miten nämä protokollat ja standardit vastaavat IPDC:n käyttötilanteen asettamia vaatimuksia. Esitys ja arviointi käyttävät lähtökohtana viisikerroksista IPDC-arkkitehtuuria, joka esitetään työn alussa. IPDC:n suurimmat avoimet kysymykset osoittautuvat kirjoitushetkellä sijaitsevan istuntokerroksella.

Avainsanat:	IPDC, IP Datacasting, arkkitehtuuri, arviointi, yksisuuntaiset lähetysverkot
Kieli:	Englanti

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Espoo March 10th 2004

Linda Staffans

Abbreviations and Acronyms

2k/4k/8k mode	COFDM operation modes
3GPP	3rd Generation Partnership Project
AAC	Advanced Audio Coding (MPEG-4)
AH	Authentication Header; An IPsec security protocol
ALC	The Asynchronous Layered Coding protocol
ATSC	Advanced Television Systems Committee
AVC	Audio/Video codec
CA	Conditional Access. A protection scheme developed by the DVB Project.
CAT	Conditional Access Table; An MPEG signalling table
COFDM	Coded Orthogonal Frequency Division Multiplexing
DRM	Digital rights management
DSM-CC	Digital Storage Media Command and Control
DVB	Digital Video Broadcasting
DVB Project	The Digital Video Broadcasting Project
DVB-C	The Digital Video Broadcasting cable transmission standard
DVB-H	Digital Video Broadcasting for handheld devices. DVB-H adds some features to DVB-T and MPE.
DVB-M	Currently known as DVB-H
DVB-S	The Digital Video Broadcasting satellite transmission standard
DVB-T	The Digital Video Broadcasting terrestrial transmission standard
DVB-X	Currently known as DVB-H
ECM	Entitlement Control Message
EIT	Event Information Table; a DVB signalling table
EMM	Entitlement Management Message
ESG	Electronic Service Guide

ESP	Encapsulating Security Payload; An IPsec security protocol
FLUTE	The File Delivery over Unidirectional Transport protocol
GPRS	General Packet Radio Service
HTML	HyperText Markup Language
HTTP	HyperText Transfer Protocol
IEC	International Electrotechnical Commission
IETF	Internet Engineering Task Force
IMG	Internet Media Guide
INT	IP/MAC Notification Table; A DVB signalling table
IP	Internet Protocol. IPv4 denotes IP version 4 and IPv6 denotes IP version 6.
IPDC	Internet Protocol Datacasting
IPDC Forum	An industry association that investigates the IPDC business concept
IPsec	IP Security
ISDB-T	Integrated Services Digital Broadcasting – Terrestrial transmission
ISO	International Organization for Standardization
ITU-T	International Telecommunication Union (ITU) Telecommunication Standardization Sector
kb	kilobit
LLC/SNAP	Logical Link Control / SubNetwork Access Protocol
MAC	Medium Access; Denotes the hardware receiver unit that is in direct contact with the radio interface.
MBMS	Multimedia Broadcast / Multicast Services
MHz	Mega-Hertz
MIDP	Java MicroEdition Mobile Information Device Profile
MPE	MultiProtocol Encapsulation
MPE-FEC	MultiProtocol Encapsulation – Forward Error Correction
MPEG	Moving Pictures Expert Group. An organization that develops standards for managing digital audio and video. The acronym also denotes the standards developed by this group.
MPEG-2	An MPEG standard for compressing and encapsulating data for transmission.
MPEG-7	An MPEG standard for object description.

MPEG-21	An MPEG standard for managing digital objects, including their digital rights
MUPPET	The Internet Media Guide Unidirectional Point-to-Multipoint Transport protocol
kb	kilobit
Mb	Megabit
OMA	Open Mobile Alliance
OMA DRM	OMA Digital Rights Management; A DRM scheme developed and published by the Open Mobile Alliance
PAT	Program Association Table; An MPEG signalling table
PDF	Portable Document Format; A document format developed by the Adobe company
PES	Packetized Elementary Stream; An MPEG-2 transmission format used to present audio and video.
PID	Packet identifier; Used to identify packets in an MPEG or DVB transport stream.
PMT	Program Map Table; An MPEG signalling table
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RTP	Real-time Transport Protocol
SDP	Session Description Protocol
SDPng	Session Description and Capability Negotiation
SDT	Service Description Table; A DVB signalling table
SRTP	Secure Real-time Transport Protocol
TPS	Transmission Parameter Signalling
UDP	User Datagram Protocol
ULE	Ultra Light-Weight Encapsulation
WAP	Wireless Application Protocol
WWW	World Wide Web
XML	eXtensible Markup Language
XPath	XML Path Language
XPointer	XML Pointer Language

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Chapter 1

Introduction

A growing amount of people spend much of their day travelling. They use mobile communication technology for business as well as leisure. Yet, the selection of content is rather limited; 2.5G telephone technologies like General Packet Radio Service (GPRS) provide at the maximum a transfer rate of 2 Mb/s. Distributing content is also expensive as telephone networks use point-to-point connections. As a result, mobile telecommunication technology is mainly used for personal communication. In general, users prefer consuming entertainment content over media that require receivers to be more or less stationary; e.g. television and high-speed Internet connections.

Internet Protocol Datacasting combines IP-based data transmission with radio broadcasting to enable distributing multimedia content cheaply to a large group of users. In simple terms, it can be called digital television on mobile handheld terminals. Unlike the leading digital television standards, IP Datacasting does not specify the content format. This first of all eases sending different types of content using different delivery protocols. Second, it enables using different compression techniques depending on the target user group. By using a picture format that is optimized for a small screen, delivery capacity is saved.

The IP Datacasting technology and its corresponding business concept have been developed for only a few years; the first research reports appeared in 1999 (see e.g. [29, 77]). The documentation of the concept is scattered and it is difficult for practitioners and researchers to find comprehensive, complete documentation. Parts of the IP Datacasting standards are at the time of writing still not published, further impeding the researcher's efforts to find information about the concept. The main objective of this thesis is therefore to provide a comprehensive overview of the IP Datacasting technology, especially of its applications in Finland. Further, this thesis evaluates how the IP Datacasting technology serves its purpose by comparing its centralmost design choices with a set of requirements.

Chapter 2

Background

The IPDC Forum is an industry forum that investigates the business concepts based on the IP Datacasting technology. They describe IP Datacasting, or IPDC for short, in the following way:

> *In IP Datacasting any digital content can be delivered cost effectively over broadcast networks to large audiences at the same time. For consumers, this means more choice in accessing multimedia content and a likely increase in content possibilities.*
>
> *IP Datacasting is a service where digital content formats, software applications, programming interfaces and multimedia services are combined through IP (Internet Protocol) with digital broadcasting. [41]*

The way IP Datacasting is used can be divided into two rough categories:

- Downloading files or applications for later use, and
- Real-time streaming

Additionally, interactive IPDC services add interactivity to the broadcast files and streams.

Similarly, a service can be intended for three types of users:

- For a specific group of users, for instance for a company or for everybody who has purchased a telephone subscription within the last month, or

- For anyone who pays for the service, or
- For anyone.

The trump card of IPDC is cheap content delivery. Digital television networks are broadcast networks, which means that all the users receive the same content. On the downside, content can therefore not be personalized in the same way as for instance in 3G point-to-point networks. Yet, the broadcast network provides enough capacity for 30-50 simultaneous services which can be designed for as many different target groups. IPDC therefore positions itself quite differently from 3G content delivery, which offers personal, on-demand services but on the other hand only serves a limited number of simultaneous users.

The transmission technology in itself is not enough to produce services. The INDICA project uses a customer centric value chain model, based on a similar model laid out by the European Commission [26], to understand what parts an IPDC service consists of. INDICA is a joint research project that investigates the IP Datacasting value chain and its business concepts. Several companies that represent different roles in the IP Datacasting value chain (Radiolinja Oy, the city of Helsinki, Elisa Research, A4 Media Oy, Fontus Oy), Helsinki University of Technology and Tampere Polytechnic participate in the project. INDICA's value chain model is presented in Figure 2.1.

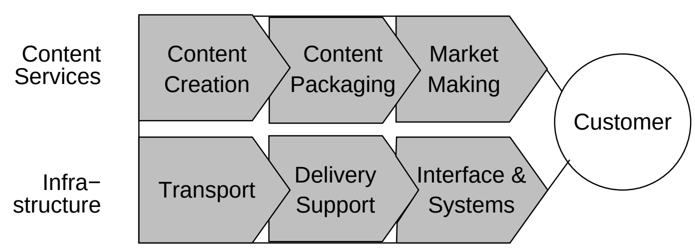

Figure 2.1: The INDICA two-layered value chain model.

First of all, there are actors who create and package the content and create a market by, among others, creating and promoting service brands and advertising the services. Second, the services need a delivery infrastructure. The infrastructure consisting of a transport medium, delivery support systems such as access control, and the customer's terminal and applications through which he uses the IPDC services.

The part of the model that concerns the content services actors is only weakly coupled to the IPDC technology standards. Still, one concern that a content packager may have is that he for example should be able to use a new coding-decoding algorithm that is pre-installed in all the IPDC terminals that are manufactured 2006 or later. This and other requirements on the IPDC technology are discussed in Section 4.

The infrastructure, on the other hand, directly affects the IPDC technology standards. To provide data transport, IPDC needs an efficient transmission technology but also transport protocols that support unidirectional multicasting. Further, the architecture must have some access control and a mechanism for buying services that an application can automate as much as possible. There should also be some common mechanism for announcing all services in a specific network. Possibly, some part of the user terminals should also be standardized so that it is as easy as possible to create new interactive or non-interactive services.

This chapter lays out the background of IP Datacasting. First, some usage scenarios illustrate what types of services IP Datacasting enable. Section 2.2 describes the IPDC value chain, and Section 2.3 defines terms used in this thesis. Then, Section 2.4 describes the objective of this thesis, and Section 2.5 restricts the problem scope. Finally, the structure of the thesis is described in Section 2.6.

2.1 Usage Scenarios

IP datacasting can be used for selling – for example – videos, music, TV shows or news. The following examples illustrate the IPDC service market.

Watching music TV. *Eve, 16, is sitting in the bus, on her way home from school. She feels like listening to music, so she switches on her IPDC terminal and chooses the publicly available music channel. She leans back and listens to the music through her earphones, while casually watching the music videos on her terminal. Now and then the music or the video is interrupted with a commercial.*

In other words, Eve accesses a public service consisting of two components: the music and the video streams.

Ordering a service. *Joe, 35, sits in the train. It's 8 A.M. and he's on his way to work. There's a news channel that provides a news flash every half hour. The terminal shows that the news channel is locked and informs Joe that watching a news flash costs 2€. Joe chooses the news flash that starts next, confirms the order, and waits 20 seconds before the terminal starts* *showing it.*

In other words, Joe accesses an encrypted service. When Joe chooses the encrypted service, the terminal orders a rights object containing information with which the terminal can decrypt the service.

There are variations on this usage scenario. Public channel operators may want to keep track of how services are used, for instance for demonstrating the value of the channel for commercial producers. In that case, the public service is encrypted and rights objects can be ordered free of charge.

Subscribing to a channel. *Dennis, 5, and his parents have a 45 minute drive ahead of them. They're going to see grandma. Dennis' Dad crawls in on the back seat with Dennis to watch cartoons with him. Dennis turns on the children's cartoons channel on Dad's IPDC terminal. The channel is protected, but Dennis' Dad subscribes to it, which enables him to watch the channel exactly as if it were free-to-air channels.*

In other words, Dennis' Dad accesses an encrypted channel. The rights object that gives access to the services in that channel must be renewed periodically, for instance once a month. The rights objects may be purchased using the interaction channel, as when specific encrypted services are accessed. Alternatively, rights objects may be distributed in other ways, for instance fetched from a web portal or delivered as an add-on to other services, such as a pay-TV cartoons channel.

Downloading a file. *Felicia, 55, goes by bus to work every morning. While enjoying her ride, she browses through the daily gossip newspaper "Evening headlines". Felicia leaves her IPDC terminal on during the night to download the morning "Evening headlines". Right now, she has a monthly subscription that ends the 30th.*

In other words, Felicia's terminal accesses an encrypted service, the content of which is stored in the terminal for later use.

Voting. *Gregory, 25, watches a reality show where one of the participants is voted out every day. Next to the video running on his IPDC terminal is a link, urging Gregory to vote for a 0,7€ fee. As Gregory chooses the link a voting form, listing the candidates, is opened. Only one vote per terminal is allowed.*

In other words, Gregory accesses some resource (the voting WWW-page) that is announced in an IPDC service. The results of the interaction (in this case, the voting results) may be delivered in another IPDC service. The terminal must provide a way for connecting to such an external resource.

Barring access to some service. *On the way home, Dennis, 5, sits alone in the back seat. He watches cartoons on the IPDC terminals. He switches between the two cartoons channels the family orders. Dennis parents have beforehand checked that Dennis does not see the action movie channels and that Dennis cannot order new services or other objects such as ring tones.*

In other words, Dennis is denied access to non-free services. Additionally, the terminal is configured to show only certain services.

Delivering a company news paper. *FinancialFit Ltd has a company newspaper that is published once a month. During the night after the publication, it is broadcast to all employees.*

In other words, a file is broadcast to a predefined group of users.

2.2 IPDC Value Chain

The INDICA value chain model, depicted in Figure 2.1, illustrates the different parts that an IPDC service consists of. Expertise in several different areas and different technical equipment is combined in every IPDC service. Therefore, several organizations co-operate in producing services for the mobile user.

As many as eight roles can be identified in the IPDC value chain (Figure 2.2). *Content producers* deliver content to *content providers*, who may aggregate or modify the contents before selling them to *content aggregators*. The content aggregator forms a well-balanced combination of services, schedules them and delivers them to the *service operator*. He, in turn, protects the content, forms price listings and access rights objects, and creates an electronic service guide. The stream is sent to the IPDC *network operator* who broadcasts it onto the air. The *user* receives the IPDC stream and watches the streams or uses the services. Some services are available directly, while he orders others from the *interaction channel service operator*. The *interaction channel network operator* maintains and operates the physical interaction channel network, e.g. a GSM or UMTS network. [72]

The network operator role is special in the sense that operating a broadcast network requires a license. In Finland, there will initially be only one physical IPDC channel and hence there will be one single network operator. The other roles, on their part, can be combined. An organization can even form a complete value chain from content production to rights provision.

The IPDC technology mainly concerns the five right-most actors in Figure 2.2; the service and network operators and the user. Therefore, the term *content producer* in this thesis refers to all the actors that produce, provide or aggregate the content. Although the content producer role is enough to distinguish between different actors in this thesis, it is a crude generalization. The relationships between content producers, providers and aggregators can be complex – rather, several more roles could be identified.

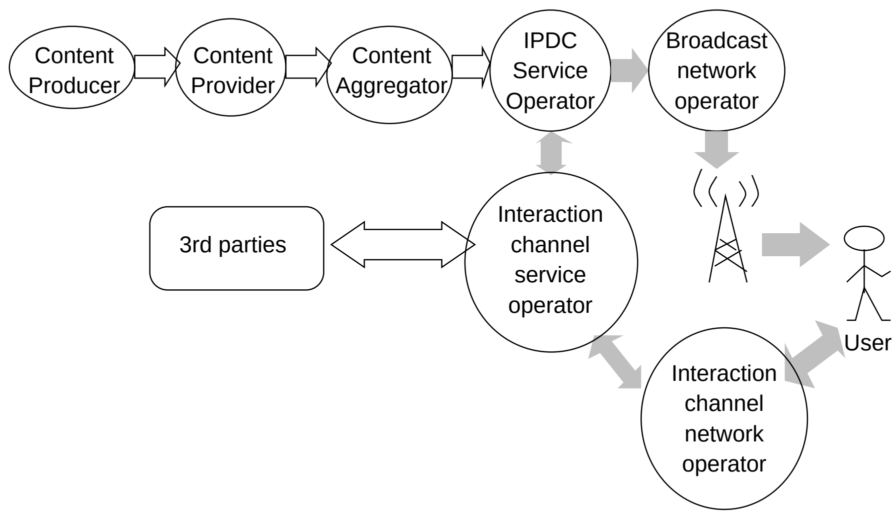

Figure 2.2: The IPDC value chain. The interactions discussed in this thesis are shaded.

