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Innovation and Creativity

The Future Mobile Entertainment Service

Multimedia Experience on 3G

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Problem Description

The thesis is to define a new entertainment service that offers top-noch multimedia experience on 3G mobile phones. The services's strenght lies in live broadcasting of events like concerts and other performances, as well as direct and on demand download of audio, video and other content.

The goal of the service is to provde an extraordinary multimedia experience never seen before, taking advantage of the allready exploding Mobile TV technology.

The service is to provide high QoS, secure the media owners, providers and the buyer’s rights, as well as prevent illegal copy and distribution of content.

There is to be established a business model for the service, which includes a payment solution proposition, and suggestion for content protection.

The system is also to be designed on an architectural level.

Assignment given: 03. January 2006
Supervisor: Leif Arne Rønningen, ITEM

Preface

This Master Thesis is a result of the entrepreneur in me. I am always keeping an eye out for new business ideas and opportunities to make money. I am creative by nature and have an urge follow my own lead. This has led me down the path of defining my own thesis problem. It was a great experience to explore Mobile TV technology and all the benefits that come along with it. It enabled me to combine two of my favourite hobbies; music and using my Mobile Phone. With this fact in mind, the Future Mobile Entertainment Service (FUMES) has been created to give artists the opportunity to reach out to the masses in a completely new way and let consumers be virtually present at events of choice, regardless of their ability to attend them in physical reality.

I would like to thank the following professors; Steinar Hidle Andresen for being the professor responsible for opening my eyes to Telecommunication, Education & Research Coordinator Kristen Rekdal for believing in me and giving me all the encouragement and useful advices throughout this project and my last years at NTNU. Additionally I would like to thank Professor Rolv Bræk for helping me with system deign questions and Professor Leif Arne Rønningen for being the coolest professor on Campus and an excellent mentor during the Thesis.

Adittionally I would like to thank Marius Meltvedt and Morten Rolfsnes at Burston; Marsteller for providing me with a Nokia N90 phone to use during the Thesis writing.

Special thanks to my loving and patient husband F. Robert Verkerk, for providing me with knowledge about the music industry, music rights and for having faith in me and always supporting and pushing me to the limit during this project.

In addition I would like to thank Gisle Østereng for giving me good advices throughout this project and having faith in my idea, my friends and my family for unlimited support throughout my “endless” studies and especially during this Thesis.

Trondheim 14.06.2006

Magdalena Katarzyna Lipska

Dedication

This Thesis is dedicated to my loving and caring parents Lidia Lipska and Leszek Lipski, I love you with all my heart!

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Abbreviations

2G	Second Generation Protocol
3G	Third Generation Protocol
3GPP	Third Generation Partnership Project
AAC	Advanced Audio Coding
AES	Advanced Encryption Standard
AH	Authentication Header
ALC	Asynchronous Layered Coding Protocol
AMC	Adaptive Modulation and Coding
AMR	Adaptive Multi-Rate
AOL	American Online
APN	Assessor's Parcel Number
ASCII	American Standard Code for Information Interchange
ATSC	Advanced Television Systems Committee
AVC	Advanced Video Coding
AVI	Audio Video Interleaved
BIEM	Bureau International de l'Edition Mecanique
BM-SC	Broadcast / Multicast Service Center
BSC	Base Station Controller (GSM)
CAMEL	Customized Applications for Mobile network Enhanced Logic
CB	Cell Broadcast
CBC	Cell Broadcast Center
CBS	Cell Broadcast Service
CC	Credit Card
CDMA	Code Division Multiple Access
CEK	Content Encryption Key
CIF	Common Intermediate Format
CNO	Cellular Network Operator
CP	Content Provider
CPIM	Common Profile for Instant Messaging
CPU	Central Processing Unit
CSE	Camel Server
DAB	Digital Audio Broadcasting
DCF	
DCT	Discrete Cosine Transform
DFC	Discrete Media Profile
DJ	Disc Jockey
DMB	Digital Multimedia Broadcast
DNO	Datacast Network Operator
DNS	Domain Name Service
DRM	Digital Rights Management – OMA standard for digital protection of content

DSP	Datacast Service Provider
DTD	Document Type Definition
DVB-C	Digital Video Broadcast, Cable
DVB-H	Digital Video Broadcast, Handheld
DVB-S	Digital Video Broadcast, Satellite
DVB-T	Digital Video Broadcast, Terrestrial
DVD	Digital Video Disc
eAAC	Enhanced Advanced Audio Coding
ECC	Elliptic Curve Cryptography
EDGE	Enhanced Data rates for Global Evolution
EFR	Enhanced Full-Rate
ESG	Electronic Services Guide
ESP	Encapsulated Security Payload
ETSI	European Telecommunications Institute
FEC	Forward Error Correction
FLUTE	File Delivery Over Unidirectional Transport
FM	Frequency Modulation radio transmission
FUMES	Future Mobile Entertainment Service
GGSN	Gateway GPRS Support Node
GERAN	GSM EDGE Radio Access Network
GPRS	General Packer Radio Service
GSM	Global System for Mobile Communications
HCSD	High-Speed Circuit Switched Data
HD-DVD	High Definition DVD
HDTV	High Definition Television
HE-AAC	High Efficiency Advanced Audio Coding
HLR	Home Location Register
HS.DSCH	High Speed Downlink Shared Channel
HTTP	Hypertext Transfer Protocol
HSDPA	High Speed Downlink Packet Access
ICQ	“I Seek You” – a proprietary chat system (see IRC)
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IPDC	IP Datacast
IPDS	IP Datacast System
IPsec	Internet Protocol Security
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
IRC	Internet Relay Chat
ISDB-T	Terrestrial Integrated Services Digital Broadcasting
ISO	International Organization for Standardization
ITU	International Telecommunications Union
ITU-T	ITU Telecommunication Standardization Sector

IUG	Interactive User Guide
JVT	Joint Video Team
LCD	Liquid Crystal Display
MBMS	Multimedia Broadcast/Multicast Service
MIMO	Multiple Input Output
MIRC	Multi-server Internet Relay Chat
MMS	Multimedia Message Service
MP3	Motion Pictures Experts Group layer 3 compression technology
MPEG-4	Motion Pictures Experts Group layer 4 compression technology
MS	Mobile Station (GSM)
MSN	Microsoft Network
MSRP	Message Session Relay Protocol
NOK	Norwegian Krone (currency)
NSD	Name Server Daemon
OFDM	Orthogonal Frequency Division Modulation Method
OFDMA	Orthogonal Frequency Division Multiple Access
OMA	Open Mobile Alliance
OS	Operating System
OSA	Open Service Access
PDA	Personal Digital Assistant
PDCF	Continuous Discrete Media Profile
PHY	Physical (layer of OSI Reference model)
PM	Private Message or Personal Message
PTP	Point-to-Point
PTM(P)	Point-to-Multipoint
PS	Packet Switched
PSS	Packet Switched Service
PSTN	Public Switched Telephone Network
QCELP	Qualcomm Code Excited Linear Predictive
QCIF	Quarter Common Intermediate Format
QoS	Quality of Service
QVGA	Quarter Video Graphics Array
RAN	Radio Access Network
RAT	Radio Access Technology
RI	Rights Issuer
RFC	Recursive Flow Classification
RNC	Radio Network Controller (UMTS)
RO	Rights Object – OMA DRM object
ROAP	Rights Object Acquisition Protocol
RSA	Rivest, Shamir & Adleman
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
RTSP	Real-time Streaming Protocol
SCTP	Stream Control Transmission Protocol
SDTV	Standard Definition Television

SDK	Software Development Kit
SGSN	Serving GPRS Support Node
SIM	Subscriber Identity Module
SIP	Session Initiation Protocol
SMS	Short Message Service
SDP	Session Description Protocol
TCP	Transport Control Protocol
T-DMB	Terrestrial - DMB
TDMA	Time Division Multiple Access
TSL	Transport Socket Layer
UDP	User Datagram Protocol
UE	User Equipment (UMTS)
UMTS	Universal Mobile Telecommunications System
ULTRAN	UMTS Terrestrial Radio Access Network
URI	Universal Resource Identifier
VCEG	Video Coding Experts Group
VLC	Video Lan Client
VLS	Video Lan Server
VoD	Video on Demand
VoIP	Voice over IP
WAP	Wireless Application Protocol
WCDMA	Wideband Code Division Multiple Access
WiFi	Wireless Fidelity
WiMax	World-wide interoperability for Microwave Access
WLAN	Wireless Local Area Network
XML	Extensible Markup Language

Executive Summary

This thesis proposes a new entertainment service for mobile phones over 3G. The service main feature is live event broadcasting over the 3G Network with future plans for DVB-H broadcasting. Investigations of current and future proposed mobile phone technologies and services have been done, as well as a study of Digital Rights Management.

The paper introduces a business idea based on novel theoretical framework, elements from innovation theory, investment analysis and mobile technology combined with personal entertainment experience as a DJ and music producer.

The business idea is carefully explained thru a business plan, which was developed during the 2006 venture Cup and offered to several Network providers such as Netcom and Vodafone.

1 Introduction

Downloading music and movies has become one of the biggest trends of the internet usage. For more than ten years, this has been done by using computers connected to the Internet. Many online record stores offer the latest music releases and a tremendous catalogue of back-stock. Buying music is now only a few clicks away.

On the other hand, file sharing through Peer2Peer networks such as Kazaa, Bit Torrent and Napster and illegal distribution through pirate websites are among the most common ways to get obtain free access to music and movie files today. This piracy causes the international music and film industry to suffer enormous financial losses.

On a legal and commercial level, files are not only being distributed through computers anymore, they can now be downloaded directly to your mobile phone. Communication plays a very big part of our lives and being connectable 24/7 is becoming a necessity as well as a curse.
People today are completely addicted to their mobile phones and carry them at all times. Possessing the latest technology in mobile phones is considered status, especially among today's youth, and with the growing hunger for the best and most fancy mobile gadgets, the demand for new advanced mobile technology is ever increasing.

Fast connection, ability to stream video and large attachments, more than 1.2 Mega pixel cameras, PDA support, emailing and web surfing are the latest trends.