2.3 Terms

The most central concepts that are used throughout this thesis are the following:

A carrier	is an electromagnetic wave that is modulated in order to transmit data.
A codec	is a compression/decompression algorithm.
A multiplex	is an MPEG transport stream that has been formed by multiplexing several MPEG transport streams.
A profile	is a term used in the MPEG standards. In this thesis, it refers to the different MPEG-2 mechanisms for transmitting data.
A program	is a term used in the MPEG standards and denotes something similar to a TV channel, such as EuroSport.
A rights object	is a data package containing information with which the terminal can access a specific service.
A service	is a content module that is delivered over IPDC. An electronic newspaper or a cartoon that contains video and audio are examples of a service.

2.4 Problem Statement

The main objective of this thesis is to describe the IP Datacasting technology and sketch the direction of its development. To highlight its strengths and weaknesses, a set of design requirements is deduced and the IPDC technology is analyzed against those requirements.

2.5 Scope of the Thesis

As the INDICA value chain model (Figure 2.1) reveals, the IPDC technology covers everything from the transmission network to content servers and payment systems. However, only the transmission protocols and the service announcement techniques are currently being standardized – open or proprietary industry standards will instead be developed for content servers, payment systems and the rights object that a user needs to access a protected service. Although the thesis shortly discusses and analyses also delivery support systems that will not be standardized, the main focus is on common standards development.

Further, the thesis focuses on the IPDC technology when used within a specific IPDC network. Interoperability between different networks, for instance how a user can continue using a service when moving from one country to another, are discussed shortly.

2.6 Structure of the Thesis

This thesis starts with the background to IP Datacasting. The first chapter shortly introduces the IP Datacasting target market. Then, Chapter 2 lays out the background to IP Datacasting by illustrating the services that can be developed with IP Datacasting and describing the IPDC value chain. The chapter also specifies the objective and the scope of this thesis.

Next, in Chapter 3, the thesis gives an overview of the IP Datacasting technology. Chapter 3 first defines a layer model for the IPDC transmission protocols and then discusses each layer independently. Last, the interaction channel is discussed separately.

The focus then switches to analyzing the information presented so far. Chapter 4 therefore derives constraints and requirements of the IP Datacasting technology. Accordingly, Chapter 5 analyses the IP Datacasting technology against these requirements.

Chapter 6 finally combines the information from the previous chapters by presenting an example IPDC network architecture. The next chapter, Chapter 7 sketches how the IP Datacasting technology and its business concepts will develop in the next few years.

Finally, Chapter 8 concludes by identifying the strengths and weaknesses of the IP Datacasting technology, based on the presentation in this thesis.

Chapter 3

IP Datacasting

The strength of IPDC is broadcasting data to a group of users. Figure 3.1 illustrates how the audio stream of a cartoon is packaged and sent. Video streams are handled similarly. In this fictive example, CrazyCartoon Finland Oyj wants to distribute its newest episode of "Antti, the Courageous Ant" to the public. At least one user, Benny Boy, eagerly waits for the cartoon to start.

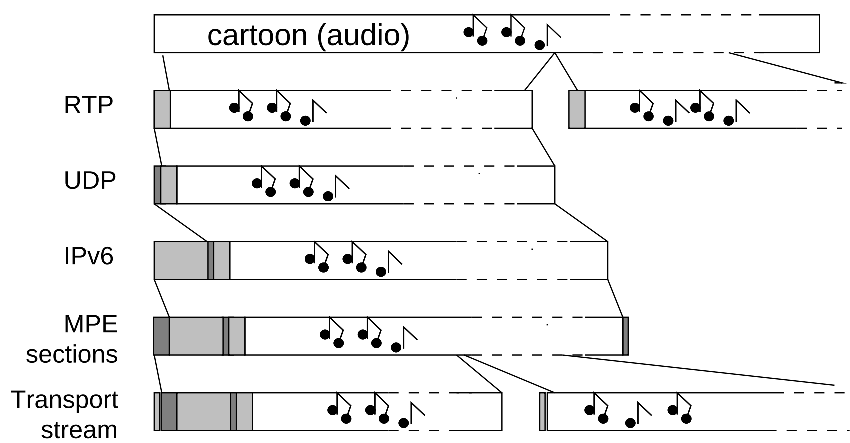

Figure 3.1: Encapsulating an audio stream for transmission over a DVB transport stream.

> *The distribution chain naturally starts with creating the cartoon itself. CrazyCartoon Finland Oyj uses a heap of creativity and time plus their favorite animation tools to produce a 15 minutes cartoon strip of "Antti, the Courageous Ant". The cartoon maker*

> *stores the cartoon files on a content server and schedules them for delivery on Saturday at 14 o'clock.*

CrazyCartoon Finland Oyj optimizes the resolution and compression for small screens of about 10cm $\times$ 6cm. The result consists of two streams; a video stream and an audio stream. They apply the H.264 video encoding and the MPEG4-AAC audio encoding algorithms, yielding total file sizes of about 120 Mb and 230 Mb for the video and audio streams, respectively.

> *Crazy Cartoon Oyj has also entered "Antti, the Courageous Ant" into the electronic service guide. The electronic service guide contains all services available in the IPDC network. The service guide is broadcast every few seconds.*
>
> *Saturday comes and when the clock strikes 14, the content server prepares the files for transmission and sends them to the IPDC encapsulator.*

The two files are split up into datagrams. Some synchronization information is added, mainly to allow the receiver to synchronize the two streams and deal with any delay and jitter introduced during the transmission. In this case, the Real-time Transport Protocol (RTP) is used to send the two streams, and the RTP Control Protocol (RTCP) to send some additional timing information. The RTCP packets are not shown in Figure 3.1. Each RTP datagram contains a three seconds of the stream; in other words about 750 kb of the video stream or 380 kb of the audio stream. The RTP header adds 12-72 bytes to each RTP-datagram, and the RTCP-datagrams are at least 28 bytes each. On the other hand, the total RTCP-traffic is marginal compared to the RTP traffic.

Next, the sender has to choose a suitable transport protocol. Since the broadcast network is a unidirectional, it is natural to choose the User Datagram Protocol (UDP) which contrary to the Transmission Control Protocol (TCP) does not require any acknowledgements from the receiver. The UDP header adds 8 bytes overhead. Next, the UDP packets are packed into Internet Protocol version 6 (IPv6) datagrams. The sender addresses the datagrams to a specific IP multicast group. In addition to the normal IPv6-header, an additional protection header is added and the datagram payload is encrypted. Each IP datagram contains at the most about 4000 bytes payload (the whole datagram can be 4080 bytes long, including the 40 bytes IPv6-header and the IPSec protection headers).

The central invention of IPDC is to use a digital television broadcast network as the physical carrier for the IP data. There are several such transmission standards; in Europe, the digital video broadcasting transmission standards of the DVB Project are used. Especially, the DVB terrestrial transmission standard was enhanced in 2001-2003 to provide better support for mobile receivers with limited power supplies. The resulting configuration is called DVB for handheld receivers (DVB-H). The Finnish broadcast network in 2005, when this fictive example occurs, is a DVB-H network designated for IPDC-transmission.

The IP datagrams are first prepared for delivery over DVB-H by attaching a MultiProtocol Encapsulation (MPE) section header of 12 bytes to each datagram, plus a 4 bytes checksum trailer. A section is at the most 4096 bytes and since the maximum IP datagram size was 4080 bytes, there is always one multiprotocol encapsulated IP datagram in each section. At this stage some additional error correction information is computed over groups of four IP datagrams and sent in MultiProtocol Encapsulation sections of their own.

The cartoon datagrams are now about to enter the DVB transport stream that has a capacity of about 12 Mb/s. The transport stream is divided into 30-50 slots of a few hundred milliseconds each; one of these slots is used for the cartoon streams. The sections are now divided into transport packets of 184 bytes, each receiving an additional 4 bytes header. Additional error correction is applied and the resulting stream is modulated, dividing each symbol over about 4000 subcarriers.

> *Benny Boy has now already turned on his receiver. He finds "Antti, the Courageous Ant" in the service list. The cartoon costs 30 cents, so when Benny picks it, the terminal asks him to confirm the price. Benny now eagerly waits for the cartoon to start.*

The receiver uses the service guide to find out which multicast group the cartoon is sent to. Further, the guide reveals that the cartoon is encrypted; accordingly, the receiver fetches decryption parameters from a rights vending server. The decryption information forms a rights object and is stored in secure memory. It further resolves which transport stream packet identifiers corresponds to the IP-address. Benny Boy's terminal is now prepared for receiving cartoon datagrams.

> *The data stream containing "Antti, the Courageous Ant" now hits the transmitter and is broadcast.*

The modulated stream is transmitted using a low-power transmitter, and all users within about 30 km of the transmitter receive it. The cartoon transport stream packets arrive also at Benny's receiver. The receiver filters out the cartoon packets using the transport stream packet identifier, unwraps the datagrams of the different layers one by one, and finally delivers the RTP-streams to a player application.

> *A jingle arouses Benny's attention. "Antti, the Courageous Ant" has started! Euphoriously, Benny Boy drifts away to the land of the courageous ants.*

As this example illustrates, the IPDC architecture provides general support for data transmission and service announcement, but several choices concerning content formats, protocols, protection mechanisms and the digital broadcasting technology must be made. The next sections describe the architecture layers thoroughly, clarifying the differences between IPDC and digital television or stationary Internet connections. The last section of this chapter discusses the interaction channel.

3.1 Layer Model

The IPDC architecture can be though of as consisting of five layers. As Figure 3.2 reveals, these layers are dedicated to application, presentation, session, transport and link level functionality, respectively.

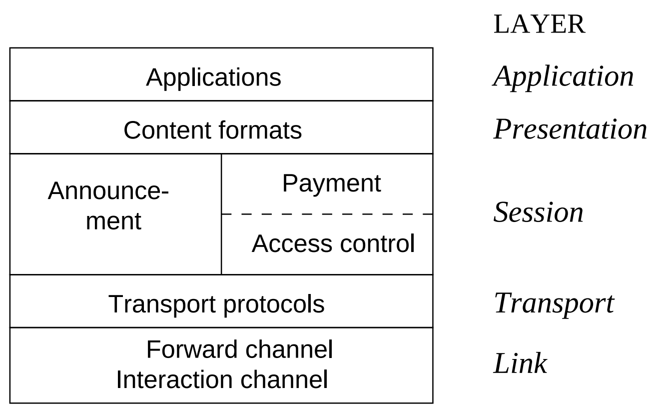

Figure 3.2: The IPDC architecture

The model is similar to the OSI seven-layer model [70]. However, the session layer is less rich in functionality and adapted for simplex data transmission combined with a bandwidth-limited duplex interaction channel. OSI-layers are further combined; the IPDC transport layer combines the OSI transport and network layers, and the IPDC link layer corresponds to the OSI link and physical layers.

The following sections review the forward channel on each of the architecture layers, describing the functionality they provide and presenting their most topical corresponding protocol candidates. Section 3.7 discusses the interaction channel.

During the presentation, the reader will notice that some protocols include features from several layers. The reason is simple; Several protocols are developed for other purposes, but are used in the IPDC architecture to provide specific layer functionality. For instance, the DVB transmission standards are used only for transmitting data, although they also can be used for announcing the broadcast services. Further, especially the session layer functions are reflected on several layers. For instance the protection needed for access control can be implemented on any of the layers. The IPDC model further makes some assumptions about the layer implementation; IP is for instance always used in the transport layer. Contrary to the OSI model, the layers are therefore not fully transparent.

3.2 Link Layer

The central invention in IP Datacasting is to use a digital television broadcast network for transmitting data. It is clearly a mass distribution channel.

There are several digital broadcast standards; most importantly the DVB transmission standards developed in Europe; ISDB-T, which is used in Japan, and ATSC which is used in the United States. All of these are based on the MPEG-2 standards collection that specify how content is compressed, streamed, packetized and multiplexed to form a single digital transport stream. The broadcast transmission standards add error correction processing and modulation schemes, enabling analogue transmission. Additionally, some signalling information is added.

The digital television broadcasting standards were intended mainly for broadcasting video and audio and include detailed specifications of video and audio processing. Although not widely used in digital TV, the standards also include several mechanisms for transmitting data. This makes it possible to use a digital broadcasting network solely for data delivery. IP Datacasting does just that. Later sections describe the alternative data transmission profiles and their applications in IPDC.

This section describes the IPDC link layer, focusing on the standards used in Europe. The common denominator for all major digital television broadcasting standards is the MPEG-2 standards collection. Hence, the overview first explains the parts of MPEG-2 relevant to digital television and highlights the differences between video/audio and data transmission. Then, the European broadcasting standards – the DVB transmission standards – are introduced. Special attention is given to DVB-H, which is specially developed for mobile users. Finally, the DVB data transmission extension, MultiProtocol Encapsulation, is discussed. MultiProtocol Encapsulation facilitates IP Datacasting and is also implemented in the North American broadcasting standard ATSC, but it is not strictly required for IP Datacasting.

3.2.1 MPEG-2

The Moving Pictures Expert Group (MPEG) was originally a working group within the Joint ISO/IEC Technical Committee on Information Technology. They have developed standards for compressing and transmitting audio and video. Their second standard collection, MPEG-2, has been adopted as a base for the three major digital broadcasting standards; the digital video broadcasting standards (DVB) in Europe, Advanced Television Systems Committee (ATSC) in North America, and Integrated Services Digital Broadcasting (ISDB) in Japan. The basic data delivering entity in MPEG-2 is the transport stream (Figure 3.3).

In MPEG-2 [78], a content source is called an elementary stream, and the goal is to transmit an elementary stream to a receiver, minimizing the required bandwidth while maintaining good content quality. Elementary streams are first chunked up into packets. The packet header contains information, such as time stamps, with which the receiver can synchronize different elementary streams (for instance, an audio and a video stream). The streams are accordingly now called Packetized Elementary Streams (PESes). The packetized elementary streams contain some additional information that help the receiver play the packets as a continuous stream and synchronize different streams. Next, the PESes are multiplexed into a transport stream consisting of 188 bytes datagrams with a 4 bytes header. Each transport stream packet carries a Packet Identifier (PID). All transport stream packets that carry a specific elementary stream use the same PID. Several transport streams can

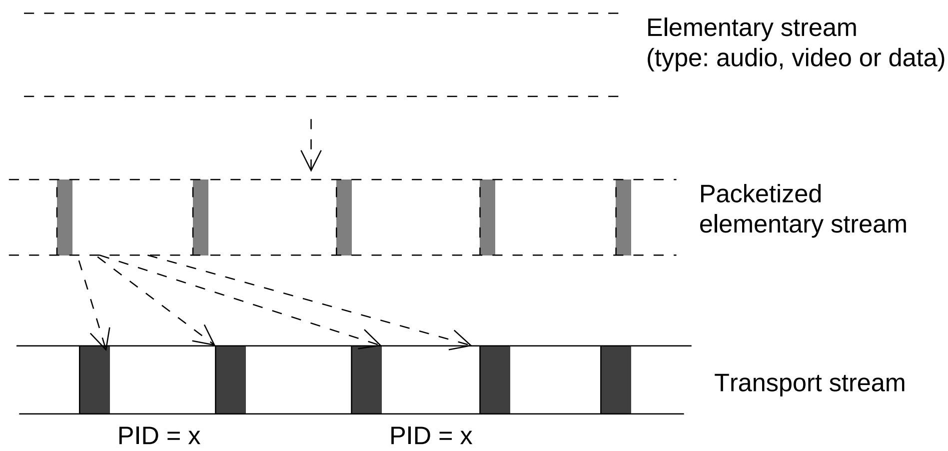

Figure 3.3: The transport stream is the base unit in the MPEG-2 content transmission.

further be multiplexed together. The resulting transport stream is called a *multiplex*.