What is the next step, so we can make the next leap?

1.1 The Idea

There are two basic ideas for this thesis, and both are based on my own experience as a performing producer. They are what I believe to be the future way to communicate with consumers and fans, bypassing stationary computers and heavy laptops to market entertainment through a multimedia experience that wasn't possible until today. Both ideas can be implemented for usage in other branches as well.

The basic idea is to define a service that offers the user the possibility to watch a live concert or performance on his or her mobile media device, which is commonly referred to as a mobile phone. The main focus is on the possibilities that live streaming introduces as a new service, as well as the way to make it available and affordable to as many consumers as possible. If such is achieved, the service will be as common and addictive as SMS and MMS. Extra features like chatting, inviting friends to join a show, subscription etc. are all included and described later in this thesis.

Additionally the service should provide high QoS and secure the media owners and providers as well as the buyer's rights. It shall also prevent illegal copy and distribution.

There is to be established a business model for the service, which includes a payment solution proposition, if possible with co-operation with one of the mobile phone companies in Norway.

The system is also to be designed on an architectural level.

1.2 The Report

The report is structured in the following manner;

Chapter 1 Introduces the idea behind the thesis
Chapter 2 Describes the Problem and Methodology used
Chapter 3 Describes the Business plan with also was my contribution in the 2006 Venture Cup.
Chapter 4 Describes the requirements for the service

Chapter 5 Describes briefly the Digital Rights Management, more specified in the Appendix.
Chapter 6 Introduces Mobile TV and Streaming Technology
Chapter 7 Informs where the information about the Broadcasting Standards can be obtained.
Chapter 8 Covers Broadcasting and Streaming Protocols
Chapter 9 Covers the Return Channel for Mobile Interaction
Chapter 10 Introduces Video and Streaming

Chapter 11 Introduces thoughts about live content production
Chapter 12 Describes the System Design
Chapter 13 Describes the System Functionality
Chapter 14 Describes the System Implementation Considerations

The Thesis ends with a Conclusion followed by References and Appendix.

2 Problem Description and Methodology

New mobile entertainment systems are becoming a big moneymaker on today's mobile-phone market. However, developing a successful service is easier said than done. To be able to understand the requirements of a successful future mobile entertainment service system, it's important to know about the main services it provides, as well as creating a demand for the service among desired customers and making sure that the world knows the service exists. Only then there is the possibility of success.

2.1 Business

The intention is to develop a service and present it as a successful business opportunity to major Mobile Network providers like Vodafone, Netcom or Telenor, in the hope that they will also see it for the moneymaker it is, based on the current interest and development of Mobile TV. This can result in the service being implemented and offered to the mobile users around the world. Alternatively the writer of this Thesis can invest or find other investors to be able to develop the service, and then offer it to mobile users in co operation with a Network Provider.

2.2 User Scenarios

Live streaming of music and video (live broadcasting)
The users can view live broadcast events all over the world. Users can subscribe to the upcoming events they want and / or get invited by other users to join a live show.

Streaming video on demand (broadcasting of past events)
The users can view past events. If they have purchased an event while it was broadcasted live, they have limitless access to view it. Additional past events can be purchased on demand.
It is therefore neither necessary nor possible to download any live events to a mobile unit.
Streaming costs will be covered and limited by subscription packages, similar the way ISPs offer access to the internet.

Streaming and downloading of music on demand
The users can pay to listen to past concerts, DJ sets and mixed compilations. Tracks and compilations can be uploaded by artists or record companies for users to purchase in the mp3 format.

Streaming of news (audio/video) on demand
Users can choose to view news items that promote forthcoming events. Users that subscribe to certain artists will be presented with relevant news when they log in.

Streaming of commercials on demand
Sponsors, promoters, artists, record companies and other parties of interest can buy commercial slots that will be streamed before, during and after a live show.

Interactive content
Users can download / purchase other interactive content while using the service.

Invites
Users can invite other users to join them for an event and / or chat session, whether they already have a subscription or not is thereby irrelevant.

Chatting
Users can interact with each other and artists whenever they are online through a chat client similar to MIRC.

2.3 Methodology

To carry out a project successfully, an appropriate methodology must be used. It should present the principles and ways of working as well as specific methods and techniques used during the developing. There are a lot of different methodologies to choose between, and most of them include a process that takes the developer step by step through the project, starting with requirements, design, implementation and testing.

2.3.1 The phases of a software project

All software projects are divided into individual phases which are arranged in a chronological sequence. This is called the software life cycle: a time span in which a software product is developed and used, extending to its retirement.
The cyclical nature of the model expresses that the phases can be carried out repeatedly in the development of a software product, till the result is satisfactory and the project ends.

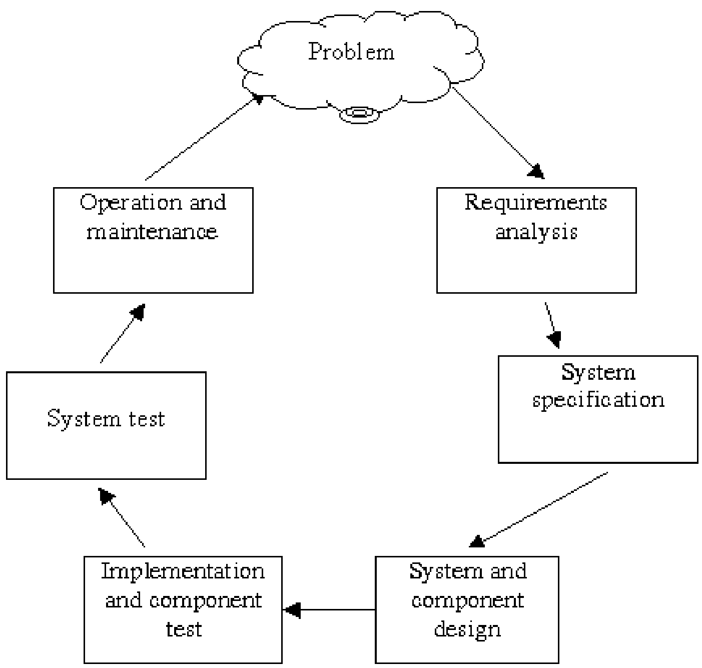


Figure 1 Life-Cycle Model [1]

Requirements analysis and planning phase
Produce the user requirements and decide what steps need to be carried out and the nature of their mutual effects. Decide what parts are to be automated, and which recourses are available for the project to be realized. After the requirement analysis is complete and the problem domain is narrowed down to the most important functions of the project, the system components must be sketched, followed by an initial estimate of the scope and the economic feasibility of the planned project. A rough project schedule should also be present.

System specification phase
Specify the system and the requirements definition. Establishing an exact project schedule and validate the system specification as well as the economic feasibility of the project.

System and components design
Design the system architecture; determine which system components cover which requirements in the system specification and how these system components will work together. Design the underlying logical data model and algorithmic structure of the system components and validate the system architecture and the algorithms to realize the individual system components. A description of the above should be presented and the design decisions should be documented.

Implementation and component test
Make the design executable on a computer by implementing the design by coding the algorithms with a programming language, compile the code and fix all errors, check the syntactical correctness of the algorithms. Then test and syntactically and semantically correct nonworking system components. Log all tests.

System test
Run the system and test the mutual effects of system components under conditions close to reality. Detect and fix all errors that occur and fix and make sure that the system implementation really fulfills the system specification.

Operation and maintenance
Maintain the software by fixing and documenting all detected errors during real-time system run. Modify the system if needed and make sure it provides the desired quality. Quality assurance includes analytical, design and organizational measures for quality planning and for fulfilling the following quality criteria;

- correctness
- reliability
- maintainability
- portability
- User friendliness.

In this project we use the Waterfall model [2] because if any, then only parts of the system will be implemented. The steps of the Waterfall model will be repeated like in the classical sequential software life-cycle model till the desired quality of the project is met.

The Waterfall model represents an experience-based refinement of the classical sequential software life-cycle model and introduces iteration between the phases along with the restriction of providing iterations, if possible, only between successive phases in order to reduce the expense of revision that results from iterations over multiple phases. It provides for validation of the phase outputs in the software life cycle.
This approach modifies the strictly sequential approach prescribed by the classical life-cycle model and advances an incremental development strategy. Incorporating a

stepwise development strategy for the system specifications and the system architecture as well as phase-wise validation helps to better manage the effects of poor decisions and to make the software development process more controllable.

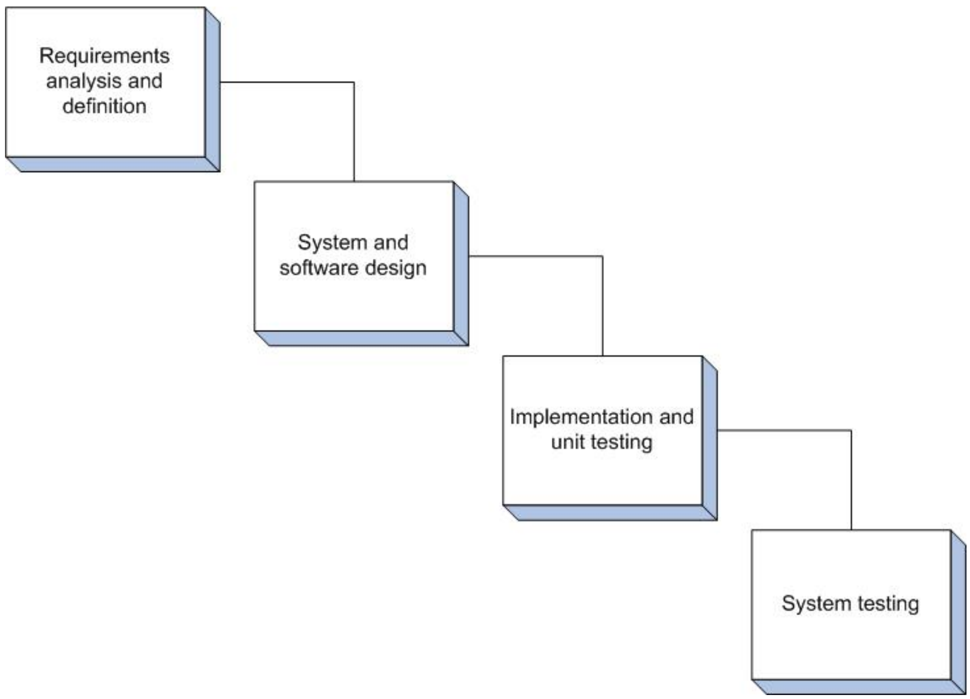


Figure 2 The Waterfall Model

3 Business Plan

The Future Mobile Entertainment Service (FUMES)

3.1 The Business Idea

To profit on enabling consumers to attend virtually any desired concert or event (music, sports, opera, seminar etc) at any desired time, regardless of their whereabouts by simply logging on to our service and purchase an extraordinary real-time multimedia event experience using a mobile phone

3.2 The Vision

Mobile phones, mp3 players and PDAs are today's standard equipment for busy people. Now they are about to presented for a completely new type of mobile technology that will change the way people think about entertainment, communication and information: Mobile TV.

Enabling busy people to experience the absolute best within live entertainment through Mobile TV, will make mobile TV exploit its advantages and be a goldmine for the content providers.

3.3 Mobile Entertainment Today

Mobile phones evolve quite rapidly all the time, and have become more and more advanced when it comes to design, software, extras like cameras, mp3 players, video streamers and screen sizes. Even video and picture quality have increased tremendously. People use the phone for absolutely everything these days and it has become a substitute for many other devices we use in our daily lives, like portable mp3 player, cameras, radio and even the portable DVD players. The main task of a mobile phone is still to communicate with other people, but with so many providers on the market pressing the prices down and the cheap or free IP calling communication systems to compete with, the network providers need to focus on other means to make long term profits. We already see that ring tones have become a huge success all over the world, and in Norway the turnover for this service has been more than 100 million NOKs last year alone. According to Dagbladet [3] the mobile network providers are now focusing on music, games and video to expand their services and keep the customers entertained.

Due to the recent changes in people usage of the mobile phone where the entertainment is becoming more and more focused on, the product I am developing is a service that provides an extraordinary real-time multimedia experience on a 3G mobile phone, a service that hopefully will be highly common on the 3G mobile market within a short amount of time.

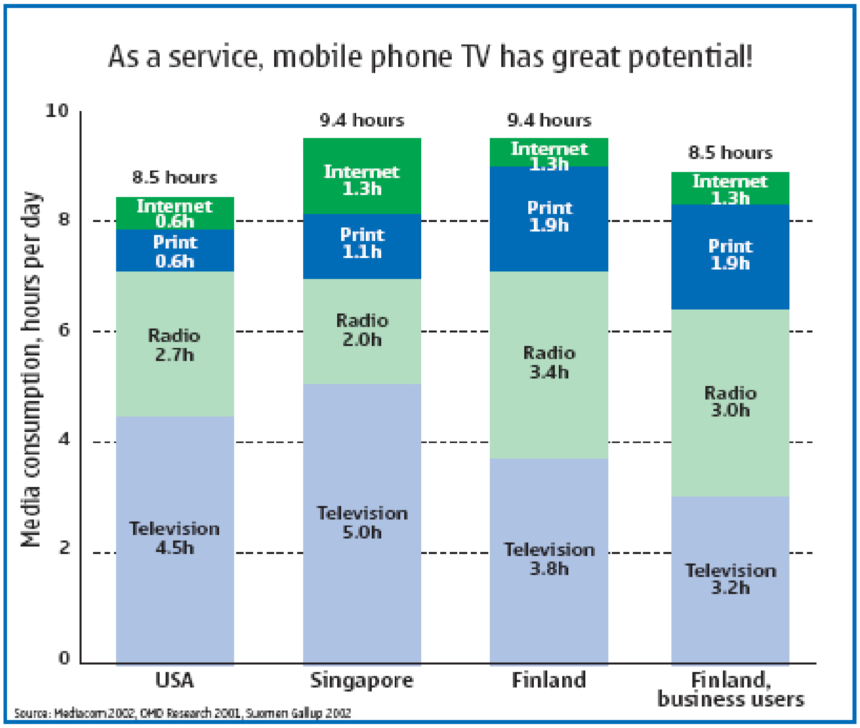


Figure 3 Mobile Phone Service Potential [4]

Entertainment plays a big role in most peoples lives [4], may it be something as simple as watching TV, listening to the radio, going to an opera, football match or gaming on the portable Playstation. Where ever you look, the bottom line is that people want to be entertained one way or another. Mobile phones are among the most popular personal technologies around these days.

According to Netcom GSM almost every person from 15 years old and up owns a mobile phone in Norway alone. The ability to take the mobile phone anywhere makes this gadget one of the ultimate entertainment toys of the century.

Based on information from Statistisk Sentrabyrå [5], 85% of the Norwegian population between 9 and 79 years old, have been using a mobile phone during 2004. In some countries like Japan, and USA every 1/3 person owns a mobile phone. Considering that today's generations are sort of speak 'on the run' at least 80% of their time and in Norway alone 72-95% of the people between 16-44 years use their mobile phone daily for chatting, sending SMS and MMS.

According to the online Tech-On magazine, "Macromill, Inc., a research company, recently conducted a survey targeting users of mobile phones supporting Japan's mobile TV broadcast service. To a question how often they use the service, the largest stake of 26% answered "2-3 days a week," followed by 21% of "almost everyday." [6]

The research company also asked about the genre of programs the users use the Mobile TV service for, “an overwhelmingly large portion of 76% answered "news and reportages," followed by 36% of "variety shows" and 35% of "sports." Asked if they plan to enjoy the "FIFA World Cup Germany 2006" to start on June 9, 2006, through this broadcast, a total of 50% respondents showed their intention to use the service, combining 11% of "I certainly will" and 39% of "I probably will." “.

We already see that the there is a lot of money to be made on music distribution, but due to piracy and lack of registration of the music used in clubs, on events, pirate broadcasting etc a lot of the producers barely make any money. The labels have moved from vinyls to CDs over the years and now to mp3 which offer new ways of music distribution and money making.

Currently more than 1 billion NOK is spent annually on buying short ring tones for high amounts of money (15-30 NOK per 10 seconds of music) [3]. More money is made on ring tones than i-Tunes makes on selling full version mp3’s. Of curse the huge commercial spamming on basically all TV channels and magazines makes people spend money on this service without even thinking twice, usually of plain boredom while on the buss stop or in queue at the store. Media like TV and newspapers all over the world constantly focus on the shopping trends among youth. They all come to the same conclusion, the fact that youth spend money on things they think are cool, hype and gives them status in the eyes of other youth. One example is a simple charm chain to decorate their phone with. Despite the fact of it being completely unnecessary, if a celebrity has it, the fans want it too. The mobile charm chain is a simple and huge money maker!

Mobile phone ownership and usage are still growing dramatically on a global basis. The around 2 billion paying customers out there are using their mobile phones for more than communication, namely multimedia, entertainment (video streaming, music and games), news and information services. Mobile TV is easy to understand and use and there is already a genuine consumer interest for this type of entertainment. Nokia estimates suggest that “around 20% of active mobile phone users are highly interested in acquiring the service and prepared to pay a realistic charge for it – around 10 to 12 euros a month”.

However one should take note of that 70-80% of the operators make most of their income on Voice. The following figure shows TeleSoneras [108] Mobile Service revenue based on the Nokia Mobile Service forecast made in 2004.

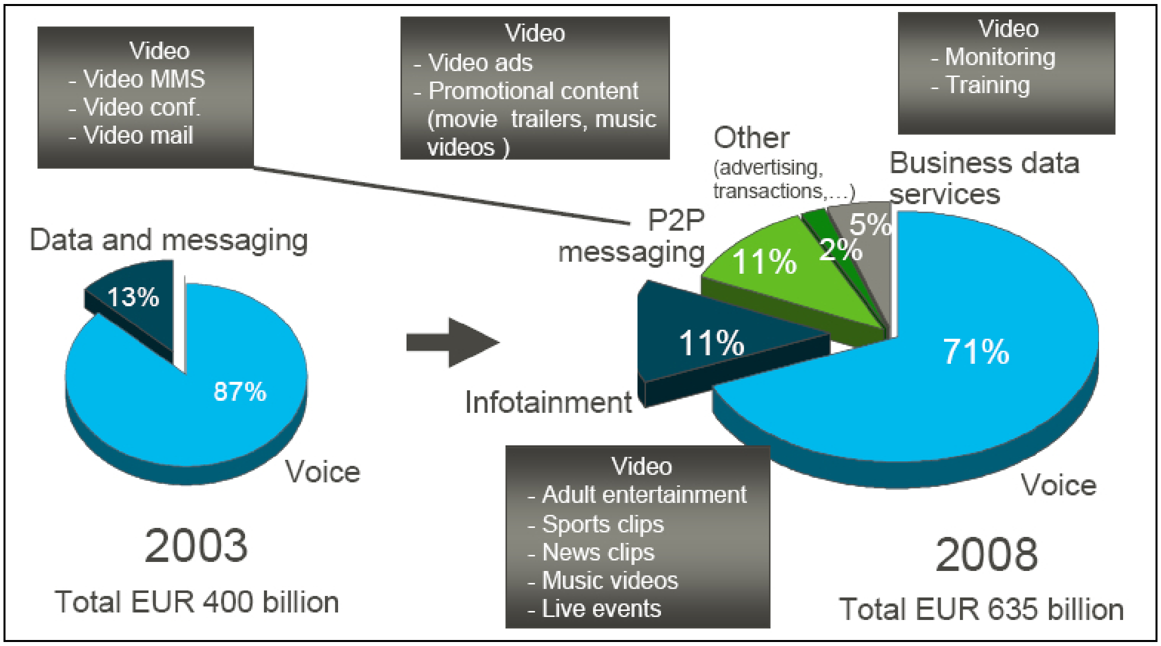


Figure 4 Global mobile service revenues 2003-2008 [7]

For example in Finland, in addition to the basic services, the following Mobile TV channels are provided for pilot test users;

- MTV3
- YLE TV1
- Nelonen/Channel Four Finland
- Other theme channels

"In addition to the basic service, users can subscribe to other services, such as news or sports programs.";[8]

In October 31st, 2005, Vodafone UK and British Sky Broadcasting (Sky) announced an agreement to launch a mobile TV service in the UK. The very next day the new service - Sky Mobile TV was a reality. The service offers a wide choice of channels, including 24-hour news channels and sport, entertainment as well as documentary programmes, to all Vodafone live! with 3G customers.[9]

- Vodafone live! Offers today 19 TV channels including;
- Sky News
- Sky Sports News
- MTV,
- Cartoon Network
- Discovery
- Sky One
- Living TV

There are several different customer groups when it comes to the usage of mobile services; children (up to 12), teenagers (13-18), students and grownups. It's important to split the customers in different groups because of their different needs, ability to pay for a service and the method of payment. Things like income, social status, nationality, surroundings, education, wealth and interest also have an impact.