Elementary streams are always part of a program. A *program* is an MPEG-2 concept that resembles a TV channel; the receiver finds out the currently available elementary streams by checking what each program offers at the moment. In this thesis, the term *program* always refers to an MPEG-2 program.

Stream and program metadata is delivered in Program Specific Information (PSI) tables. There are several table types, one of which can be used to carry any data. As later chapters reveal, IP Datacasting uses tables for carrying data as well as for locating streams. Tables are encapsulated in sections before they are multiplexed into the transport stream, as Figure 3.4 illustrates. Sections are simple structures that help the terminal rebuild the table. The section header contains the payload type and the length of the payload. It also indicates whether the table uses a type specific or a free format.

The Program Map Table (PMT) maps a program to its current streams. It describes the streams and lists their PIDs.

The Program Association Table (PAT) locates the Program Map Table for each program by listing the PID of the PMT of each program. It uses the transport stream PID 0.

The Conditional Access Table (CAT) is used only if a stream uses the

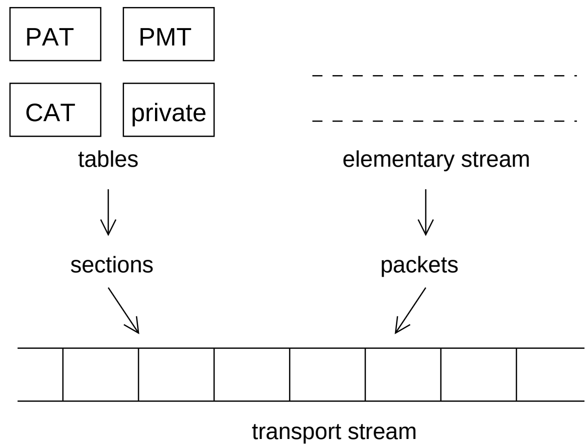

Figure 3.4: Tables are packed into sections before they are encapsulated into the MPEG transport stream.

MPEG-2 scrambling functions. It contains some descrambling information and uses the transport stream PID 1.

The private tables can contain any data.

The PMT, PAT and CAT sections are at the most 1024 bytes while private table sections can be 4096 bytes long [4].

Figure 3.5 illustrates how a receiver locates an elementary stream. The PID is always the key that allows the receiver to filter the correct packets from the transport stream. The PAT is easy to find since it uses the PID 0; to find the PMT the receiver must know the program number.

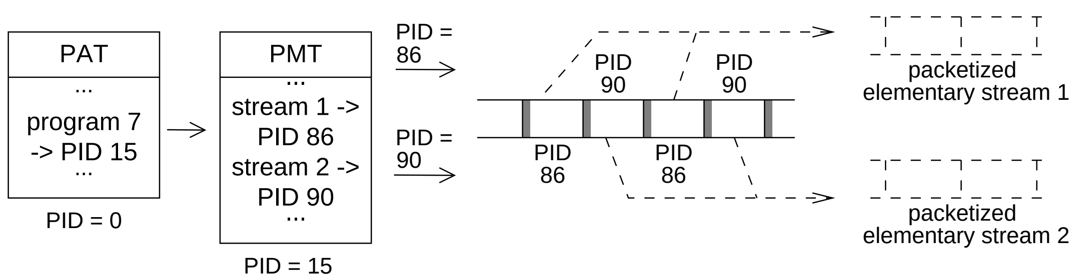

Figure 3.5: The receiver resolves the location of a stream using its program number.

Data Transmission over MPEG Transport Streams

The previous overview revealed that MPEG-2 specifies different ways for treating the content depending on whether it is audio, video or data. Contrary to current digital television broadcasting, IP Datacasting treats all content as data. MPEG-2 defines three techniques for transmitting data, as Figure 3.6 illustrates. In MPEG-2, these techniques are called *profiles*.

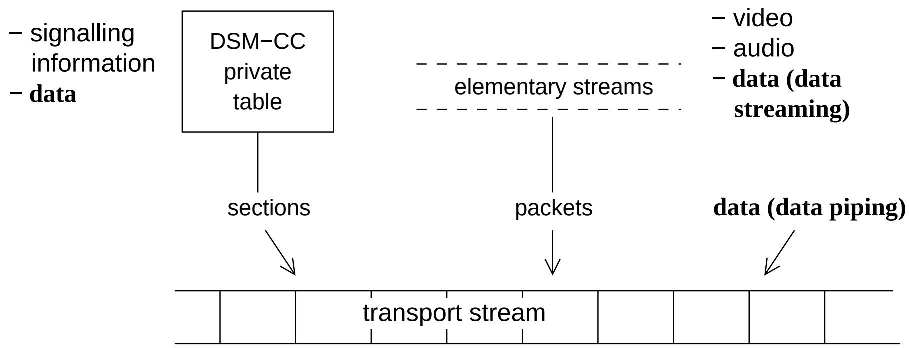

Figure 3.6: MPEG-2 data transmission techniques.

In ***data piping*** data is encapsulated directly in transport stream packets. There is no mechanism for announcing piped data (recall that the PID of elementary streams is announced in PMTs). According to the Tektronix MPEG Guide [78], data piping is mostly used in proprietary systems, for instance for updating digital television terminal adapters. However, the IETF is developing an encapsulation technique – Ultra Light-Weight Encapsulation (ULE) – based on data piping. ULE is described in Section 3.2.3.

Data streaming uses the MPEG-2 data elementary stream type. It enables processing a data stream in a synchronous or synchronized manner. In the first case, timing information is included to reconstruct the original stream pace, while the latter refers to synchronizing the data stream to other streams. Data streaming could be useful for transmitting large amounts of data, but according to the Tektronix MPEG Guide [78], it is used mostly in proprietary systems.

Private tables can carry any data. Contrary to data piping and data streaming, there are several data encapsulation techniques that use private tables. Some of these are presented in Section 3.2.3.

3.2.2 DVB Transmission Standards

The European digital video broadcasting standards are developed in an industry consortium called the DVB Project. The basic unit in the broadcasting standards is the MPEG-2 transport stream [4]. The DVB project has enhanced the transport stream with several error correction mechanisms and specifies modulation formats for different transmission media. Further, it adds new tables so that TV channels and program events can be better described.

The first transmission standard concerned satellite transmission (DVB-S [14]) and was approved in 1994. Later, a standard for cable transmission (DVB-C [18]) followed, and in 1997, a standard for terrestrial transmitters (DVB-T[15]) was developed. DVB-T is the newest and hence the most sophisticated one [24, 13]. In Finland, DVB-T has been chosen as the general digital television transmission standard [23].

A few years back, the idea of using digital broadcasting standards for delivering data to fast moving receivers was introduced. Such receivers have several limitations; small antennas directly attached to the terminal and limited power supplies. Further, the Doppler Effect caused by the motion impedes signal processing. A working group within the DVB Project started investigating how their current standards could be used to support mobile reception. Although they concluded that DVB-T in some modes does support mobility, they decided to develop a new standard that better would deal with all these limitations. The new standard, Digital Video Broadcasting for Handheld devices, DVB-H (previously also called DVB-M and later DVB-X), extends the DVB-T standard.

The objective of this section is to describe DVB-H. Yet, the presentation starts with the earlier DVB transmission standards, since the pre-modulation processing stages of the MPEG transport stream are similar. The focus then switches to DVB-T, explaining the physical modulation and the transmission modes. The background of DVB-H is then laid out and the last subsection finally introduces DVB-H itself.

Announcing Streams

The MPEG-2 tables give rather limited alternatives for describing the broadcast content. The DVB Project therefore adds several tables. The most interesting ones are the service description table and the event information table [21].

The Service Description Table (SDT) complements the Program Association Table (PAT). The PAT helps the terminal find a program, such as EuroSport, by giving the PID of the corresponding channel PMT, but the SDT describes the program. It may contain the program name in several languages, couple the program to a program group, list which countries it is available in, divide the terminal screen into several screens, or announce a replacing channel in the same or another network. The SDT is typically rather static. (Note that 'service' in the name Service Description Table is analogue to the MPEG term 'program'.)

The Event Information Table (EIT) on its part complements the Program Map Table (PMT). The PMT helps the terminal find the streams that a program currently contain, but the EIT describes the current and upcoming content of the program. It contains starting and ending times for specific events such as a movie or the first half of a soccer game. The EIT also reveals the current status of the event (not running, running, paused, soon starting, and undefined). It can also contain other information such as age-limit recommendations. The EITs are updated often – for instance, whenever an event starts or ends a new EIT with the current status is sent.

Further new tables are the network information table that lists information about the physical network; the bouquet association table, which describes channel groups; the running status table, which informs that the state of an event has changed; the time and date table and the time offset table whose names are self-explanatory; the stuffing table used to fill up sections; and the selection information table and the discontinuity information table which are used when saving streams, for instance in a recording set-top box [21].

Preparing the Signal for Modulation

The MPEG-2 transport stream is as such not robust enough against transmission errors. Regardless of the transmission medium, it is necessary to add some forward error correction and choose a suitable signal element constellation for the modulation. All three DVB transmission standards treat the transport stream similarly [77] and three processing stages can be identified:

Energy dispersion and impulse noise protection [15]. The transport stream is first randomized in order to avoid long stretches of 0's or 1's and hence distribute the signal energy more evenly. The Reed-Solomon error correction code is performed on each 188 bytes transport packet, adding 16 bytes to each packet while protecting against at the most 8 bytes of error. The packets are then interleaved to decrease the impact of impulse noise.

So far, the transport stream is processed in the same way in all three DVB transmission standards [77].

Inner convolution coding. Next, the stream is convolution coded for additional error protection. There are several protection levels that offer a payload to error protection overhead ratio of 1/2, 2/3, 3/4, 5/6 or 7/8. All modes are based on a specific convolution code that adds one protection bit for every bit of payload. In the 1/2-mode the code is used as such. In the other modes the code is punctured; some predetermined bits are discarded at the sender, forcing the receiver to insert bits at the corresponding positions but mark them as deletions. The deletions have little impact on the transmission quality since the transmitted information helps correct or ignore them; the precise choice of which bits are deleted has been decided through testing in the DVB Project [73]. Stott [73] provides a comprehensive explanation of punctured codes and their impact on the DVB-T encoding combination. Inner convolution coding is used in DVB-S and DVB-T [14, 15] but not in DVB-C [18], since cable transmission is less exposed to noise. In DVB-T, the resulting stream is further interleaved.

Choosing signal element constellation. At this point, there is no need to modify the bit stream, and the bits are mapped onto signal elements. DVB-S uses QPSK [14] which delivers two bits per signal element. The other standards offer alternatives; DVB-C supports QAM modes from 16 up to 256 bits per signal element [18], while DVB-T supports QPSK, 16-bit and 64-bit QAM [15].

The signal is next modulated for the transmission on the physical *carrier*. This stage naturally differs between the different transmission media; the DVB-T modulation is described in the next section.

DVB-T

The DVB-T transmission standard introduces error protection and correction mechanisms, modulation schemes, and network design.

First of all, DVB-T [15] divides the signal onto several thousand orthogonal subcarriers using Coded Orthogonal Frequency Division Multiplexing (COFDM). This makes it possible to insert a long pause or guard interval between two consecutive signal elements. Echoes then have time to arrive before the next signal element starts. If a single carrier was used, the guard intervals would constitute a large portion of the total transmission time. However, when the signal is divided on subcarriers, the signal period increases [73] since the capacity of each subcarrier is smaller than that of the entire channel. Hence, the ratio between the guard interval and the signal is acceptable.

Using several subcarriers also protects against narrowband interferences within the channel, since they affect only the subcarriers using the disturbed frequencies.

The six transmission parameters that have to be chosen are the number of subcarriers, whether a hierarchical signal is used, the inner code rate and the signal constellation which are discussed in the previous section, and the length of the guard interval. Table 3.1 summarizes the DVB-T parameters.

The number of subcarriers mainly affects the guard interval and the signal tolerance to the Doppler Effect. The more subcarriers, the longer the symbol length, and the longer guard intervals are practically possible. There are two modes; one using 6817 carriers (the 8k mode) and one using 1705 carriers (the 2k mode). The name 8/2k refers to the Fast Fourier Transform (FFT) algorithm that is used to calculate the symbols. Since the algorithm is remarkably more efficient if the number of carriers form a power of two [34], virtual carriers are inserted during the computation, resulting in about 8000 (8k) and 2000 (2k) carriers.

8k-mode cells are significantly larger than 2k-cells, since the longer guard intervals and the longer signal periods of the 8k-mode decrease the negative impact of delays and attenuation. The cell size is more or less directly proportional to the guard interval. The amount of carriers also influences the mobility. When the receiver moves, the Doppler Effect causes a slight shift in signal frequency that is proportional to the receiver speed. The farther apart the carriers are, the smaller the Doppler Effect is in proportion to the subcarrier width and the less it disturbs the signal.

The length of the guard interval is always a fraction of the total symbol length. The available parameters are 1/4, 1/8, 1/16 and 1/32. The longer the guard interval, the longer echoes the signal tolerates.

A hierarchical signal consists of two different signal streams [15]. Typically, a stream is split up into a small robust stream and a more error-prone stream with higher bit-rate. The transmission is hence adapted for as well simple as more sophisticated receivers. In practice, the hierarchical mode is hardly used.

Not all of the subcarriers carry data; some are used for among others synchronizing, and a few are used for Transmission Parameter Signalling (TPS) to announce the signal constellation, whether the signal is hierarchical, the code rate and the guard interval. Some bits are reserved for future use. As

Physical channel	8 MHz (also 6 MHz or 7 MHz possible)
COFDM mode (number of subcarriers, subcarrier width, signal element length)	8k (6817, 1116 Hz, 896 μs) or 2k (1705,4464 Hz, 224 μs)
Guard interval (8k/4k duration)	1/4 (224/56 μs), 1/8 (112/28 μs), 1/16 (56/14 μs) or 1/32 (28/7 μs)
Inner code rate	1/2, 2/3, 3/4, 5/6 or 7/8
Signal element constellation	QPSK, 16-QAM or 64-QAM

Table 3.1: The DVB-T transmission parameters.

the next section shows, DVB-H takes some of these reserved bits into use.

It is possible to build either single frequency or multifrequency networks [16]. In single frequency networks, adjacent transmitters use the same frequency. Since the COFDM modulation is tolerant to echoes, the signals do not disturb each other. Rather, the receiver can combine the signals, which increases the signal strength in the cell boundaries.

Alternatively, in the multifrequency network adjacent cells can use different frequencies. The transmitter signals are then independent from each other since they do not interfere with each other. Contrary to single frequency networks, content can be cell-specific.

The DVB-T capacity ranges from 4,98 Mb/s (4QAM with code rate 1/4 and guard interval 1/4) to 31,67 Mb/s (64-QAM with code rate 7/8 and guard interval 1/32) [16]. In Finland, the 8k-mode with a 2/3 code rate, 64-QAM modulation and 1/8 guard interval, resulting in 22,12 Mb/s [77].