To sum up, I would say that most consumers use more and more advanced mobile technology services, also multimedia services. However, most of these have been offered as "nice-to-have" services" more than giving the consumers the feeling of "need-to-have". This will be the challenge and mission of FUMES.

3.4. The Service

The service will allow users to experience live dj shows and concerts in real time using their mobile phones. Additionally it will allow subscription to events and content of interest all around the world. User will also be able to interact with each other as well as the artists during the live broadcasting using the services chat functions. We deliver top quality sound and video real time, with standard methods for payment.

The service is based on the Mobile TV Broadcasting solution, such as for example provided by Nokia and is unique by being the only service today providing extraordinary real-time experience on 3G and in the future DVB-H.

To be able to use the service, a user *must* have an officially registered SIM card with one of the mobile network providers in their name as well as access to a 3G/DVB-H mobile phone . Without the specific SIM card in the phone as well as the user's password which is linked to that SIM card, the user will not be able to log onto the service and the users profile.

Once the user has an officially registered SIM card in his or hers phone, the user uses WAP to logon to a website which runs a check in the mobile providers systems to see if the current SIM card belongs to the claimed user and the users current phone is the one registered with the current SIM card. If those criteria match, the system registers the user and automatically sets up the phone for the usage of the service.

There are several groups involved;

- User (also known as a fan of a live act)
  - User (regular user of the system)
  - Artist
  - Promoter
  - Recording Team
- System Administrator
- User Administrator
- Music Licensing Administrator
- Mechanical Rights Administrator
- Content Administrator
- Service provider
- Content provider
- 3rd Party, also known as companies and private people that want to sell adds etc.to the service).
- Content Operator

Seen from the common users' aspect, the service provides several interesting user options on top of the possibility to view live events, some of those are;

- Possibility to subscribe to events/artist.
- Interaction similar to IRC /MSN with other users / artists during the live events.
- Possibilities to invite others to view a particular event where the inviter can choose if he/she wants to just send an invitation where the other party has to pay for the viewing themselves or send an already prepaid invitation.
- Teasers for first time users, where the user gets to view between 10 seconds and 5 minutes for free of a show before being offered to pay to continue.
- Prepayment of upcoming events.
- View & buy past event.

- Search engine.
- Sale of merchandising items, such as DVDs, CDs and clothing.
- Sale of interactive content.
- Viewing sponsor messages (advertisement)

Payment can be made in five different ways; Direct payment with a credit card, the payment can be added to the phone bill (mobile operator subscribers only) or charged directly from the prepay card. The service checks the prepay card / Credit Card to see if there is enough money on it before it charges it. If the user wants to add the charge to his/hers phone bill, the service first checks the users maximum credit on the bill, and that the charge will not extend the current bill
(a limited maximum amount will be set). The user can also use his credits that he has gained through promotion of events done by inviting others to join an event or buy other content and the users own purchases. Last the user can use his PayPal account to buy content. For every time someone, that the user has sent a link or invitation to purchase content, buys the content, the user gets rewarded with credits. Credits are also obtained through prepaid cancellations.

The other groups have same user options as the common user, but with extended menus serving their purpose according to the roles they play in the system.

All legal-, mechanical- and licensing fees will be paid according to BIEM regulation.

All legal-, mechanical- and licensing rights will be taken care off according to the universal music laws in Europe and it will be made sure that the correct party gets paid accordingly.

Illegal distribution of content as well as copying will not be possible, as the events can only be streamed, not recorded on a portable unit. Also when viewing past events, the SIM and the phone which is registered with a certain user profile must match.

Backup of the system should be available through internet, but this is a temporary suggestion.
There must also at all times be backups of the content which is being added to the content database.

It is important that the service is affordable by the average youth, so the idea is to operate with fixed prices pr event combined with bonus systems, for example buy 4 get one for free etc. By giving young customers a possibility to experience their biggest stars live and maybe end up chatting to them, even if it’s just through their mobile phone, it will be a huge personal experience.

3.5 The Customer

The main targets of the service are groups of customers which are very often too young to go clubbing or to a concert or simply can't afford it.

According to a presentation [10] during Forskningsdagene in 2002 at Høyskolen I Molde, Per Bjarte Solibakke writes "as late as in 2001, 42% of youth aged 15-24 had a part-time job and 38% of the youth wanted to have a part-time job". Most youth under 15 get pocket money from their parents who also in most cases are responsible for getting their children mobile phones to be able to be in touch with their kids in the first place. Youth above 15 usually have part-time jobs along studies and or scholarships from Statens Lånekasse. Others either get full time jobs which provide them steady income or they become unemployed and live on support from the social service. Still about 99% have a mobile phone. By keeping the price for the use of the service low, everybody will be able to afford to use it.

Today's teens spend most their money on clothes and gadgets that make them cool and trendy in the eye of others. To have the coolest mobile phone and being able to use such service and share it with their friends will quickly make the person owing it the coolest of the group. No matter how little money the average person has, he or she still has and uses a mobile phone.
The power of being trendy and up to date within certain things on the marked should not be underestimated.

To better understand the need of such service, consider the following scenarios;

1. Mobile TV is here to stay and will become the most used mobile entertainment service in the coming years. Mobile owning customers will be using this service one way or another.

2. To be able to make money on mobile entertainment services, one must develop unique services that can take advantage of the already existing technology and make it a *must have* service for the customers.

3. Users under 18 aren't allowed access to concerts is a large number of countries around the world. Still, children between 11 and 19 are the main target group of artist concepts such as Britney Spears and sorts. At this time, children under 18 have to wait for video registration to become available in stores or on television.

4. Users that are in fact old enough to attend a concert or event of choice may not be able to afford the usually high entry fees, or the particular event might be sold out before they have a chance to obtain a ticket.

5. The service will enable it's users to travel virtually to an international event of choice. Artists on tour often attend more than one country. Fans can now witness a whole tour, instead of just the edition in their hometown.

6. It would be possible to attend virtually a Dance music festival, such as Sensation in Amsterdam, with a potential live audience of over 40.000 people, Love Parade in Berlin, with over 2 million visitors on it's last edition, or Tokyodome in Tokyo, an event that most Europeans or Americans would probably never even think of to attend, if it wasn't for this service.

7. If we look beyond teenagers and students, there are many people around the world that travel for either business or pleasure. Instead of paying a lot of money for a hotel movie service, they can now participate virtually in an event of choice, regardless of their whereabouts. They can witness a concert or event while they enjoy a cocktail on the beach or in the private surroundings of their room.

8. For artists and record labels, aside from generating more income, this service offers a whole new way of communicating with fans. The interactivity will make the connection between artist and audience more personal, more direct and intimate. Additionally, the live-element will offer a new means of promoting and even selling merchandising while the event is in progress. The option of live chatting with the artist involved, will take the demand for this service to a whole new level.

3.6 Today's Service Situation and Future Plans

Since the technology is rapidly evolving, the main functionality of the service 'live video/audio streaming' already exists. Like mentioned earlier, Mobile TV is the new entertainment that will hit the marked hard and hence the service in this thesis has good possibilities of become a major player in the entertainment business. The issue here is to mount this with the other desired functions in such way that the service in this Thesis becomes unique and wanted by the customers. Already several DJs that have been asked to air their opinions about the service, are positive to the possibilities to be able to promote themselves by direct interaction with fans as well as being able to offer their music productions to fans more directly.

3.6.1 Mobile TV Forecasts

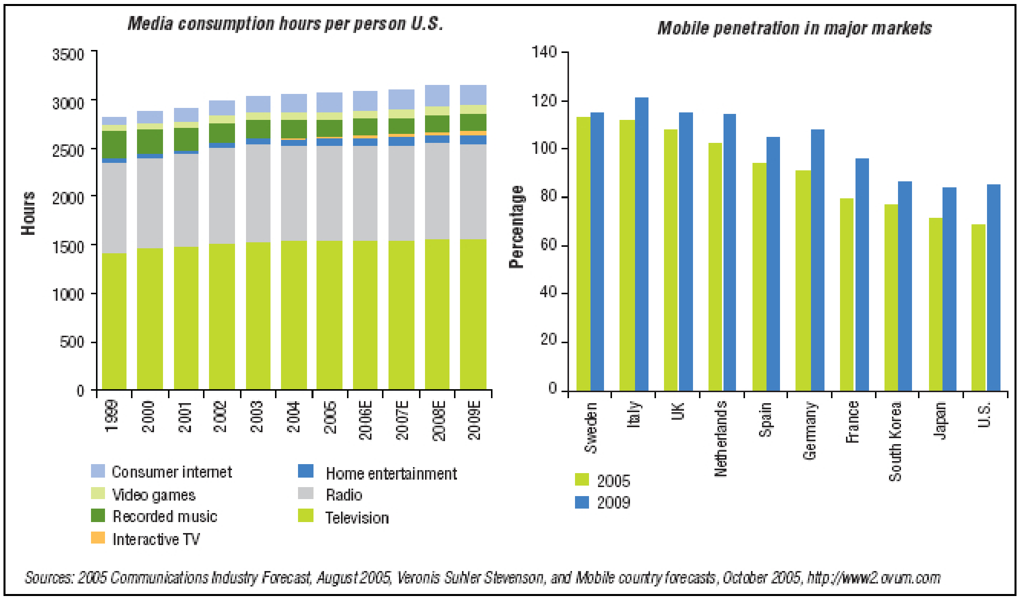


Figure 5 TV consumption and mobile penetration

Figure 5 illustrates the TV consumption and mobile penetration, based on the Communications Industry Forecast, August 2005, by Veronis Suhler Stevenson, and Mobile country forecasts, Ocotber 2005 [11] shows clearly that consumers invest more time and hence money on mobile phone usage.