DVB-H

Following DVB practices, work on the DVB-H specifications started by developing commercial requirements for the new standard. A first draft of the commercial requirements was specified, and the technical module started working on the actual technical specification in summer 2002 [32]. The standard specification will be completed in the beginning of 2004, validated during 2004 and finally approved in the end of 2004. The DVB-H standard will bring 3 new features: Time slicing, forward error correction on the MPE layer and a new 4k Coded Orthogonal Frequency Division Multiplexing mode [33].

The capacity of a DVB-H channel is about 11 Mb/s.

To cope with the more complex transmission parameter signalling, a few of the reserved TPS bits are taken into use. Jukka Henriksson et al. [33] describe these features:

Time slicing. The largest problem of DVB-T is that it requires the receiver to process the broadcast stream constantly. According to a Nokia IPDC-terminal expert [40], this is simply impossible for handheld devices using current-technology batteries. Contrary to conceptions of phantasy literature [63], time slicing is a straightforward maneuver. The broadcast stream is divided into slots of some hundred milliseconds, and the receiver turns itself off while waiting for the next burst. This requires a 2 Mb buffer in the transmitter as well as the receiver.

4k COFDM modulation mode. As the previous section reveals, the COFDM modulation mode determines the amount of subcarriers and hence the cell size and the highest possible receiver speed. The 4k mode provides wide enough spaces between the subcarriers for fast mobility, while the signal element period is long enough for cells of about 30 km. To add the 4k modulation and demodulation mechanism is cheap since it requires minor changes to DVB-T 2k/8kmodulators and demodulators. Signals using different COFDM modulation modes can naturally not be multiplexed together since they use different physical channels.

Forward error correction in MultiProtocol Encapsulation (MPE-FEC). MultiProtocol Encapsulation is a technique for encapsulating datagrams in sections. DVB-H adds some forward error correction on the MPE-level; both MPE and MPE-FEC are described in the next section.

3.2.3 Extensions to MPEG-2 Data Transmission

It is possible to transmit data using over a MPEG-2 transport stream, as explained in Section 3.2.1. However, the MPEG-2 data profiles are too general in several common situations. There is a collection of specifications, the Digital Storage Medium Command and Control (DSM-CC), which extend the private table format for different data transmission purposes. First, they specify a quite simple private table format for encapsulating data. They also define a framework for managing data as entities or objects. The broadcasting standards (primarily DVB and ATSC) incorporate the DSM-CC specifications and accordingly provide similar data transmission profiles. There is also an initiative within the IETF for developing an efficient MPEG-based encapsulation technique for IP datagrams.

This section describes MPEG-2 extensions for transmitting data over MPEG-2 networks. The overview starts with the DVB implementations of the DSM-CC based techniques – MultiProtocol Encapsulation (MPE) and carousels. Of these two, only MPE is simple enough to work as a delivery mechanism for IP traffic – but for completeness, the carousels are described shortly. Since this thesis focuses on the European standards, the overview does not cover DSM-CC implementations in the other broadcasting standards. However, all three major digital video broadcasting standard families (DVB, ATSC, ISDB) support carousels [78], and ATSC further includes MultiProtocol Encapsulation [36]. Next, a current IETF specification proposal, Ultra Lightweight Encapsulation, is described. A comparison of the specifications concludes the section.

Carousels [78] resend data at specific intervals. Data carousels are simple and treat the content as modules. There is no information about a module; the receiver is expected to know what to do with it. Modules can be grouped together. In object carousels content is treated as objects that each can be identified and has some properties. The carousel contains a service gateway which is a list of the carousel objects. The carousel can be interactive, allowing the receiver to retrieve objects on-demand, or simply send all objects periodically, in which case the receiver reads in objects selectively. Object carousels use the Broadcast Inter-Object Resource Broker Protocol.

MultiProtocol Encapsulation (MPE) [22] mainly adds a header and a trailer to any datagram. The encapsulated datagram follows the DSM-CC private table format, allowing it to be placed in a section. The MPE header mainly contains the MAC address of the receiver. For IP datagrams, this encapsulation suffices – other datagrams are additionally equipped with a Logical Link Control/SubNetwork Access Protocol (SNAP) header that conveys the type of the datagram [22].

The main benefit of MPE is that it is a light-weight method for encapsulating IP datagrams. Further, the MAC-level addressing allows for targeting specific devices. Sections can even be filtered with standard reception hardware, as Grundström et al. [29] explain, since the two least significant MAC-address bytes, are placed in the beginning of the section. Using IP multicast-to-Ethernet mapping ([12, 9]), the last two MAC address bytes contain the last two IP address bytes. These often suffice to identify a device. Figure 3.7 illustrates how the MAC address is placed in the header, and how the LLC/SNAP header is inserted between the MPE header and the payload.

The MPE overhead is 16 bytes without the LLC/SNAP header (a 12 bytes header and a 4 bytes checksum). This leaves 4080 bytes for the IP datagram (including the IP header), since the section can contain a total of 4096 bytes.

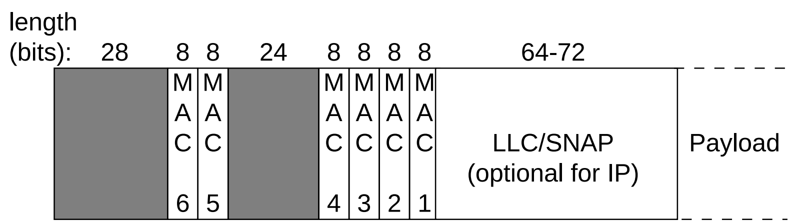

Figure 3.7: The MPE header contains the receiver's MAC address and an LLC/SNAP header with the payload type (the gray areas are other header fields).

MPE streams are announced in tables similarly to normal audio and video streams. All streams belong to a program and hence a stream is listed in its Program Map Table as an MPE stream. The MPE stream can also be announced in the DVB Service Description Table or the Event Information Table [21]. However, this way of announcing streams is difficult to combine with IP addressing; the terminal has to monitor and scan all MPE sections to find out the IP destination address of the encapsulated datagram. This is impossible for handheld receivers due to their limited power supplies. Therefore, a better alternative is the IP/MAC Notification Table (INT) that the DVB Project introduced in May 2003. The INT is not mandatory, but the DVB Project recommends that it is used whenever using MPE.

With INT, MPE streams are still organized into programs and program components. However, the INT maps sets of addresses to specific streams [22]. It now takes three table lookups to find the transport stream PID corresponding to a specific IP address: First, the INT maps an address to a program number and a program component. The terminal then consults the Program Association Table to find the corresponding Program Map Table, where the terminal finds the PID of the one program component it is looking for. The addresses do not have to be IP addresses; the INT also supports smartcard identifiers, MAC addresses, serial numbers, and IP netmasks. With IP addresses, it is possible to specify the destination addresses as well as the sender address.

The INT was developed to support IP platforms that range over several access media; a platform could for instance use two MPEG-2 transport streams and a dial-up line. A transport stream may contain several platforms; for instance one per Internet Service Provider. It is possible that a platform provides services only for its own customers. The INT would then list also the identifiers of the platform customers – for instance, their serial number – and rely on the terminal to perform access control based on the INT. It is difficult to describe a service using the INT and the other DVB tables, so IPDC uses an electronic service guide for describing services (see Section 3.4.1).

How frequently the INT changes, depends on the network. Some network operators can use a static INT with a handful of IP streams that are used only from time to time. The INT can also be updated frequently if new IP streams are added often.

The DVB Project considered several alternative techniques before deciding on the INT. According to Väre [82], one alternative was to modify the existing tables so that it would be easier to list IP addresses. Alternatively, the transport stream could contain an IP Control Channel Message with which the sender could describe the network topology, a specific service, or accessibility of services. Väre [82] himself proposes using service description streams; a program that contains a data service component would also contain a service description stream. The descriptions would use the Session Description Protocol (SDP).

The DVB Project will further introduce optional Forward Error Correction at the MPE-level to cope with the varying reception conditions of mobile receivers. The error correction, called MPE-FEC, virtually interleaves the MPE sections before calculating checksums and sends the checksums in separate sections. This way, simple terminals can easily skip the forward error correction. The terminal uses the IP datagram checksums to find which datagrams have been corrupted, and can then correct one byte for each byte of the MPE-FEC checksum. Without the IP datagram checksums, only half as many bytes can be corrected. The MPE-FEC overhead is 25%, but MPE-FEC improves the protection against impulse interference and the Doppler Effect. It requires a 2 Mb buffer in the transmitter and the receiver, but the same buffer can be used for time slicing.

Ultra Lightweight Encapsulation (ULE) [28]. Fairhurst and Collini-Nocker [28] draft a simple datagram encapsulation technique based on data piping. In ULE, a short 4 bytes header and a checksum are added to the datagram which is packed into MPEG transport stream packets. Since ULE does not use sections there are no restrictions on the datagram length. The MPE header is three times as long, but on the other hand, the ULE header mainly reveals the type of the datagram and the datagram length (Figure 3.8). Currently, ULE can only carry IP and frames that are bridged between two networks. It is possible to include the receiver's MAC address directly after the ULE header. Unlike MPE, ULE does not reorder the address bytes, which means that the last two address bytes do not fit within the first 8 bytes. Hence, at least older receivers will not be able to hardware filter ULE datagrams. Yet, when transmitting IP datagrams without MAC address filtering, ULE is more efficient than MPE.

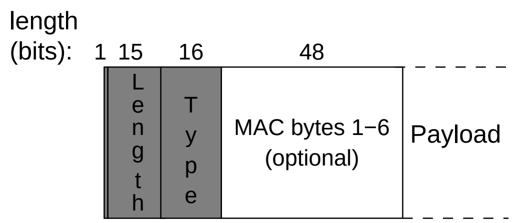

Figure 3.8: The ULE header.

Comparison. The carousels are intended for object-oriented data transmission and are therefore difficult to use as a general delivery mechanism for IP. MPE and ULE, on the other hand, are both optimized for IP. Their data overhead is similar. In the example in the beginning of the thesis, MPE would cause 3.9% overhead: 157 bytes for 4032 bytes of stream or file content causes, assuming that a section starts directly after the previous ends in the transport stream. (Here, one UDP datagram fits into one IP datagram which fits into one MPE section. The overhead consists of 8 bytes UDP header, 40 bytes IPv6 header, 12 bytes MPE header and 4 bytes MPE trailer, 23 transport stream packets headers of 4 bytes each, and one byte for pointing to the beginning of the section in the first transport stream packet. The calculation does not include the protocols above UDP since they depend on the content type and may use datagrams of variable size, nor does it include the MPE-FEC.) In ULE, the overhead is 3.7% with 149 bytes overhead for 4032 bytes of stream or file content. If MAC-addressing is used, the overhead increases to 155 bytes or 3.8%. The ULE datagrams are yet somewhat more simple to process than the MPE sections.

The overhead increases if transport stream packets are stuffed so that a datagram starts at the beginning of the packet. This is true especially if the MPEG-2 network is used for any type of traffic, including connection-oriented protocols such as TCP, since they cause control traffic consisting mostly of short messages. ULE always places a datagram directly after the previous one, but some MPE implementations use stuffing. Clausen et al. [6] report overhead values of 13-15% for MPE-encapsulated traffic in the general case, while Fairhurst et al. [27] note that control traffic, that cause several short messages, can cause overhead of up to 500% when using MPE. However, most IPDC connections will be unidirectional and there will be little control traffic. The typical overhead is therefore around 4% for MPE as well as ULE which is little already compared to MPE-FEC overhead of 25%.

ULE was one of the first efforts within IETF to define an IP encapsulation for MPEG-based transmission networks. The IETF has now established a working group [36] for developing a general-purpose IP encapsulation technique for MPEG networks. It will likely draw on ULE as well as MPE, and since MPE already is commonly in use, it will be compatible with MPE [36]. The group will also work on a general protocol for resolving IP-to-MPEG address mapping [27].

MPE will most likely be used at least in the early stage of IPDC, since ULE is not finished yet and provides only small benefits compared to MPE. The new IETF encapsulation specification is currently only an idea and will therefore neither be an alternative to MPE for several years.

3.3 Transport Layer

IP Datacasting – as the name itself reveals – uses the Internet Protocol as the basic data transmission protocol (e.g. [31, 80]). This has benefits as well as disadvantages. However, IPDC therefore also uses an IP-based protocol stack. This is a new standpoint compared to the digital television transmission standards that integrate all specifications needed to transmit content, ranging from the physical radio parameters to content compression formats. Especially handheld terminals benefit from a less rigid standard collection since terminal technology as well as content compression algorithms develop fast and parts of the standard set may need to be upgraded every few years.

This section describes the IP Datacasting transport layer protocols. These are divided into three groups; transport protocols, streaming protocols and file transfer protocols. As Figure 3.9 illustrates, the Internet Protocol is the common data transmission protocol. It is used together with the User Datagram Protocol to provide data sessions. Files are transferred with the File Delivery over Unidirectional Transport (FLUTE) protocol, based on Asynchronous Layered Coding (ALC), while streams use the Real-time Transport Protocol (RTP) and the RTP Control Protocol (RTCP).

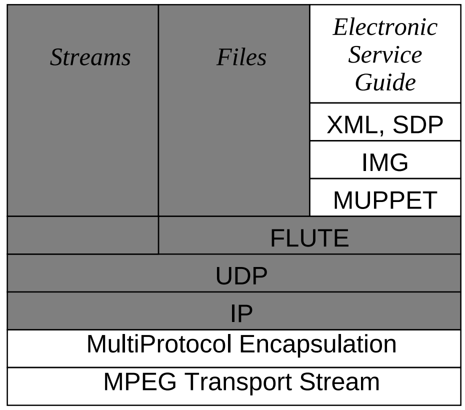

Figure 3.9: The IP Datacasting protocol stack (the transport layer protocols are shaded). Adapted from [31].

3.3.1 Transport Protocols

The Internet Protocol (IP) [62, 10] is designed for delivering datagrams over different networks. Its most important tasks are providing globally unique addresses, allowing fragmenting and refragmenting of datagrams, and error detection. However, the link layer already provides error detection and supports MAC addresses such as Ethernet addresses, and fragmenting is easy to avoid by choosing a suitable datagram size. The benefits come mostly from stream-lining IP Datacasting with other data transmission technologies. Since most terminals already include an implementation of the IP protocol stack, is it easier for application developers if they can treat the DVB network as another IP-based access network. Either IPv4 or IPv6 can be used.

In IPDC, a TV program corresponds to an IPDC multicast group. When a user chooses a specific piece of content, the terminal IP implementation joins the corresponding multicast group by adding the address to a list of addresses that it listens to. Using IP multicasting also makes it easier to distribute the content for delivery to the different encapsulators and transmitters (see Section 3.2) since the distribution paths can be set up dynamically with a multicast routing protocol.

The User Datagram Protocol (UDP) [61] is a transport protocol. IP is typically used in conjunction with a transport protocol. Since the broadcast access link is unidirectional, it is sensible to use the User Datagram Protocol (UDP) which does not require the terminal to confirm the transmission. UDP simply adds a header containing the source and destination ports or access points, the length of the datagram and a checksum.

3.3.2 Streaming Protocols

The Real-time Transport Protocol (RTP) [69] is a simple protocol for delivering streams. It has two important features; a sequence number that helps re-assembling the received datagrams in correct order and detect missing blocks; and a timestamp, which enables synchronizing separate streams. RTP is mostly used in an interactive mode where the sender adjusts the speed and the content quality based on receiver feedback. It can, however, also be used in a unidirectional environment, but how the sender then optimizes the transmission parameters is out of the protocol scope. Market surveys and other statistics then naturally form valuable sources of information. The RTP Control Protocol (RTCP) is used together with the RTP to negotiate and give feedback on (in the interactive case) or announce (in the unidirectional case) transmission parameters.

RTP is intended for any type of payload, and it is therefore mostly used together with a protocol extension for a specific payload type. For instance, to transmit MPEG-4 elementary streams, the sender would use the RTP Payload Format for Transport of MPEG-4 Elementary Streams [11].

3.3.3 File Transfer Protocols

The most appropriate IPDC file delivery protocol may be the new File Delivery over Unidirectional Transport (FLUTE) protocol. FLUTE extends the Asynchronous Layered Coding (ALC) protocol, and hence, this review first presents ALC before moving on to FLUTE.

Asynchronous Layered Coding (ALC) [46] is designed for transmitting binary entities over multicast networks. Here, the binary entity is called an object. The goal of ALC is to support large scale transmissions with any number of receivers. The receivers may have different capabilities and join and leave the session at different times. Hence, ALC encodes the content using a Forward Error Correction (FEC) scheme in such a way that a receiver can reconstruct the content as soon as it has received a certain number of parts. First of all, this provides an alternative to carousels. There are FEC schemes that can transform a piece of content into a large number of encoded parts and a sender broadcast a specific content a long period of time using different parts every time. If a receiver misses one part, it can use the next one rather than having to wait a full carousel cycle. Second, receivers can adjust their own reception rate if the sender splits the content over several channels. If an object of 200 Mb is divided over 4 channels each carrying 200 kb/s, a slow receiver joins one channel and receives the whole object in 977 seconds, while a faster receiver joins all four channels and receives it in about 245 seconds. Several forward error correction schemes cause some overhead but this is negligible if the object is large.

ALC does not distinguish between different types of content; all content items are treated as objects. The sender must always distribute a session description – ALC does not define how – and the receiver must somehow get the description before joining a session. ALC operates on top of UDP.

Architecturally, the ALC protocol consists of three building blocks. The layered coding transport building block [47] defines the transport sessions and channels, and the forward error correction described above is managed in a building block of its own [48]. Further, the Internet Engineering Task Force requires reliable multicasting protocols, like ALC, to use some congestion control [53]. In the general multicasting situations, the multicast traffic may grow fast and starve other types of traffic. This is not a concern in IPDC since the encapsulator can take care of coordinating the traffic from the different content producers. Yet, ALC is intended as a general-purpose multicast object delivery protocol. There is currently no suitable congestion control building block available and ALC is therefore still an experimental IETF Request For Comments, waiting to enter the standards development process.

File Delivery over Unidirectional Transport (FLUTE) [60]. ALC is scalable and robust but sending files with it is difficult, since ALC does allow the sender to describe its objects. FLUTE extends ALC to support file delivery. It specifies a file delivery table that describes the files in a session, and an ALC header extension that carries the table or parts of it. The file delivery table lists at least the uniform resource identifier and the ALC object identifier of each file. The sender can also include other information in the table such as the file length and type, forward error correction parameters and a checksum.

3.4 Session Layer

There are certain functions in IPDC that are clearly related to specific sessions. These functions belong to the session layer. In this thesis, the session layer has three tasks: It announces the available services; it protects them so that a user cannot access a service unless he has paid for it; and it handles service payment. Some of these functions stretch over several layers – for instance access control can use protection techniques from different layers – but since these three functions clearly relate to sessions they are conceptually included in the session layer.

This session describes these three functions and reviews related standards and protocols. Figure 3.10 illustrates the protocols in the IPDC protocol that primarily belong to the session layer; additionally, different layers may provide some specific session functionality, such as content encryption.

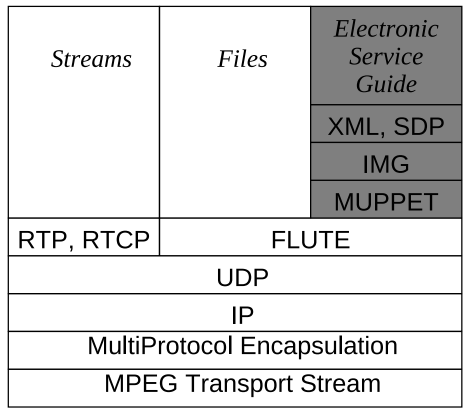

Figure 3.10: The IP Datacasting protocol stack (the transport layer protocols are shaded). Adapted from [31].

3.4.1 Service Announcement

It does not matter how great an IPDC service is, if the user does not know about it or does not know how to tune his receiver to it. The IP Datacasting Forum [80] presents the idea of an *Electronic Service Guide (ESG)* that would carry all the service information a user needs to start using the service. The ESG would for instance list service names, their prices, their starting and ending times, their IP multicast address, and protection parameters.

There is no specific Electronic Service Guide, but the Internet Engineering Task Force (IETF) is developing a framework [57] for making and using guides for different Internet-based events – remote collaboration sessions, net meetings, etc. They also have sketched a distribution protocol using multicasting over unidirectional links. This Internet Media Guide (IMG) framework could form the basis for the ESG.

This section describes the Internet Media Guide and the IMG transport protocol MUPPET.

The Internet Media Guide (IMG) [50] describes one or several services. It contains as well technical information, such as different transmission parameters, as conceptual information that helps the user choose between different services. There are two parts that are mainly intended for the human recipient; the relational part and the content part. The relational part bundles and categorizes services or service providers. Furthermore, it contains information on how to find additional service information or value-added services such as ratings. The content part allows for describing individual pieces of content, for instance by naming them and describing the content genre. It is possible to give age recommendations at several levels ranging from a service category (e.g. action movies) to an individual piece of content. The IMG further carries session transmission parameters, most importantly the session address (i.e. the destination IP address, the transport protocol type and port number), protection parameters, codec parameters and the session starting and ending times.

The guide format is still not decided, but the IMG developers [50] are currently considering using MPEG-7 (for the content part), SDP or SDPng (for the session part), XPath or XPointer (when referring to parts of a guide) and XML.

Internet Media Guide Unidirectional Point-to-Multipoint Transport Protocol (MUPPET). The IMG typically describes several upcoming services, programs or events. Some service descriptions can be valid for long times while others are published late and soon become outdated. Therefore, as Nomura et al. [57] describe, the guide can be sent periodically, or receivers can explicitly request a guide or specific parts of it. MUPPET [49] defines how the guide is transmitted. The sender can transmit either a complete guide, a part of the guide or a pointer to a location that has changed within the guide. He always sets up a MUPPET session consisting of one channel for each of the transmission modes he uses (complete guides, partial guides and pointers). The channels are FLUTE file delivery channels.

3.4.2 Access Control

IPDC content is broadcast to all users that are within the range of the transmitters. To enable charging for services, the content must be protected so that only paying customers can access it. Whatever information is needed to access the services is packed into rights objects that are sold to the customers. There are several different protection schemes that could be used in IPDC; some of them are presented here.

MAC-level addressing. The MAC-level addressing also allows for targeting specific devices [22]. In principle, MAC-level multicast addressing could be used for digital rights management. This is still difficult to implement since the terminal then is responsible for most access control functions; For instance, it is responsible for keeping track of which groups the user may access and checking if the user's group subscriptions are valid. Further, the scheme completely relies on the terminals; dishonest terminals can access any content. When using MAC-level addressing, there are no rights objects; the users simply register for a service.

DVB Conditional Access (CA) [13]. The conditional access scheme, developed by the DVB Project, consists of three blocks. Only one of these, the Common Scrambling Algorithm (CSA) is standardized, while the Subscriber Management System and the Subscriber Authorization System are developed independently by several commercial organizations. The common scrambling algorithm can apparently be replaced by any other protection algorithm. Several commercial CA implementations use proprietary algorithms and two interviewed IPDC experts noted that most CA systems are rather easy to compromise [38, 39].

To deal with the variety of CA providers, DVB has promoted two different mechanisms. In Multicrypt, the receiver uses different CA implementations; each CA implementation may require a smart card of its own. Simulcrypt allows the user to access several service providers' offerings using one single CA system. The service providers must then agree on supporting several CAs. For this purpose DVB has produced an agreement model, the "Code of Conduct".

DVB CA protects the content at the transport stream level. It is also possible to protect packetized elementary streams, but as described in Section 3.2.3, data encapsulation methods that use sections or the transport stream directly are more attractive than transmitting data using packetized elementary streams.

The default scrambling algorithm uses a control word to scramble the content. Every time the control word changes, the control word is encrypted with a service key and sent as an Entitlement Control Message (ECM). The control word changes every few seconds. Each user further has a user key. The service key is encrypted with each user key individually, resulting in one Entitlement Management Message (EMM) per user key. These are sent often enough so that a user learns the service key soon after starting to watch a service – e.g.

every ten seconds. The terminal finds the messages using the DVB tables; the Conditional Access Table (CAT) points to the EMM while the Program Map Table (PMT) reveals the location of the ECM [4]. Sending one message per user key naturally limits the number of users.

This CA model does not use any rights objects; users simply register for a service. Alternatively, the EMMs can be sold as rights objects.

IPsec [42]. IPsec is a framework for securing IP traffic. It provides access control, connectionless integrity, authentication of data origin, rejection of replayed packets, confidentiality and limited traffic flow confidentiality. The last service is accomplished by tunneling traffic between two networks, in which case the addresses of the endpoints can be encrypted and packet stuffing hides the exact length of a message.

The cornerstones of IPsec are security policies and one-directional security associations (SA). The security policy expresses how different traffic is to be protected. A security association is formed for each specific security service required, for instance a specific type of encryption. The security association defines what security protocol, what algorithm and what algorithm specific parameters are used. It is possible to apply several security services at once; in that case several security associations are combined.

IPsec defines two security protocols: The Authentication Header (AH) ensures the integrity of the body and parts of the header. The Encapsulating Security Payload (ESP) secures only the body, offering confidentiality, integrity or both. AH as well as ESP provide anti-replay protection. Confidentiality is often enough in IPDC since the sender only wants to prohibit users that have not paid for a service to access it. Integrity is often not as important since integrity checks in the different transport protocols detect most errors and it is difficult to make specific integrity attacks since each sender needs a DVB-H transmitter. However, if an IPDC service contains sensitive data such as electronic tickets, the data may need additional integrity protection. In both cases, ESP provides sufficient protection – AH can be used to ensure the integrity of parts of the header when sending extremely sensitive data.

IPsec is specified for unicast purposes. Negotiating the security associations is more complex in a multicast or broadcast environment since several parties participate in the negotiation. In the general case, using IPsec with multicasting requires a specific negotiation system [42], such as the one presented by Hardjono and Weis [30]. There is further no general solution for managing IPsec keys – one of the latest achievements in that area is the new pki4ipsec working group that IETF has established for developing a public key infrastructure for IPsec [37]. Negotiating security associations with all receivers in an IPDC cell before a session starts is naturally difficult, if not impossible, especially since new users can enter a cell at any time during the service. It is therefore likely that the sender will not negotiate the parameters. Instead, he can simply create a security association for his service and sell the security association as a rights object.

The Secure Real-time Transport Protocol (SRTP) [3]. SRTP is developed for securing Real-time Transport Protocol (see Section 3.3.1) traffic. It ensures the confidentiality of the payload and the integrity of the whole datagram, and protects against replay attacks. A session is encrypted or authenticated or both. There are six different session keys and salting keys, all of which are derived from one master key and one master salting key. Baugher et al. [3] claim the way the session keys are derived from the master keys to be secure. Before a session starts, the sender and the receiver or receivers must exchange a master key – SRTP does yet not specify how. In IPDC, the service master key can be sold as a rights object. The SRTP specifications define a few mandatory algorithms; other ones can be added by extending the SRTP specifications. SRTP is at the time of this writing an IETF Internet Draft. The latest draft version has been approved as a proposed standard [79]; at the time of this writing, the new request for comments is not available.

MPEG-21. The Moving Graphics Expert Group offers a framework that gives broad support for managing digital objects, including the immaterial property rights coupled to them [78]. In the development stage of this standard, called MPEG-21, Soininen [71] appreciates that it will provide the most flexible DRM capabilities, as it can be applied on any type of content. With MPEG-21, it is possible to give very fine-grained access conditions [67] such as "This piece can be played twice between 2003-12-04 14:00 and 2003-12-06 14:00". The MPEG-21 standard will consist of about 12 parts; two of these are published and six were in July 2003 still under development.

OMA DRM v1.0 [58, 59]. The OMA DRM scheme is an application-level protection mechanism. There are three modes. In the first mode – forward lock – the terminal simply ensures that the content is not forwarded. According to a Nokia terminal expert [40], only this mode is currently implemented in the OMA DRM compliant mobile phones. In combined delivery, the terminal additionally enforces a few different access conditions such as presenting the content only a predefined number of times. There may be some use for combined delivery in IPDC, for presenting service teasers, for instance. However, in separate delivery the content is encrypted and the decryption parameters are delivered separately in a rights object. The content can be forwarded but the rights objects cannot. The OMA DRM scheme relies on the terminals not to distribute the content. Although dishonest users with modified terminals can order the rights object once and then distribute it, separate delivery is the strongest of the three modes.

In principle, any mechanism can be used for distributing the rights object. The terminal must at least support unconfirmed Wireless Application Protocol Push Over-The-Air (WAP Push OTA) [58]. WAP Push OTA is designed for low-capacity bearers without TCP/IP support such as Short Message Service (SMS). However, almost any link that supports the Wireless Datagram Protocol (WDP) [83], including TCP/IP-supporting bearers such as General Packet Radio Service (GPRS) links, can be used.

Other DRM schemes. Application-specific digital rights management mechanisms will most likely be used to some extent. Often, application specific access control is used to complement lower-level access control. Further, there are some hardware based schemes, such as the TV Anytime Rights Management and Protection scheme.

Comparison. Most of the proposed schemes can be used for very simple access control; a user either is or is not allowed to access a service. Using MAC-addressing is the simplest scheme. However, it is easy to bypass since it completely relies on honest terminals – it is not difficult to manufacture dishonest terminals that accept any MAC address. With the other schemes, the content can be encrypted. The CA scheme could be used for all type of content since it protects the transport stream directly. However, experience shows that the scheme is easy to compromise. Additionally, the EMMs cannot be prepared in advance since new users can register for a service at any time, and if there are many users, the EMM traffic uses a large part of the capacity. IPsec can be used together with IP and practically all services are IP-based. With IPsec, it is possible to choose the protection algorithm and the algorithm parameters individually for each service. It is therefore easy to keep the service security level up-to-date; when a specific algorithm or key size becomes weak the sender can start using a stronger one. Secure RTP (SRTP) works similarly to IPsec but can be used only for protecting RTP streams.

MPEG-21 is remarkably more complex than MAC-addressing, DVB CA, IPsec and SRTP. The access conditions can therefore be much more precise, but the client application is accordingly more complex. Possibly, it could be used with some IPDC services, such as music and graphics download. Since MPEG-21 still is under development, it is, however, difficult to evaluate its suitability for IPDC in general. With OMA DRM it is possible to give a few different access conditions. Further, OMA DRM separate delivery specifies a rights objects delivery method, so OMA DRM could be used for ordering and delivering the rights objects, regardless of which protection method is chosen. With OMA DRM, the user further cannot forward the rights object to a friend – assuming that the user has an OMA DRM compliant terminal. Since the rights objects are identical for all users in all these schemes except DVB CA, there must be some mechanism that prevents users from distributing the rights objects, such as the OMA DRM separate delivery. There will naturally always be some users with terminals that allow forwarding but if the protection keys are changed often, users with honest terminals that want to use a specific service will not have time to wait for an illegally distributed rights object.

3.4.3 Payment

What users actually pay for is rights objects. A rights object is a data package that contains the information needed to decrypt a specific service. Typically, the customer's terminal orders them from an e-commerce platform. But sometimes – depending on the content provider's and service operator's preferences – the user downloads a rights object with a browser from a WWW portal and forwards it to a friend by SMS or e-mail.