Despite incompatible standards, vendors and developer around the globe believe that Mobile TV will live up to its promise based on the new digital broadcasting technology. Backed by a wide range of research, it is predicted that the market will grow to between 5 and 27 billion US $ worldwide by 2010. An example is Datamonitor [12] who is convinced that they will have 69 million MobileTV subscribers globally in 2009, generating revenues of around US$5.5 billion. Figure 6

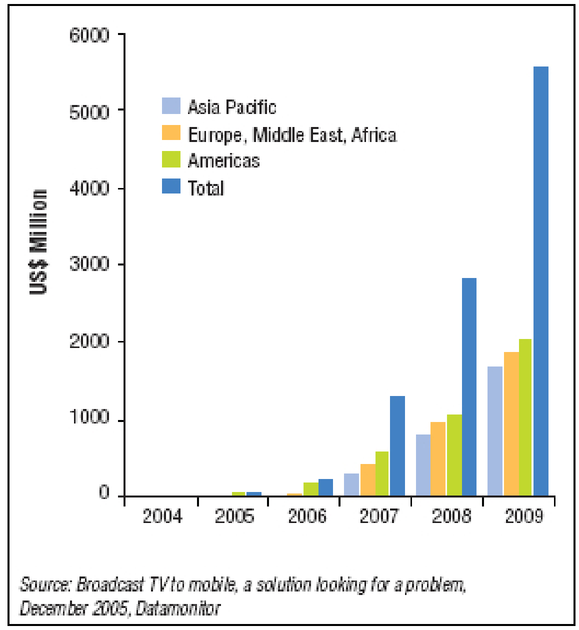


Figure 6 Subscribers and Revenue forecast for Datamonitor

3.6.1 Development Plan

The developer and owner of the idea behind the service is Magdalena Lipska, who is responsible for developing and designing the service and it's functionalities on an overall architectural level as part of her Master Thesis at NTNU 2006. The thesis is due 14.06.2006 and is supervised by Professor Leif Arne Rønningen and adviser Kristen Rekdal. If time permits, Lipska will start to implement the most needed basic functions so that the service can be tested properly on one or several dj events during summer.

The next step is to try to patent the service concept and then sell it to one of the bigger network providers in Europe. to be developed and implemented into their system. The network provider can then expand the service.

Lipska is currently looking for a team who might help her realize the idea. Experienced entrepreneurs and telco experts will be asked to at least join her network of advisers. A commercial CEO and Vice President of Sales and Marketing will be recruited, and a company will be incorporated with the support of Innovation Norway (Invanor).

The concept would be faster realized with a huge partner like e.g. a network operator like Netcom or Vodafone. Lipska is currently in process of pinpointing potential business partners.

3.6.2 Service Functionality

The most important goal of the service is to be able to provide what the service is promising; great entertainment in the form of live event broadcasting with optimal sound and picture quality. The service is to offer high dependability, availability and quality of service.

Second the service *must* take care of all the legal aspects of music and the payments to the artist and make sure that everybody else who get a piece of the cake get a fair share according to the role they play in the event. Even the tiniest errors when it comes to the mechanical rights and licensing can shut the service on the spot.

The service must also be cheap to use, especially in the testing period which can expand over a half years time. Most 3G services offered today are free of charge (3G videoconferences offered by Telenor are free of charge till end of March 2006 to get the customers hooked on the service).
One possibility to offer cheaper events is to allow commercial popup or banners between, before and right after an event is done. This can only be allowed during concerts and other events where there are natural 'breaks' included in the program. DJ events and raves have no breaks, as the artist just changes on the last record spun usually with no or a short intro of the next artist.

Phones evolve month by month and the amount of extra gadgets and accessories one can buy for a mobile phone are massive. Therefore an idea has presented itself which in the future allows the user to be able to connect the mobile unit to a LCD HDTV (High Definition TV) to view the service on a big screen at home, or use 3D goggles while being 'on the run' and have the same great experience on a train or buss. Having both these possibilities and be able to connect them through a wireless solution would be great and the user could experience the entertainment wherever whenever! Since the screens of the mobile units are increasing in size as well as the screens pixel resolution is growing rapidly, the picture a user views on the phone itself already has a very good resolution. Another desired future functionality would be to have same zoom options on a mobile phone as the one on the Cannon Digital Cameras. With that, a user could zoom in the desired part of the complete frame being streamed.

In the long run this service can be modified to be used for any other purposes, like stream live sports events (currently offered by others), worldwide conferences, lectures at universities and other similar events. Artist blogging can also be a part of the service.

3.6.3 Promotion

The service needs to be promoted directly towards the potential customers and at least as hard as the Jamba ring tones but less stalking, so that is will not scare the potential customers away. Radio stations can be used to promote the service in the same way that for example radio NRJ and Radio1 promote concerts and happenings. There should definitely exist a web portal that explains the service and its functionalities etc. Record labels and artists that agree to stream their event through the service will include this feature in the promotional media for their tour or event. No better promotion than an artist telling his/her fans to get online and participate. Adds in online magazines like VG and Dagbladet, article in IT Magasinet will also be valuable promotion in Norway. Extreme stunts can be used during the summer in the big cities in combination with stands that promote certain events both abroad and in Norway.

The best promotion will still be made by the live events and the promoters who are hosting them. All promotional material regarding the events as well as the information about the possibility to view it through 3G/DVB-H will be promoting the service through the events website, tickets, posters and likely on the artist websites and through other media.

3.6.4 Sponsors

Sponsors will be offered to have banners and similar that can be shown in between the breaks of concerts and before and after the beginning of a content stream. The sponsors will also be promoted on the service's website. The sponsor money will contribute to the service providing minimal usage costs for the users.

3.7 Competitive Advantage

The service concept is unique in the user experience that this service offers, by giving a complete feeling of being physically present at the event or concert.

The customer can follow their artists all over the world and don't limit themselves to see one or parts of a performance that for example is cut down to a DVD or even broadcasted live on TV. Where other services limit to a certain spot or place (radio / television) or other media are restricted to the availability of other players (online gaming), there are no limitations restricting your ability to virtually travel the world at any desired time.

Therefore other entertainment competitors like Cinemas, DVD's, Movies on TV, online gaming and similar are poor compared to the feeling of actually being part of the event. The service offers more than just the picture and sound, it offers an emotional experience of actually being there of live involvement even if it's only electronically.

3.8 Goals and Strategic Decisions

- Target the following first five customers with the following yearly events:
  - ID-T, The Netherlands [13]
    - Sensation Black & White (First two weekends of July)
    - Innercity (3rd weekend of December)
    - Thunderdome
    - Trance Energy (New Years Eve)
    - Mysteryland (August)
  - Q-dance, The Netherlands [14]
    - M Q-Base
    - Defqon.1
    - Qlimax
    - Inqontrol
  - UDC, The Netherlands [15]
    - HQ
    - Dance Valley
    - Impulz
  - Event Makers, Norway [16]
    - Tidy Events
    - Dance Arena
  - Wonderland, Norway [17]
    - White Edition
    - Black Edition
    - Wonderland
- Be open to different business models depending on local conditions with relations to network operator, content owner and other regulators
- Aim to be the number one Live Mobile TV enabler in Europe by 2010
- Revenues of more than 10 MNOK by 2010

3.9 Profit & Loss

This is based on a simple business model – a cut of 20-25% to the content owners and broadcasters - and an increasing price starting at 20 NOK per event per user ending at 40 NOK after 3 years. The number of users per events is somewhat high, which means that powerful marketing is essential.

Since most of the technology for live broadcasting as well as Mobile TV is already available, the main costs of the service will be to develop the applications that the service will be running on. Implementation and design for a pilot customer is estimated to be around 500.000 NOK. The costs for usage of the broadcasting channels have to be added to the content costs. Customers can subscribe to content on regular subscription basis or pay pr view/download.The events are planned to be sold at a maximum price of 30-50 NOK pr event and the service will also offer the users to buy real event tickets keeping the sales provision for those as well as other content.

Profit & Loss (NOK)	**2006**	**2007**	**2008**	**2009**	**2010**
Revenues					
NOK per event per user	20	30	40	40	40
Number of events	3	7	20	50	80
Number of users per event	1 000	2 000	5 000	5 000	5 000
Content and broadcasting costs	20 %	25 %	25 %	25 %	25 %
Total revenues	48 000	315 000	3 000 000	7 500 000	12 000 000
Costs					
Implementation and design					
Prototype	500 000				
Technical development & equipment	300 000	300 000	300 000	300 000	300 000
Staff	300 000	1 000 000	2 000 000	2 000 000	2 000 000
Travels, marketing, etc	500 000	300 000	300 000	400 000	400 000
Total costs	1 600 000	1 600 000	2 600 000	2700000	2 700 000
Profit	**-1 552 000**	**-1 285 000**	**400 000**	**4 800 000**	**9 300 000**

Table 1 Profit and Loss

3.10 Risk Factors

Economic risks:
Content owners or broadcasters are too expensive or want to deliver their service on their own

Patent risk:
We might not get the patent rights to this service concept

Premature service:
The service must be fully developed to establish the “need-to-have” feeling among young people worldwide

Regulatory risks:
Too difficult to establish a global service delivered in different countries with different operators, costs, legislations, cultures, etc.

Partner risks: Not getting the right partnership relations in the initial phase

Operational risks: Not being able to attract talent to run the business

4 Requirements

For the scenarios mentioned in chapter 2 to be usable, the system has to fulfill several requirements. [18]

There are several different users of the multimedia system;

1. The *operator* that owns the main operating system that allows other smaller service providers to offer this type of service. In our case Telenor is the company that owns the Norwegian network.