The payment system must be able to authenticate and bill the user. A suggested solution is that each interaction channel operator runs an E-Commerce platform of its own; users then order rights objects from their own interaction channel operators. If the terminal is a mobile phone, the SIM card can contain the information needed to set up a connection to the E-Commerce platform. The format of the order and the delivery mechanism may differ between E-Commerce platform vendors. For instance, some platforms could support different interaction channels; the terminal could then list its interaction channels when it orders a rights object, letting the platform choose the most suitable one. Additionally, any other actor in the value chain can start vending rights objects, for instance through a tailored web portal. Building a customer base and building a billing system and a customer relationship management system may then be more challenging than the building the payment system itself. In the beginning, it is likely that the cell phone operators will manage the billing since they already have large customer relationship management systems and can authenticate their customers' terminals.

The user may be able to buy rights objects to some services himself, but in the general case, ordering rights objects should be transparent to the customer. When the user chooses a service with the ESG application, the application should automatically order the rights object. Built-in ESG applications may yet support only the simplest ordering mechanisms, for instance WAP Push over SMS. If a user wants to take advantage of more complex features that his interaction channel operator's payment system provides he may have to install a new ESG application. The application could naturally be distributed as an IPDC service.

In case the Open Mobile Alliance Digital Rights Management [58] separate delivery method (see Section 3.4.2) is used the user and the payment system exchange two messages. Figure 3.11 illustrates the interaction. First, the terminal sends an HTTP GET request to the payment system [59]. The payment system then checks that the terminal is allowed to purchase the content; for instance that the user is old enough and that the service he orders still is available. It then returns a rights object, for instance over SMS or GPRS using the Wireless Session Protocol (WSP) and the Wireless Datagram Protocol (WDP).

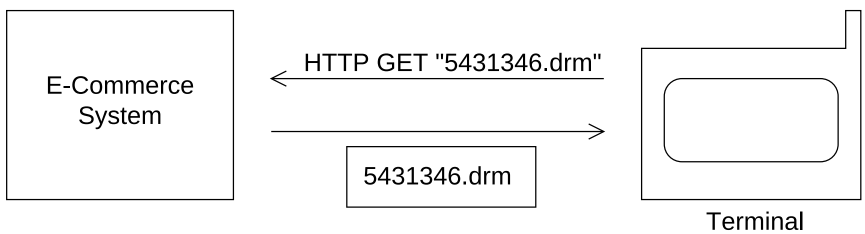

Figure 3.11: The user terminal and the payment system exchange two messages when the terminal purchases an OMA DRM rights objects.

The user terminal and the payment system exchange two messages when the terminal purchases an OMA DRM rights objects.

Actually, according to the OMA DRM specifications [58], the content should contain the URL of the rights object, and the terminal should use this address when obtaining the rights object. However, if users obtain their rights objects from their own interaction channel operator, there is no single URL, and the terminal must generate the address itself.

The Open Mobile Alliance further specifies a protocol for downloading chargeable content over the air. The purchasing process is similar to the OMA DRM separate delivery process described above, but the terminal must additionally confirm the transaction. The payment system charge for an event, that is writes a Charging Detail Record, only if it receives a positive confirmation. Lindström [45] evaluates this approach, concluding that it is efficient for selling low-value digital contents for mobile devices such as ring tones and games. Yet, Lindström notes that the protocol is vulnerable, since a dishonest terminal can access any content for free simply by not confirming the transaction. The content-over-the-air download model can also be used for selling IPDC services such as cartoons or news flashes, but monthly subscriptions need a more reliable payment system.

3.5 Presentation Layer

The IPDC services consist of audio and video streams, files and applications. In principle, there is no restriction on the content format. However, using a vast variety of formats is disadvantageous since downloading players and readers to a handheld device is at the least inconvenient.

Industrial recommendations naturally guide the manufacturers' choice of built-in decoders. Currently, two major players; the DVB Project representing broadcasting organizations, and the 3G Partnership Project (3GPP) consisting of telecommunication companies, are preparing to choose stream coding-decoding algorithms (codec) for their data transmission standards. Within 3GPP, the topical working groups are the Multimedia Broadcast/ Multimedia Services (MBMS) and the end-to-end packet switched streaming service (PSS). Within DVB, codec evaluation and election is conducted in the AVC (Audio/Video Coding) working group. Both organizations consider similar codecs. Since many IPDC terminals may be integrated into cellular telephones, it would naturally be beneficial if the consortiums agreed on common codecs.

Feasible stream codecs, judging from McCann's review of the new DVB codecs [54], are the MPEG-4 Advanced Audio Coding (AAC) for encoding audio and the H.264/AVC (Advanced Video Coding) algorithm for encoding video. The latter is jointly developed by the Moving Picture Experts Group of ISO/IEC and ITU-T. It will be published as H.264 within ITU-T and as Part 10 of the MPEG-4 specification (14496-10) within ISO/IEC. The DVB Project has, as a matter of fact, developed a specification for sending IP traffic using H.264 [52], but the H.264/AVC-algorithm has not yet been published since some patent holders cannot agree on the licensing terms [56].

There are no similar recommendations regarding file formats; at least HTML, XML and PDF are likely to be used. Applications are written in whatever programming language most user terminals understand. Applications for digital TV receivers often use the Multimedia Home Platform [20], which is a limited Java environment developed in the DVB Project. Among IPDC receivers, the closest counterpart is the Java MicroEdition Mobile Information Device Profile (MIDP) [74]. At least Symbian OS v7.0 [75] supports Java MIDP. Other programming languages, e.g. C++, and platform specific languages will be used as well.

3.6 Application Layer

The two most important applications in the IPDC terminal are the electronic service guide application and a multimedia player. Discussing the application-level protocols is outside the scope of this thesis, since they are not specific to IPDC. Instead, this section shortly discusses the ESG application, which is the user's interface to the IPDC services.

The Electronic Service Guide (ESG) application lists all services that are available in the broadcast stream. It uses broadcast service descriptions – Section 3.4.1 describes the Internet Media Guide framework that can be used to describe IPDC services. Most importantly, the ESG application shows how much each service costs, when they start and end, and possibly describes the services shortly. When a user chooses a service, the ESG application checks if the user has the right to access the service – that is, that the service either is freely available or that the user has purchased the rights to it. If not, the application warns the user of the service charge and asks him to confirm the transaction. If the user accepts it, the terminal automatically orders the corresponding rights object (see Section 3.4.3) and then launches a player or reader that shows the service. Some ESG applications may allow for personalizing service descriptions; this way content providers and the network operators can brand their services.

3.7 Accessing the Interaction Channel

The digital television broadcast network is unidirectional; hence, interactive services must use a separate telecommunication channel. Further, the terminal uses an interaction channel for ordering rights objects. As discussed in Section 3.4.3, rights objects can be ordered in several ways, but typically, the terminal will use SMS for ordering a rights object from the cell phone operator's e-commerce system. This section discusses ways for accessing the interaction channel from within an interactive IPDC service.

It is possible to distinguish between two types of interactivity; pseudo-interactive and interactive ones. In pseudo-interactive services, a service provider produces a bundle of services, some of which are interactive (e.g. a chat) and others are not (e.g. a video stream delivered over IPDC). The services are marketed as a bundle and when sending the IPDC service, the locations of the interactive services are announced. The Internet Media Guide framework, which can and probably will be used for making ESGs (see Section 3.4.1), includes some address elements. It is yet too early to say whether these suffice or if the guide format would have to be extended. The service usage experience depends partly on how the user accesses these interactive services. The ESG application could include a WWW browser and automatically take the user to the announced services, in which case the user more easily experiences the service bundle as a single usage experience. However, it can be difficult for the service provider to influence the look and feel of the usage experience since the provider cannot influence how the ESG integrates the separate service contexts.

Interactive services manage the interaction channel themselves. Applications can naturally use platform-dependent functionality and use any transmission protocol the platform supports. Additionally, the DVB Project has defined a stack of protocols that on the one hand hides the actual interaction channel from the application [17] and on the other hand defines how these transparent services should be implemented with different interaction channels. The protocol stack uses the Digital Storage Medium Command and Control (DSM-CC) specifications that are part of the MPEG-2 standard. There are four different services: general data transfer over IP and UDP or TCP, application-level object retrieval based on the object carousel (see Section 3.2.3), session control, and network management [17]. The specifications determine the protocol stacks used for the forward as well as the interaction channel.

The DVB Project interaction channel protocols are complemented with network dependent specifications. Of the mobile cell network technologies, GSM is currently the only one for which such a network dependent standard exists. Further, this GSM interaction channel standard [19] is likely of little practical use since it merely refers to existing GSM standards.

The DVB interaction protocols are developed for terminals with interaction links without transport-level data transmission support, such as modem cables or satellite uplinks. Since most IPDC terminals will have access to some TCP/IP-based link, e.g. GPRS, the DVB interaction protocols are unnecessarily complicated for the IPDC environment.

Chapter 4

Constraints And Requirements

The IP Datacasting concept evolved from an existing technology; the digital television networks. As well new technology as a new business concept had to be created, while taking the constraints that the digital television network sets into account. This section reviews the constraints and requirements coupled to the IPDC technology.

4.1 Constraints

The digital television network is a broadcast network, and therefore all terminals that are within the range of the transmitter receive the same signal. The first constraint is therefore the following:

> *C1. The same content is broadcast to all users within the range of the transmitter.*

Further, there is always only one sender at any specific instance on a physical digital television channel. If there were several, the signals would interfere with each other. There is therefore a dedicated channel operator who combines the traffic from all content producers, manages the Electronic Service Guide and transmits the content. The end-users, on their part, use a separate point-to-point channel, such as a GPRS connection or their home ADSL connection, for interactive services. This introduces the second constraint:

> *C2. The interaction channel is a point-to-point channel between the user and the service provider.*

Since one single operator controls the distribution channel, the content producers and service operators are forced to collaborate. In Finland, the Communications Market Act [81] requires the network operator to provide all service operators fair access to the network. However, this also means that service providers and content producers must collaborate at least with the network operator. Additionally, the interaction channel will often be a cellular network connection (GSM, SMS, GPRS or UMTS). Since the cellular network operators have the best opportunities to build IPDC charging systems the IPDC actors will also collaborate with them. In all, it is a value chain that together produces the IPDC services rather than the individual actors. The third and last constraint is:

> *C3. The IPDC services are co-produced in a value chain.*

Although quite simple, these three criteria set the framework within which the IPDC technology is developed. The next section reviews the requirements for the technology.

4.2 Requirements

The average IPDC user is a person on the move. He may be walking around or sitting in a car or on a train. The IPDC device is in general quite small; the IPDC functionality is integrated in mobile phones, personal digital assistants or laptops. This leads to several requirements:

> *R1. The technology must support a fast moving receiver.*

> *R2. It must be possible to use the services with light, handheld receivers using current-technology batteries without charging the battery more than once a day.*

> *R3. The receiver must be able to move between cells in the IPDC network and in the interaction channel network without interruptions in the service.*

Even if the requirements just listed are met, IPDC will not be used unless the users feel they need the IPDC services. The Wireless Application Protocol (WAP) is a warning example. In the mid-1990s, the Wireless Application

Protocol was supposed to create an information service similar to the World Wide Web for mobile devices. However, few organizations started creating WAP-content, such as a WAP-version of their corporate WWW site, and there is still no clear need for WAP services. On the other hand, services based on the Short Message Service (SMS) are popular although the user in practice usually purchases a short text string. Then, at a minimum:

> *R4. The technology should support a wide supply of services.*

This further leads to the following requirement:

> *R5. The technology should support a wide variety of protocols and content formats.*

To ensure the availability and quality of IPDC services, the time the content producers spend on learning and using the technology should be minimized to allow them to focus on developing the service content. Therefore:

> *R6. The IPDC technology should from the content producer's point of view differ as little as possible from other data communication channels.*

From the content producer's point of view, it is further important that only paying customers receive the service. This leads to the following requirement:

> *R7. The services must be protected so that only authorized users can access a specific service.*

However, a news provider may want to offer a news service across several countries, or two service operators may have an agreement that allow their customers to use either service, depending on which network they currently are in, with a single subscription. Therefore:

> *R8. The technology must support roaming.*

The user should not have to be aware of the technology behind the services; he simply uses an IPDC application on his terminal. At a minimum, therefore:

> *R9. The IPDC application should automatically retrieve descriptions of all available services.*

R10. After the user has picked a specific service and approved of the service price, the IPDC application should automatically acquire the corresponding access rights.

R11. The IPDC application should automatically set up interaction channels.

The next chapter evaluates the IPDC technology against these requirements.

Chapter 5

Evaluation

The objective of this section is to review the design of the IPDC architecture and to analyze how it meets its requirements. In this evaluation, one or a few design choices are identified as the main solutions to a specific requirement. This simplifies reality; in practice, the design choices are not independent solutions to independent problems but all requirements have to be considered in all solutions. However, it helps the reader compare IPDC to other, similar technologies.

As Figure 5.1 illustrates, the requirements can be categorized according to which layer the requirement mainly affects. Most requirements affect only one layer, although some stretch several layers. The most important design decisions have to be made on the link, transport and session layers, but some requirements are also reflected on the presentation and application layers. This evaluation discusses each layer separately, highlighting how the layer meets the requirements that are relevant for it.

5.1 Link Layer

The link layer is mostly affected by the physical reception conditions. However, also the service type requirements are reflected on the link layer; the link should not restrict the amount of services or the content types.

R1. The technology must support a fast moving receiver.

When a working group within the DVB Project first started investigating how digital television standards – most importantly DVB-T – could be used for delivering data, one of the problems is the fast moving receiver. A mobile

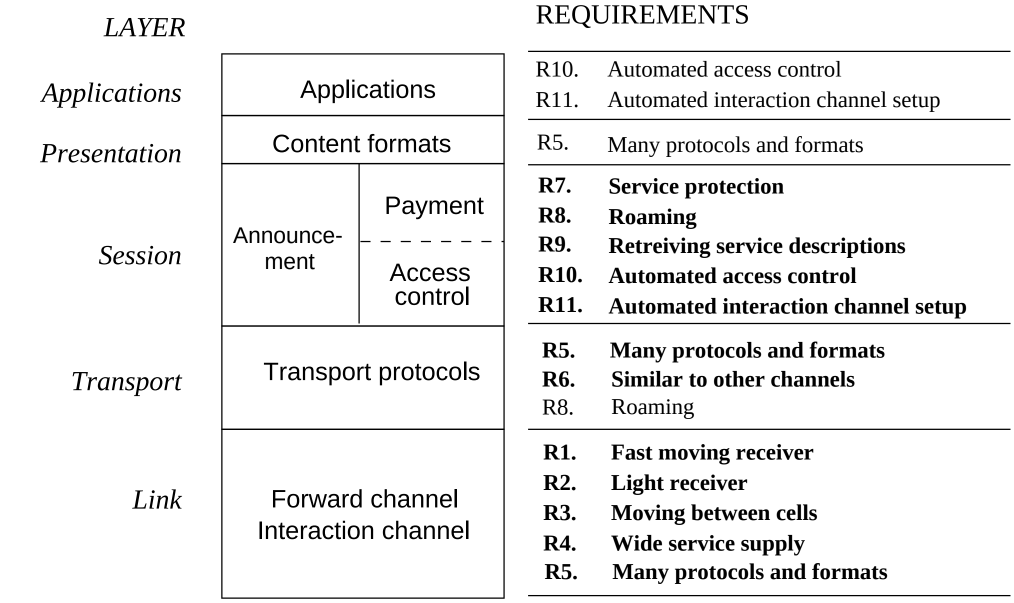

Figure 5.1: The requirements mapped to the IPDC layers. Requirements that are especially relevant for a specific layer are bolded.

user can receive DVB-T transmissions especially if the receiver combines the signals from two antennae; studies in Australia [25] show that a user moving up to 120 km/h can receive a DVB-T 8k hierarchical transmission using 64-QAM modulation and a 3/4 code rate within 40 km of the transmitter.