2. The service *provider* that offers the multimedia service, such as Netcom, Vodafone, Chess or Tele2 who rent the network from Telenor, or Telenor itself.
3. Administrators that take care of the system and control the usage by the other users
4. Regular users also know as fans, hereby reffered to as ‘users’
5. Artists such as DJs (Tiesto, Miss Jarea), Bands (Linking Park, Nightwish) etc.
6. Promoters such as ID&T, Q-dance (NL), EventMakers (N), Clubs & Artist managers / promoters (DJ Booking Norge)
7. Others / Enterprises, companies that buy advertisement spots within the breaks of the live shows

Note: The service will be offered by a service provider such as Netcom or Vodafone. Broadcasting TV channels like MTV can function as promoter of the events they are broadcasting live and in such way offer the event on behalf of the artists and the TV station having all rights to the content.

4.1 User Requirements

4.1.1 The user

The user should be able to do the following with the system

1. Register
2. Log On
3. Choose a Service
    a. See Live Show
    b. See past live shows
    c. Edit and manage Live Show Schedule
    d. Browse archives with past events
    e. Search for Artists & Events
    f. Edit and manage user profile
        i. Personal details
        ii. Payments details
        iii. Subscribe to events
    g. Invite friends to preview the live shows and receive a bonus (marketing)
    h. Invite friends to join the live show by paying for them and receive a group discount
    i. Chat with friends while listening / looking at the show
    j. Chat with the artist
    k. Receive credits for inviting others to use the service
    l. Pay for the usage of the service
    m. Pay for others using the service
    n. See payment history
    o. Listen to music (dj sets & recordings in mp3 format, can be combined with personal slideshow of the pictures the artist has uploaded to the service)
4. Log Out

4.1.2 The Artist

The artist should be able to use the system in the same way as the **user**, but with additional extras that benefit the artist

1. Add old private shows to directory
    a. Must be checked and approved by the MR- and Ladmin
2. Add DJ sets mp3
    a. Full track list must be provided and license agreements with ALL labels / producers / material used to the Ladmin.
3. Add payment details for each *private unlicensed productions* upload
4. Edit and manage Artist profile
5. View download statistics and History
6. View balance account
    a. Money made on unlicensed products
    b. Money made on licensed products
7. Log Out

4.1.3 The Promoter

The promoter should be able to use the system in the same way as the **user**, but with additional extras that benefit the promoter and the artist.

1. Add new event
2. Add tracks that will be performed on the event
    a. A group or artist like Madonna usually knows what tracks she will perform
        i. This to make sure the money is paid to the right producer/composer/writer
    b. A DJ that is performing live, must either give the promoter a track list right after the performance or the promoter needs to have a person that will me writing down the records as the DJ finishes with them
        i. This is however a really annoying thing for a DJ, to be hassled with after or upfront a gig. If the DJ plays a preplanned set, this should be easy, else a clever solution must be found such as giving the task to the event promoter to organize the play list after each performance and hand it over directly to the Recording Team.
3. Add payment details for events held by the promoter only
4. View download statistics and History of the events
5. View balance account (money earned)
6. Log Out

4.1.4 The Recording Team

The Recording Team should be able to use the system in the same way as the **user**. Additional requirements;

1. Use only recording equipment approved by the Service Provider
2. Run Live Broadcasting Tests
3. Stream Live Events on behalf of the Promoter, Artist or approved 3rd parties
4. Add other approved, prerecorded events and videos to the content database
5. Report directly to the Content Administrator

4.1.5 The User Administrator

The Administrator groups should be able to use the system in the same way as the **user**, but with additional extras that makes them the watchdogs of all the users.

1. Keep track and administer all user groups
2. Administrate user rights
3. Report to Service Provider

4.1.6 The Mechanical Rights Administrator (MRAdmin)

1. Keep track of the music being used
2. Keep track of the amount of usage for every track / product
3. Keep track of the money made on each track/product sold
4. Keep track of how much money is to be transferred to each artist based on the sales and the producer/writer/composer % division for every track
5. Report to CAdmin

4.1.7 The Licensing Administrator (LAdmin)

1. Keep track of the music being upped and confirm the license agreements
    a. This can be done by the artist/manager faxing the license agreement to the Ladmin
    b. Or the LAdmin checks himself with the record company that the product indeed is licensed where it's claimed to be, as every upped product will have a form attached to it where details like this are registered.

2. Report to the MRadmin
3. Report to the CAdmin

4.1.8 Content Administrator (CAdmin)

1. Approve Content verified by both Ladimin and MRadmin
2. Approve testing of Live Broadcasting prior to Live Events
3. Allow approved and scheduled Live Event Broadcastings to be directly broadcasted
4. Add only approved content to the Content Database
5. Report to the Content Provider

4.1.9 The Service Provider

The service provider has to specify and provide;

1. High quality of Service
2. Good maintainability
3. Possibility for *others* to sell advertisements that will be broadcasted in between breaks
    a. (Concerts only, as on DJ events there are no breaks…)
    b. Report to the Service Provider

4.1.10 The Content Provider

1. Is responsible for all content added to the FUMES content database.
2. Is responsible for keeping backup of FUMES at all times and on several storage devices placed in different geographical areas.
3. The content provider is also responsible for making sure that the Content Administrator has only added content approved by LAdmin and MRAdmin and is according to the content requirements.
4. Report to the Service Provider

4.1.11 The Operator

The operator has to make sure to provide a > 98% working network by applying operational rules and satisfying a specified quality of service.

4.1.12 3rd Party

Other 3rd party users such as companies or private people that want to sell advertisement between the breaks of the live events. Also small banners can be included on the search page and upon start of the live broadcast and at the end. 3rd party interactive content.

4.2 Content Requirements

The content added by the user groups; Promoter and Artist, as well as 3rd parties has to be approved by the Mechanical Rights Administrator and the Licensing Administrator.

The artists *must always* provide a full detailed track list of the music they are using in dj mixes, compilations, ring tones, commercial promotion etc, *regardless!*

3rd parties *must always* provide full details of the music they are using to add video/audio content to the content database and the content *must* be verified and approved by the Administrators.

The Recording Team *must* use equipment approved by the Service Provider to ensure the quality of the content.

The *must* at all times be at least 2 other backup sources of the content database stored in different preferably geographical places, under strict supervision and security.

4.3 E-Commerce

The content will be transmitted via Broadcast Network Operators' infrastructures directly to consumers' handsets and the Mobile Network Operators' networks are to be used as a return path to their current data, billing and audience measurement services, via e-commerce systems. This way the Mobile Network Operators will gain new revenue streams through offering billing and e-commerce systems to partners.

Note that for the future DVB-H solution, the IP Data cast is linked with the billing and e-commerce
systems of the Mobile Network Operator, and provides the Interactive User Guide (IUG) which moves viewers seamlessly from one broadcast channel to another, and between the broadcast environment and the Mobile Network Operators' cellular services.

4.4 System Requirements

4.4.1 Entire system

The primary requirements for the entire system it to work properly are the following;

1. The entire system must be able to handle users at different speeds and cover large areas in different geographical locations both in and outdoors.
2. The system should be able to broadcast over different frequency bands used all around the world
3. The system must be able to handle handover and roaming
4. The system must be designed for future applications, future modifications and future use
5. The systems main focus is the live audio-visual content
6. The system must have a good backup system containing several backup servers

4.4.2 Provider / Live- / TV content → Mobile Device

There are several important aspects of the way information is transmitted from the broadcaster to a receiving mobile phone unit. Most important are;

1. Possibility of broadcasting through the downstream channel
2. Downstream capacity must be able to handle TV- and interactive content
3. Downstream delay should be < 1ms
4. Backup of streams before sending
5. Possibility to secure transmission

4.4.3 User / Mobile Device → Content Provider

1. The whole interactivity channel should be IP based, both mechanisms and packets
2. The interactive channel must be able to handle everything from simple SMSes between users to the live video/audio feed back to the provider.
    a. A person that wins a ticket etc. can be broadcasted live on a huge screen during an event and for example be able to thank the artist like they for example do on MTV awards when an artist is not present to receive the award

3. The return channel must be duplex to handle personal information, content that is private for a user, keeping it private form the rest of he broadcast group.
4. The system should be able o handle requirements of interactive applications that can be desired in the future.

4.4.4 Digital Rights Management

DRM must offer encryption and mechanisms for content access. [19, 20, 21] For more details about DRM, please see chapter 5 and Appendix.

Security Requirements

1. Memory and File Management

    a. Ensure controlled access to file and memory
    b. An access-controlled file system is needed
        i. To store decrypted content
        ii. To keep state
    c. A secure memory system is needed
        i. To hold temporary sensitive data e.g. decrypted keys
    d. Need for memory separation system
        i. To ensure that if a trusted operation is running, non trusted operations cannot eavesdrop on the memory

2. Cryptographic Operations

    a. Protected content encrypted using the symmetric key algorithm AES
    b. License is bound to the content file using a hash of the content file
    c. Public-key operations such as RSA or ECC for content-key encryption, signature generation and verification etc.
    d. An embedded device like a mobile phone may not be efficient for using these algorithms in software. Specialized hardware can improve performance.
    e. If done in software, ECC shows significantly better performance than RSA for encrypting/decrypting the content key.

3. Credentials Manager

    a. Software module for securely handling database of phone's credentials
    b. Signature verification can use certificate chains which should be preferably small
    c. Keys should only be usable by OS components, decrypted into secure memory
    d. Should handle adding new certificates or revocation

4. Primary Credentials
    a. Permanent Identifiers
    b. Root key
    c. Unit keys and certificates
    d. DRM-specific keys and certificates

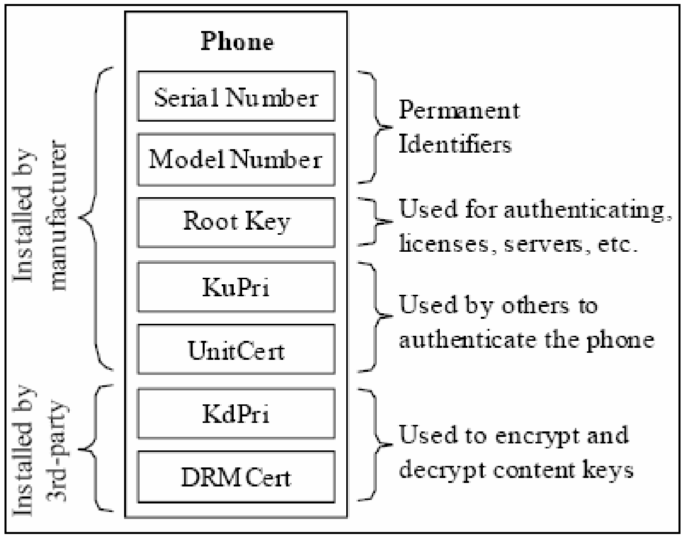


Figure 7 phone credentials [22]

The figure show the phone's credentials which consist of permanent identifiers, a root key, private/public unit keys, and private/public DRM keys. The unit keys are used to authenticate the phone and the DRM keys are used to assign content to a particular phone.

For full information about the OMA DRM requirements please see OMA DRM Requirements [22]

4.4.6 Dependability & Maintenance Requirements

Reliability

- The system is to provide an uninterrupted service
- The live stream team providing the live stream at a certain event must be reliable and do everything by the book
- For every event there must be a standby backup live stream team and equipment

Availability

- The system is to be available at all times, keeping the availability factor close to one.
- The service is to run without the occurrence of any type of failures and / or faults
- The system should be fault tolerant
- There should be a backup solution if the 3G net is down

Maintenance and Repair

- The system should be restored as fast as possible after a crash, keeping the users on 'standby' with for example images of the artist etc. until the service is up again.
- Maintenance on the system should be done when no live events are scheduled, as the users will be downloading data from stationary servers during that time.
- Repair of the system must be done immediately.
- Maintenance of equipment and backup equipment used for the recording and live streaming must be done regularly

Security

- The team providing the live stream are responsible for all data being stored and streamed
- The provider of the service is responsible for keeping all data stored and locked
- All data that is flowing through the service must be encrypted with safe encryption, this includes digital rights, billing, payment and all other sensitive and private information

Requirements are made using [23].

4.4.7 Quality Of Service Requirements

According to Enwikipedia.org, Quality of Service can be defined as “the probability of the telecommunication network meeting a given traffic contract, or in many cases is used informally to refer to the probability of a packet succeeding in passing between two points in the network.” In other words the service needs to provide consistency and be predictable in its delivery of packets (data). Depending on the device used, different users will experience the quality of the service in different ways; hence the resources must be allocated for each individual stream [24].

Based on the latest features of the upcoming N92 3G Nokia Mobile phone, the service needs to provide the following Quality of Service levels to be able to offer the best streaming results [25,26,27];

1. Real-time audio/video streaming with a delay <3ms
2. Handover can not disturb the service
3. Video Resolution of the real-time stream and other content must at least be in CIF (352 x 288pixels) alternatively QCIF (176 x 144 pixels) or SubQCIF (128 x 96 pixels), providing 15 images pr second
4. Service content should be based on DVB-H. IPDC IRD B (H.264 AVC 384 kbps@QVGA 15 images pr.sec./ QCIF 30 images pr.sec and AAC+ 48/64 kbps [
5. The system must at least offer 7 channels (User Scenarios in chapter 2).

Type of DVB reciever	H.264/AVC-Level	Video resolution	Max. bitrate	Application
A	1	QCIF (180x144)	128 kbit/s	UMTS Tel.
B	1.2	CIF (360x288)	384 kbit/s	UMTS Tel., PDA
C	2	CIF (360x288)	2 Mbit/s	Pocket reciever
D	3	SDTV (720x576)	10 Mbit/s	TV set
E	4	HDTV (1920x1080)	20 Mbit/s	TV set

Table 2 H.264 resolutions and corresponding bitrates for certain applications [26]

4.4.8 Service Specification Requirements

The service requirements are mostly covered by the different requirements mentioned above; still we can sum up more specifically in case of future edits and validation of the system.

<u>Functional Requirements</u>

- The user must be able to logon and view live streams
- The live stream must be both synchronized video/audio
- The user must have the option to chat with other users
- The user must have the option to view past events
- The user must be able to contact the content provider either by SMS, email or by calling the support team.

<u>System Requirements</u>

- The system must use open standards
- The standard used must be able to handle all types of traffic over 3G

<u>User Requirements</u>

- The use of the operating system for the service should be logical to the user
- The system should be easy to use with few button pushes
- Phone display must be large enough
- Live stream should always be view 8
- Extra information services must be possible to turn off
- Possibility for several camera views
- Possibility to watch more than one event at the same time and swap between audio for each of the events
- Different languages
- Can not turn of commercials (alternatively the user pays less if commercials are turned on?)

<u>Quality of Service Requirements</u>

- The stream can not have a bigger delay than < 3ms.
- Backup stream should be up immediately after a crash.
- Temporary commercial, slideshow of the artist/band while the service is down or similar.

4.4.9 Standards

Today, most customers are being offers some kind of Mobile TV services from their operator based on different standards. With increase in use of Mobile TV, the networks will congest and be forced to take the further step in making deals with broadcasting networks as they then can offer services directly from the ether (already offered in Southeast Asia).

System	ISDB-T	DVB-H	DMBX
Region/Country deployment	Japan	Europe/US	Korea
Codec Video/Audio	MPEG-2 (H.264) MPEG-2 (AAC)	H.264 (expected) MPEG-2 (BC)	H.264 MPEG4 (BASC)
Frequency/Channel Max bps	6 MHz 23 Mbps	8 MHz 31 Mbps	6 MHz 9.2 Mbps
Modulation	OFDM (13-seg/ch)	COFDM	COFDM
Optimized Power Reduction for Handset	Mobile use 1 seg. only	Time Slicing	None

Table 3 Broadcasting Standards

However there is a problem when it comes to a universal technology standard for Mobile TV, as the operators and the terminal device producers use the the standards that fits their purposes the most. Most of Europe has adopted the DVB-H, used by both Nokia and Vodafone.

Others like Ericsson support 3GPP PSS (Packet Switched Streaming – TS 26.234), Samsung uses the DMB and Qualcomm has actually went as far as to developing their own standard called MediaFLO, also used in USA. The rest of the world seems to be following their own lead, for example Japan who use the ISDB-T standard, the Pacific region that use the DMB technology, and USA who used both MediaFLO and DVB-H. Since all these standards are non compatible with each other, making a unified standard becomes very complex.

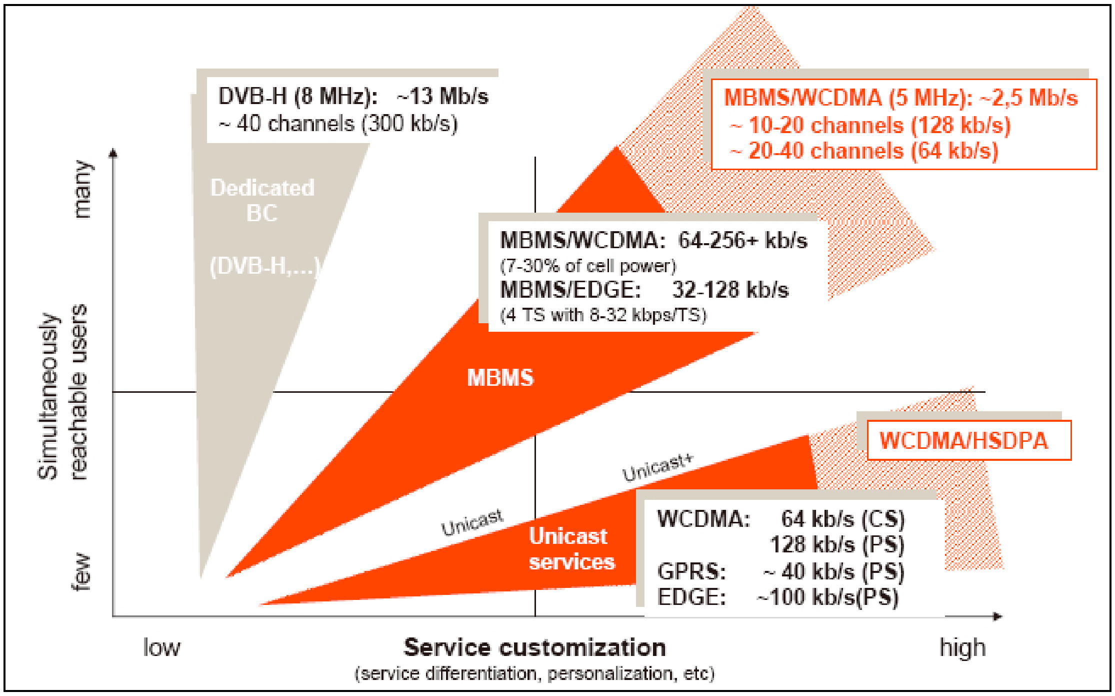


Figure 8 Technology options for Mobile TV [28]

Mass multimedia content can be delivered to mobile phones using cellular point-to-point (PtP) connections thru (GPRS) or fixed Internet, but is far more efficient when the same content is delivered to many recipients in the same cell using point-to-multipoint (PtMP), that also is used in DVB-H [29].

Figure 9 Ptp vs. PtMP

Considering that the main content in this thesis will only be transmitted once to the whole group (live stream events) not each individual user, it's clearly that PtMP is the desired broadcasting type. For this to work properly it is important to customize the capabilities for the whole group not each user separately. Using PtMP also puts constraints on the data rates, minimum capabilities; sharing etc. which leads to guarantee problems when it comes to streaming delivery and service quality. Broadcasting to large terminal groups can also lead to repeated significant delays (up to hours) for perfect delivery to 99.999% of terminals.

The service in this thesis is based on the use of 3G Nokia terminals.