However, the DVB-T standard had other problems; most importantly, it was heavy for receivers to process. The working group therefore started developing a new transmission mode for mobile receivers; DVB-H (see Section 3.2.2). DVB-H introduces a third COFDM modulation mode; the 4k mode, which lies between the existing 2k and 8k modes. The COFDM modulation mode determines the number of subcarriers, and as Section 3.2.2 describes, the spacing between the subcarriers influences how robust the signal is to the Doppler Effect. Even though the 2k modulation mode provides the largest subcarrier spacing, it requires the smallest cell size. With the new 4k-mode, the transmission network can better support fast moving receivers with a cell size twice as large as in the 2k-mode.

DVB-H also adds some additional forward error correction, MPE-FEC. The error rate usually grows as the receiver moves faster, and MPE-FEC helps the receiver deal with a higher error rate.

> *In the IPDC architecture, support for a fast moving receiver (R1) primarily stems from the DVB-H 4k mode and the MPE Forward Error Correction (MPE-FEC).*

R2. It must be possible to use the services with light, handheld receivers using current-technology batteries without charging the battery more than once a day.

The largest problem with using DVB-T for mobile reception is the power consumption [66]. In DVB-T the multiplexing slots are so short that a receiver must process the entire stream to find the transport packets it is interested in. In practice it is impossible for hand-held terminals to receive DVB-T transmissions since the heavy processing uses up their batteries almost immediately.

The DVB-H time slicing feature (see Section 3.2.2) reserves slots of several hundred milliseconds for each virtual channel. The receiver can therefore turn itself off while waiting for the next slot in the virtual channel. According to an IPDC terminal expert, in the first IPDC terminals the battery will last for 2-3 hours of continuous usage, such as receiving a video stream over DVB-H and watching it [40].

DVB-H can further be received with a single antenna. Although DVB-T can be used for mobile reception [25], the best results are received with diversity antennae, i.e. two antennae whose signals are combined. The single antenna, however, simplifies the terminal design.

> *The IPDC terminal can be small enough to carry around (R2) since time slicing enables using current-technology batteries as power supplies and a single antenna can be used to read in the radio signal.*

R3. The receiver must be able to move between cells in the IPDC network and in the interaction channel network without interruptions in the service.

In some situations the network may consist of a single cell, which for instance covers a city center. In the 4k OFDM mode the cell radius is about 30 km. However, if the same services are to be available in a larger area – e.g. along a freeway – the user must be able to move between cells in the forward and interaction channel networks without interrupting the service.

The DVB-T/H network can be built as a multifrequency or a single frequency network. In single frequency networks, adjacent cells use the same frequency and send the same content. The radio modulation technique (see Section 3.2.2) that DVB-T/H uses allows the receiver to combine the signals from the different cells at the border between the cells. The receiver therefore all the time has a signal at the same frequency within the whole network and as long as the combined signal is strong enough at the cell borders, there is no interruption when moving between cells.

In multifrequency networks, different cells may carry different content. For a user to be able to move between two cells without interrupting the service he is using, both cells must naturally provide that service. If the service is time sliced (see Section 3.2.2), the receiver is turned on only when reading in a data burst. The terminal can then switch cells while the receiver is turned off [33]; this way the cell handover does not interrupt the service.

If the service is interactive, the terminal might have a connection to the service over the interaction channel network. The interactivity connection should naturally neither be interrupted when moving between cells in the interaction channel network. The user then uses a mobile telecommunication network, such as GSM, GPRS and UMTS, for the interaction channel which all make cell handover so that the interaction channel is not interrupted when switching cells.

> *It is possible to move between DVB-H cells without interrupting the IPDC service (R3). The interaction channel network takes care of cell handover for the interaction channel.*

R4. The technology should support a wide supply of services.

Despite the large link-level overhead – in a typical transmission configuration MPE-FEC causes 25% overhead and the inner convolution coding an additional 33% – the capacity of a DVB-H channel is about 11 Mb/s [43] of transport level data. If the content producer uses an efficient content compression algorithm, such as MPEG-4, a video stream for a small screen uses about 250 kb/s. The DVB-H channel can then carry 30-40 simultaneous services. It is too early to say whether there is a market for this many services, but it is at least possible to offer a large amount of differentiated services simultaneously.

> *The capacity of a DVB-H channel, about 11 Mb/s, is enough for 30-40 simultaneous services (R4).*

R5. The technology should support a wide variety of protocols and content formats.

Handheld terminals develop remarkably faster than stationary TV sets, and users also renew their terminals quite often. Especially content compression algorithms develop fast. Contrary to the digital television standards, IPDC does therefore not make any assumptions about the content type. All content is treated as data and the IPDC link layer therefore uses some encapsulation technique (see Section 3.2.3). MultiProtocol Encapsulation (MPE) will most likely be used in the beginning since there currently are no practical alternatives.

It is in principle possible to use different encapsulation techniques in the same network. There would then be several encapsulators that each produces an MPEG transport stream; these are then multiplexed together before the result is modulated and transmitted. Yet, at least in the author's opinion, multiple encapsulators will be an exception rather than a rule.

> *The link layer does not restrict the choice of protocols (R5) since all content is treated as data and encapsulated using some encapsulation technique (initially MPE).*

5.2 Transport Layer

One of the most important targets when designing layer models is layer transparency. When layers are fully transparent, protocols at different layers are independent of each other and a layer implementation can be exchanged without the other layers noticing. The digital television broadcasting organizations have taken quite the opposite standpoint and accordingly, digital television transmission standards cover functionality from most layers, from signal modulation to content compression.

IPDC on its part follows the layered approach and uses the digital television broadcasting specifications only on the link layer. In principle, any protocol could be used on the transport layer; Yet, the transport layer requirements illustrate why the IPDC transport layer uses the UDP/IP suite.

R5. The technology should support a wide variety of protocols and content formats.

In principle, any protocol can be used on the transport layer. However, as described in the previous section, IP and UDP will often be used for practicality. Additionally, the protocols should either not require any response from the receiver, or be able to deal with responses from multiple receivers.

> *Any transport protocol that supports uni-directional transmission or multi-terminal reception can be used with IPDC (R5); for practical reasons, the protocols should additionally run on top of IP or UDP.*

R6. The IPDC technology should for the content producer's point of view differ as little as possible from other data communication channels.

In principle, it is possible to use any kind of protocol on the transport layer. However, as Section 3.3 explains, IP has advantages; service and application developers are used to using the UDP/IP suite and most terminals include UDP/IP implementations.

The most important features IP provide are addressing and a payload checksum. In principle, it is possible to use physical addresses and link-layer checksums instead of IP. However, if the application wants to use UDP and other IP-based protocols, it would have to use a modified UDP/IP implementation or implement the transport-level functionality itself. Using IP as the basic transport protocol creates a little overhead but makes it easier to develop applications since the IPDC network can be treated as one access network, although unidirectional, among others.

> *Since IPDC uses UDP/IP as its basic transport protocol, applications can treat the IPDC network as any other access network (R6).*

R8. The technology should support roaming.

When a user moves from one network to another, it takes some time for the terminal to prepare itself for receiving services in the new network, even if the services in both networks are identical and use the same transport level parameters. The link-layer parameters, e.g. the radio frequency, and the session-layer parameters, e.g. the protection, are different and the service is therefore interrupted when the user moves from one network to another.

Roaming primarily affects the session layer, since the user's access rights must be transferred from the home network to the remote network. However, the terminal must also be able to identify which services in the home and the remote network correspond to each other. The author has not found any documentation discussing roaming in IPDC. A simple mechanism to identify corresponding services is yet that the services use identical transport-level parameters, such as the IP destination address and the UDP port number.

> *The session layer is primarily responsible for supporting roaming (R8). However, one way to map a service in a home network to a service in a different networks is to use identical transport-level parameters.*

5.3 Session Layer

R7. The services must be protected so that only authorized users can access a specific service.

Section 3.4.2 presents several different alternatives for protecting the services. The Electronic Service Guide, which is discussed in Section 3.4.1, should contain information about where the IPDC application finds the rights object. The three different OMA DRM modes, IPsec, SRTP and possible MPEG-21 could all be used in different situations. However, the IPDC application should know how to order the rights object and how to treat the rights object so that it can access the protected service. Nokia offers an IPDC service system in which the services are protected with OMA DRM separate delivery. The specifications of the structure of the Electronic Service Guide and the ordering messages are openly available and IPDC applications that are used in a network using the Nokia solution must follow these specifications. Later, it is possible that an industry consortium, such as the Open Mobile Alliance, specifies how the rights object is announced in the Electronic Service Guide and how the terminal orders and treats different types of rights objects.

> *The IPDC content is encrypted using some access control scheme, and only users that purchase the decryption parameters can access a specific service (R7).*

R8. The technology must support roaming.

In this thesis, the session layer has three tasks: To announce services, to control which user can access which service, and to enable paying for services. Roaming between networks affect all three tasks.

The user's terminal should be able to understand the service announcements in the remote network. The IP/MAC Notification Table (INT) and the electronic service guide together describe the services. The INT is used in the same way in all networks, but it is too early to evaluate how strictly the electronic service guide specifications define the electronic service guide structure. It is possible that an ESG application is tailored for a specific network, in which case the user must have the ESG applications for all networks he visits.

The terminal should further deal with the access control mechanism in the remote network. It is possible that the terminal inherently supports several access control techniques, otherwise the remote network should use the same access control mechanism as the user's home network. Further, if a user subscribes to a service that is available in both networks, the access rights should transfer from the home network to the remote network. One way to transfer the rights is to use identical rights objects in both networks; a user can then access both services with the same rights object.

Paying for a service in a remote network to a foreign operator requires that the user's home operator forms roaming agreements with the foreign operator. How the payment is implemented is outside the scope of this thesis.

> *Roaming (R8) affects all three tasks in the session layer: service announcement, access control and payment. It is too early to evaluate whether all IPDC networks will use similar technology at the session layer – roaming may depend on operators agreeing on using similar technologies and parameters.*

R9. The IPDC application should automatically retrieve descriptions of all available services.

The Electronic Service Guide (ESG), described in Section 3.4.1, is the most important tool for announcing and describing services. The ESG contains human-readable descriptions and transport-layer parameters and is broadcast periodically. The IP/MAC Notification Table (INT), which was discussed in Section 3.2.3, can further map transport-layer parameters, such as IP addresses, to link layer parameters.

> *The Electronic Service Guide in combination with the INT provides the terminal with a full service description (R9), including parameters for locating it in the MPEG transport stream.*

R10. After the user has picked a specific service and approved of the service price, the IPDC application should automatically acquire the corresponding access rights.

Since the format of the Electronic Service Guide is open, it is too early to evaluate how well it describes different types of access control. The ESG is discussed in 3.4.1. It is possible that the Open Mobile Alliance specifies in detail how IPDC rights objects are announced, ordered and treated. However, the OMA DRM separate delivery mode specifies how rights objects are delivered and requires that terminals at least can use WAP Push OTA over, for instance, SMS or GPRS, for receiving the rights objects.

> *It is too early to evaluate how the final IPDC architecture will support automatic access rights purchasing (R10). It is, however, one of the most central requirements and OMA DRM separate rights delivery gives some guidelines on how the rights objects are ordered and delivered.*

R11. The IPDC application should automatically set up interaction channels.

The user uses the interaction channel for purchasing rights objects and for using interactive services. The interaction channel is discussed in Section 3.7. The previous section discussed how the terminal sets up a connection to the rights-vending server when acquiring access rights. Additionally, the user should not have to manually configure his terminal when using an interactive service.

The interactive service should automatically set up the interaction channel, relying on platform specific functions. Interactive digital television set-top use MHP, a limited Java runtime environment that is specially developed for set-top boxes. Through the MHP, the application can use the interaction channel through the DVB interaction channel specifications. There are different specifications for different transmission media. Of the cellular technologies, the DVB interaction channel specifications currently only support GSM [19] and the GSM specification is quite vague. Later, the specifications may be extended to cover also other technologies such as GPRS and UMTS. Before then, the DVB interaction channel specifications will likely be of little use to IPDC services. Interactive IPDC services can naturally use the terminal environment in the same way as any normal application. Hopefully, the IPDC terminals will provide some homogenous runtime environment; for instance, the Java Micro Edition is growing more common among handheld devices.

> *Interactive IPDC services use platform-specific functions to set up interaction channels (R11). If the DVB interaction channel specifications are extended to cover more cellular network technologies*

and the IPDC terminals start supporting them, interactive IPDC services can also use them.

There is also no common agreement on how rights objects are ordered.

5.4 Presentation Layer

R5. The technology should support a wide variety of protocols and content formats.

Section 5.1 discusses the delivery protocols; the main restriction is that they should be unidirectional and it should be possible to use them on top of IP. Contrary to digital television standards such as the DVB transmission standards, IPDC does not restrict what content formats are used. The content format choice depends on what formats the users' terminals support. Many users will likely use only the applications that are pre-installed on their terminals, but it is possible to download new players and decoders to the terminal. A service provider could distribute a coder-decoder as an IPDC service to make sure that all terminals can show his content correctly.

IPDC does not restrict what content formats are used (R5).

5.5 Application Layer

The application layer is not standardized and it is impossible to control how the application layer behaves. However, the user should not need to be aware of the technology beneath an IPDC service when using it, and at the minimum, the application layer should meet the following two requirements:

R10. After the user has picked a specific service and approved of the service price, the IPDC application should automatically acquire the corresponding access rights.
R11. The IPDC application should automatically set up interaction channels.

The session layer will, however, provide the application layer with tools for automating the ordering of access rights and setting up the interaction channel.

The application layer is not standardized and it is therefore impossible to evaluate, how it will meet its requirements. Application developers should concider how access control can be automated (R10) and how the interaction channel is set up (R11).

Chapter 6

Example Architecture

Given the variety of actors involved in producing every specific IPDC service, successful large-scale IPDC service provisioning requires a specified, open architecture that ensures compatibility between the different stages in the service provisioning process. There is still no such common architecture. Aaltonen [2], however, presents a general IPDC architecture that describes what commercial architectures will look like. In Finland and in Europe DVB-H will be used as the forward channel, and Henriksson et al. [33] describe the DVB-H network elements.

6.1 General IPDC Architecture

Aaltonen's [2] solution addresses the service delivery and provision process starting from the management of service delivery to the provision of rights objects. It focuses on the tasks of the service operator, the network operator and the user terminal. On the other hand, the interactions between content producers, providers and aggregators are left outside the scope of the general architecture since these interactions vary from case to case. Aaltonen's solution is based previous work by Grundström et al. [29] and later by Luoma [51] on using the DVB network as a general IP data transmission network, by Väre [82] on IP service announcement and by Qvist [64] on controlling DVB-T transmitted data in a handheld device.

Figure 6.1 depicts the example architecture. The services enter the architecture at the *service delivery platform*. The platform receives the content – streams or files – in advance, stores it, converts it to formats generally used among the user terminals, protects it if it is not to be freely available, and

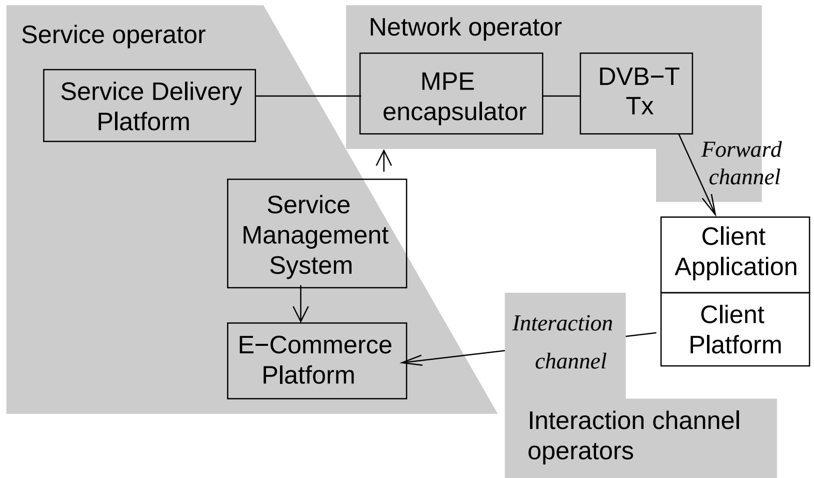

Figure 6.1: An example IPDC architecture (adapted from [2]).

relays it according to schedule. Aaltonen's solution uses MultiProtocol Encapsulation (see Section 3.2.3), and the content is then delivered over some network – in Aaltonen's solution the Internet – to the *multiprotocol encapsulator (MPE encapsulator)*. The MPE sections are relayed to the *DVB-T modulator (DVB-T Tx)* and broadcast. Next, the *user terminal* – consisting of the *client platform* and the *client application* – receives the stream. The client platform demodulates the signal, removes the MPE-encapsulation, and delivers the data to the client application. Depending on the protection scheme chosen, either the platform or the application or both must apply the rights object to access the data.

To access protected content, the user terminal must possess the corresponding rights object. The terminal can acquire the rights object by ordering it from the *E-Commerce platform (EC)*. The *service management system* is used to configure the encapsulator and the E-Commerce platform. In this example architecture, it manages the protection parameters and the service announcement. It therefore configures the encapsulator to protect each service according to the protection settings of that service, using the correct protection parameters, and delivers the corresponding rights object to the E-Commerce platform. Further, it generates the Electronic Service Guide (ESG, not shown in Figure 6.1) which informs the user of the available services and their costs and the client platform of how the service is accessed and how the rights objects ordered.

The interaction channel service operators and interaction channel network operators are not addressed in the architecture, as the IPDC-related interaction channel traffic from their point of view does not differ from other traffic on the interaction channel network. However, the interaction channel operators may take over parts of the service operator's tasks, for instance run an E-Commerce platform.

6.2 DVB-H Network Architecture

Aaltonen's [2] architecture, which the previous section discusses, does not address the broadcasting network. In Finland, DVB-H will be used as the underlying broadcasting network. Henriksson et al. [33] draw up the the DVB-H network architecture similarly to Figure 6.2.

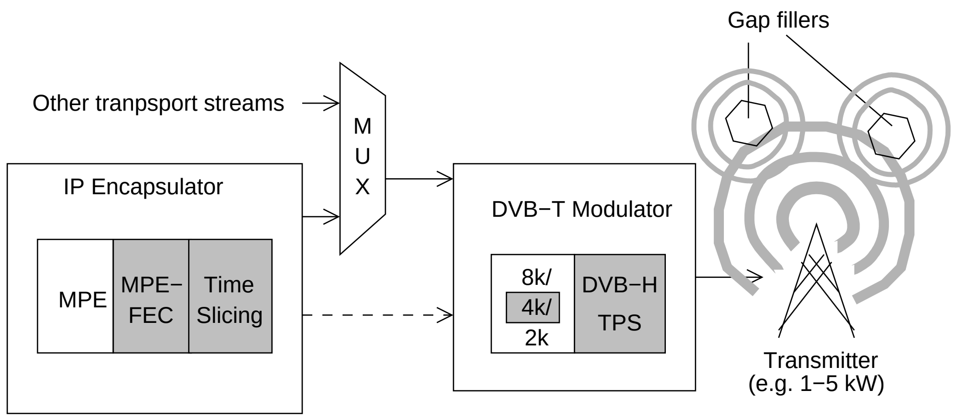

Figure 6.2: The DVB-H network elements (adapted from [33]). Elements specific to DVB-H are shaded.

The first element is the *IP Encapsulator* that converts the IP traffic to a transport stream. The IP Encapsulator is responsible for two of the DVB-H features; adding extra forward error correction at the MPE level (MPE-FEC) and time slicing the traffic. Some technology vendors may offer additional functionality, for instance allowing the network operator to plug in some DVB Conditional Access system. All encapsulators will most likely provide different addressing mechanisms, for instance offer using IP-to-MAC conversion when assigning MAC-addresses to the MPE sections. Further, it may be possible to configure different capacities and priorities for different content providers.

Several content aggregators can stream content to the IP Encapsulator using a regular IP connection. Management interfaces (e.g. for administrating addressing) are not standardized.

It is possible to mix DVB-H and DVB-T traffic – in that case a *multiplexer* or *MUX* is needed. Since the multiplexer operates on the MPEG Transport Stream level [15], the multiplexer does not distinguish between DVB-H and DVB-T traffic.

The stream is now transformed to an MPEG transport stream and is next delivered to the *modulator*. If DVB-H is sent as part of the DVB-T stream, the multiplexed stream must be modulated according to the DVB-T standard. In that case, the 8k or the 2k mode is used. In a designated network also the 4k mode can, and most likely will, be used.

The modulated stream is next transmitted. In Finland, a Single Frequency Network will be used. As in DVB-T, terminals that receive signals from several transmitters can take advantage of all the signals. The size of the cell is proportional to the length of the guard interval, which on its part depends on modulation mode, which can be 2k, 4k or 8k. Hence, the lower the mode used, the smaller is the cell size. A large number of gap fillers are required to eliminate signal gaps that would annoyingly interrupt the IPDC service when the mobile user enters the gap area. Especially ensuring indoors coverage is critical. Kohtala et al. [43] estimate that the Finnish network will need 200-300 main transmitters of about 1-5 kW, and up to ten times more gap fillers.

Chapter 7

Future of IP Datacasting

Currently, IP Datacasting is developing at three levels; the IPDC technology, the IPDC content and the IPDC business models. The first research results are promising.

7.1 Current Situation

In 2001-2003 several tests have been performed to lay out the basis for IPDC. In Finland, the VTT Technical Research Centre of Finland conducted user tests using 3G and WLAN networks [44]. The results indicated that to the users, the mobile services are similar to television services. They used their mobile TV terminals to watch their favourite programs and especially for watching programs they had missed previously. Some differences were however found; for instance, the TV programs should be shorter when watching them on the mobile terminal. The services in the test were free but the users appreciated they would pay about 50 cents for a TV program or 15-20€ for a monthly subscription.

Also the IPDC technology has been analysed and developed through prototypes. An IPDC test network, using DVB-T, has been operating in Finland since September 2002 [68]. Further, in Berlin, an advertising company already uses Digital Audio Broadcasting technology, which is closely related to Digital Video Broadcasting, to deliver commercials and information flashes to television screens in the Berlin metro.

The European Commission created a directive called the Common Regulative Framework for Electronic Communication Networks and Services in 2000. This directive renews the European broadcasting policies and strives to separate the delivery networks from the services they carry. This would mean that any type of content may be delivered using TV or radio technology [7]. The EU member countries were to modify their national legislation to enforce these directives by July 2003. Accordingly, Finland passed a new Communications Market Act that came into force July 25, 2003 [81]. Operating an IPDC network will require a license [7].

7.2 Next Steps

In the next few years, the technical development continues while the IPDC content is investigated and the business concepts are sketched.

In 2004, the technical validation of DVB-H starts. The Finnish test network will start using DVB-H features in spring 2004 [76]. Initially, it will use QPSK-modulation which provides a capacity of about 5-6 Mb/s. Later, it may start using 16QAM-modulation, which will increase the capacity of 11 Mb/s. In autumn 2004, user tests with 500 users in the Helsinki region will start to develop new IPDC services [65]. In Berlin, another technical pilot in the regime of the Broadcast Mobile Convergence project will start in 2004 [35], also.

Once the technical pilots are finished, the next step is to start IPDC services on the existing networks or build dedicated IPDC networks. In Finland, a working group [43] recommended in 2003 for the Ministry of Transport and Communications that the IPDC services use a new, dedicated radio broadcast network. The IPDC network would use DVB-H and provide one physical radio channel of 11 Mb/s. The analogue radio and television broadcast transmissions in Finland are planned to stop 2007 [55], which leaves radio frequencies free. If the market for IPDC services evolves, it is possible to allocate frequencies for additional networks at that point.

7.3 Towards Convergence?

The services that are delivered over IPDC is similar to those delivered over 3G. However, IPDC is a mass media, while a 3G user can order a personalized service on-demand. Delivering services to several simultanous users will yet become cheaper in the 3GPP Release 6 than in the previous releases. This next release includes, as an optional feature, Multimedia Broadcast/Multicast Service (MBMS) [1] which makes it is possible to deliver services in the 3G network using physical broadcasting or multicasting. MBMS will support cell handover. Further, it will have built-in support for charging for services and for protecting the services so that only paying customers are able to access them. The MBMS services will use similar data rates as the IPDC services, since they, similar to IPDC services, will use efficient compression algorithms. It may be easier to develop interactive services for MBMS than for IPDC since the 3G network is bidirectional. MBMS also has better support for delivering services on demand, for instance after a certain number of users has ordered a service, already since the cell is smaller than an IPDC cell. However, when the number of simultanous users is large, it is cheaper to use IPDC for delivering the service.

IPDC and MBMS will serve different purposes, and some researchers envision complete convergence between the technologies. For instance, the Broadcast Mobile Convergence project [5] aims at developing platform that combines the DVB and the 3G network so that users do not need to be aware of which network is used to deliver a service – a service could even use both networks interchangably. Also the DVB Project envisions a convergence platform using both the DVB and the 3G network.

The DVB convergence platform is depicted in Figure 7.1. The DVB project intends to specify the interfaces, leaving the implementation of the modules to the manufacturers. IP Datacasting can be seen as a special case of this framework where a radio broadcast network is used as the forward channel and the telephone network as the interaction channel. Since there is no handover between the two networks, IP Datacasting uses only the I_MT-interface in Figure 7.1.

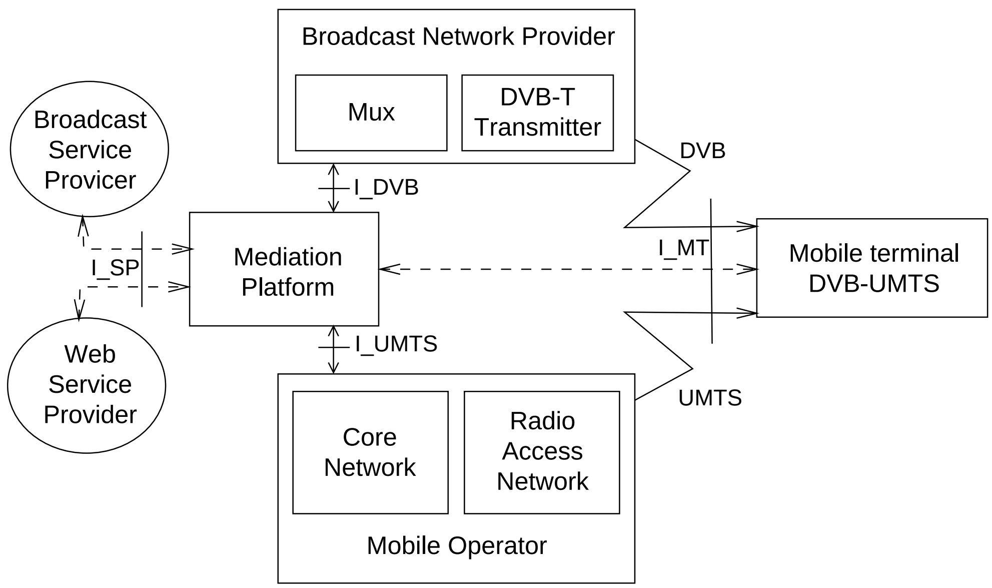

Figure 7.1: A framework for datacasting using broadcasting and radio networks (redrawn from [8]).

Chapter 8

Conclusions

IP Datacasting is developed for sending different types of content to mobile users with small handheld receivers. The strength of IPDC lies in using a digital broadcasting network for delivering the data. The digital broadcast network is a cheap delivery technology and IPDC therefore has the potential to form a new mass media. If IPDC is taken into use, DVB-H will in Finland be used as the digital broadcasting technology.

Most transport protocols and access control mechanisms are designed for bidirectional channels. IPDC uses a unidirectional broadcast network for delivering the data and a separate interaction channel for all bidirectional communication. This affects the entire protocol stack. The objective of this thesis was therefore to review the protocols and standards related to IPDC and evaluate how they meet the requirements of the IPDC architecture. The focus was on the standards that will be used in Finland.

This thesis used a five-layer model for describing and analysing the IPDC architecture.

The link layer is the most developed of the layers, unsurprisingly since the link layer is the cornerstone of the IPDC concept. In principle any digital broadcasting technology could be used, but technologically the most sophisticated is the DVB-H digital broadcasting specification which is especially designed for mobile users with handheld receivers. There are at least two candidates for the link level data encapsulation, MPE and ULE, but since ULE is still not fully specified or tested, MPE will be used initially. Mobile cellular technologies such as GSM and UMTS can be used for the interaction channel and IPDC does not introduce any changes to them.

The most important component at the transport layer is IP. IP and UDP can be used without modification. Yet, the unidirectional delivery channel affects the higher-level transport protocols. RTP, which already is used for streaming in traditional networks, can be used in a unidirectional mode in IPDC. Completely new protocols are, however, developed for file delivery. The file delivery protocol FLUTE will likely be finished within one year and the transport layer can be finally evaluated only after the protocols are finished. However, the transport layer in general seems to meet its requirements.

The session layer, on the other hand, contains several open questions. There are several qualified content protection techniques, and there are high-level suggestions for the structure and format of the electronic service guide. Unless these are specified more precisely, the user must somehow download an application to his terminal that enables him to use a specific IPDC network. It is also more difficult to roam between networks. It is, however, likely that the Open Mobile Alliance or some other industry consortium specifies the session layer.

The presentation and the application layers do not need as much support from standardization organizations. New terminals will contain support for efficient codecs and service and application developers will use codecs and formats that most terminals on the market support. Yet, the service and application developers carry the responsibility for making the IPDC concept attractive to the users. The other layers provide the tools, but in the end, the services and terminal's IPDC application determine whether the IPDC concept will win ground.

The IPDC services can further be roughly divided into interactive and non-interactive services. The technology evaluation reveals that the IPDC technology supports non-interactive services quite well – even if the session layer is not specified in detail, IPDC can be used if the IPDC application is tailored for a specific network. In Finland, there will at least initially be only one network and this is therefore possible.

It will not be difficult to add some interactivity to a simple streaming service; for instance, the service provider can provide a WWW chat or a voting page. Truly interactive services are similar to applications for the hand-held terminals. In digital television networks, set-top boxes that can use interactive services support a common platform, MHP. There is no corresponding platform designed especially for interactive IPDC terminals, although several terminals contain a Java runtime or a Java MicroEdition runtime environment. It is, however, difficult to evaluate how the interactive services will succeed.

Currently, there are pilots evaluating the performance of the transmission technology. The next step in the technology development is to develop the session layer. It is also important to gain a better understanding of the IPDC service and business concepts. Hopefully, it will not take too long before Benny Boy can watch the cartoon "Antti, the Courageous Ant" on his IPDC terminal in the lazy Saturday morning hours.

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