4.5 Mobile Phone Requirements

4.5.1 Design

It is important that the mobile phones that the service is running on include an integrated tiny antenna (<1/10 alpha), work well on limited power supply and also have integrated cellular radio. They should offer strong reception under difficult conditions and handle manmade noise. The terminal should also be used at least 1m above floor or ground level. In today's phones the antennas for signal reception are integrated in the phones as well as the amount of terminals is increasing tremendously, hence the user should be able to get a really good signal almost anywhere.

Terminals must satisfy users in the following ways;

- User-friendly interface at the portal.
- Power saving standby and in-use time.
- Mobile devices should be small and light weighted
- Private and secure protection of the mobile phone.
- The mobile phones must be affordable to everyone;
    - Diversity of mobile phones will give people a choice of portal features in a wide range of prices.
    - The most compact mobile phones will have a reduced power-period, but me more comfortable to carry around and look neater.
    - Mobile phones with lots of features will probably provide a great portal interface but be more complex to use and harder to protect against soft-attack.

4.5.2 Requirements for good streaming handsets

- High-resolution screen, >64K colors
- Loudspeaker and stereo audio output for headphones and 3D glasses
- Enough processing power for decoding
- 3G network interface for ~100 kbps video quality and support for DVB-H and/or MBMS
- Support for multiple APNs and/or multiple contexts (for separating streaming and WAP traffic)
- Open SDKs to enable development of various applications and embedding the player
- Player support for horizontal/landscape full screen viewing modes as well as 3D features
- Support for 3GPP PSS and MPEG-4 standards

4.5.3 Testing of Terminals

Proper testing of a IP Data cast terminal requires a laboratory set-up including (minimum), the following:

- DVB-H transmitter (network).
- IP encapsulator to encapsulate the IP packets for DVB-H transmission.
-
- PC to run a set of IPDC test suites including ESG data and corresponding content items or,
    - A service system used for actual provisioning of IPDC services.

- Testing the payment and purchase of IPDC services will require:
    - A cellular connection with SMS and/or GPRS support.
    - An e-commerce system for recording purchase and payment transactions and for delivering rights objects to the terminal.

Carrying out field tests related to the performance of the IPDC receiver will require also a real scale DVB-H test network [30].

Testing of a 3G streaming requires;

- MPEG-4 or 3GPP converter
- Steaming Server
- UMTS network
- PC to run a set of IP casting test suites including ESG data and corresponding content

- Testing the payment and purchase of the service will require:
    - A cellular connection with SMS and/or GPRS support.
    - An e-commerce system for recording purchase and payment transactions and for delivering rights objects to the terminal.

Simple mpeg or 3gp videos can be tested by using a simple streaming server and opening the "rttp://" stream using WAP technology.

4.6 Network Requirements

Reference [31] was of very much help in writing the Network Requirements

4.6.1 Network 'Must Haves'

- Must satisfy users' requirements mentioned in 4.3.2,
- Capacity must offer high data rates on demand.
- Convergence: common access of seamless applications.
- Content: infotainment.
- Coverage everywhere and anywhere.
- Cost: common infrastructure and platforms to reduce call overheads.
    - Common infrastructures ease coverage restrictions and reduce costs.
    - Common platforms reduce costs and support convergence.
    - Common services make convergence easier and content more widely available.

4.6.2 Streaming Requirements

- Minimum EDGE, WCDMA bearer (GPRS ok for piloting)
- QoS especially in EDGE networks
- Enough capacity (both radio and transmission)
- Traffic differentiation support (for billing, reporting, etc.)
- Scalable streaming server/proxy platform
- Multicast support (DVB-H and/or MBMS)

5 DRM

Digital rights management (DRM) protects the rights of all suppliers of digital content and offers them safe distributing and selling of their content. DRM makes sure the content owners will be paid for the use of their content, offers the operators to be able to bill fairly for content and addresses the whole issue of how to control content distribution [22, 32].

The Open Mobile Alliance (OMA) has developed an international standard of the OMA Download, which includes;

1. Applying Digital Rights Management (DRM) to content and its distribution
2. Enabling controlled (i.e. reliable) delivery of generic content objects.

DRM prevents illegal distribution of media content and provides new business models such as preview, backup, export super-distribution, rights updates and more. A user can for example download an event or concert by his favorite artist for a certain period like a day or a week, and be given the option to buy refreshed rights after his original rights have expired. For content providers, DRM will provide new opportunities and distribution channels, which encourages developers to produce more applications and high quality content, knowing for sure that they will be paid properly for their work.

DRMs role is to govern and control the use of mobile-centric content types in distribution of content. When a user buys content, she or he may agree to certain constraints – like for example choosing between a free preview version or a full version of the content at cost. Alternatively she or he may agree to pay a monthly fee for content usage. DRM allows this choice to be translated into permissions and constraints, which are then enforced when the user accesses the content.
More details about the DRM can be found in Appendix.

6 Mobile TV

Ever since 2005, the interest in Mobile TV has been growing rapidly. Many operators are already offering Mobile TV services on a commercial basis as they see its big advantage of reaching the mass-market easy, especially considering there are more than two billion mobile phone subscribers worldwide [33]. The have turned out to be more interested in the Mobile TV services than the operators anticipated at first, forcing them to advance their services as well as demanding more from the mobile devises used for them.

In Europe, Mobile TV is offered by operators such as Netcom, O2, Telenor, Orange, T-Mobile and Vodafone through 3G and DAB. Though the services on the Norwegian market are limited at the moment (Telenor offers voting system on Voice TV for tracks that are sent), other countries like France can enjoy more than 50 TV channels on their UMTS network and Vodafone offers currently 24 different channels in 12 countries, including an entertainment pack called Vodafone Live (no live stream content is offered of any events though, only download of past events and snapshots / previews from the past events they have held themselves). The Mobile TV video stream is transmitted over wireless networks using unicasting. This means the video stream is sent to every user in a separate stream over the network. Multicasting is currently under development and with the hopefully future worldwide DVB-H broadcasting standard we will all be able to enjoy Mobile TV.

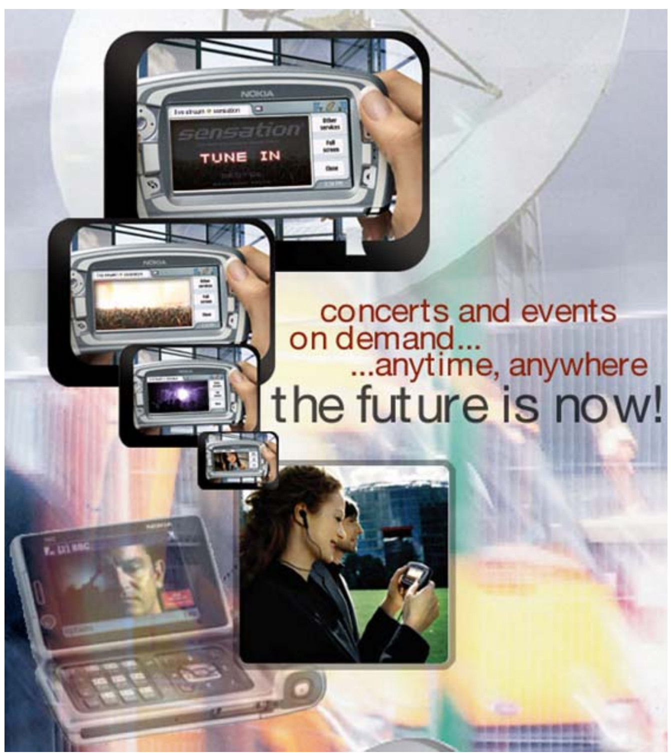


Figure 10 Future is now!

6.1 Streaming Technologies

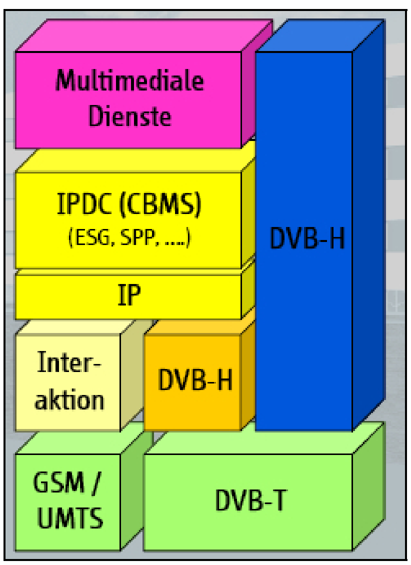
6.1.2 IP Datacasting

The IP Datacast technology was developed to enable efficient and cost effective broadcasting of digital content like live streaming and Mobile TV to mobile devices such as mobile phones, PDA's and over IP [35].

	Digital TV over DVB-T	IP Datacasting over DVB-H
Display	large TV screen	Small, mobile phone screen
Antenna	Large, roof top	Small, inbuilt
Power Supply	Fixed, continuous	Battery power, limited
Receivers	Fixed	Mobile

Table 4 IP Datacasting over DVB-T and DVB-H

IP Datacasting uses the Internet Protocol (IP) format distributing digital content on the Internet in the form of IP data packets. By combining IP with DVB-H (Digital Video Broadcast - Handheld) putting IP protocol on top of the DVB-H transmission; the system can be easily adapted to new future services.



IPDC is a combination of the DVB-H standard (6.2.2.4) and IP technology and is optimized for distributing and receiving of mobile content. The first digital convergence handsets are already on their way and with IP datacast, all content will be delivered in the form of data packets (same format as used for digital content distribution on the internet) which will automatically make the digital content available for broadcasting. This concerns all types of content that will become digitalized, even traditional TV content [4].

IP Datacast easily adapts to small screen sizes such as the ones on mobile phones. Since a normal TV channel requires only 128-384kbs per 'channel' per TV program to offer high quality video to handheld devise with small screens, IP Datacast increases the broadcasting efficiency making it possible to stream between 50 and 80 different TV programs over one network.

Figure 11 IP Services over DVB-H [35]

IPDC offers the availability of an interactive return channel over the cellular network (6.2.1) which can be used for content purchasing, viewing information on-line, voting etc. Broadcasting is done through a broadcasting channel (6.2.2)

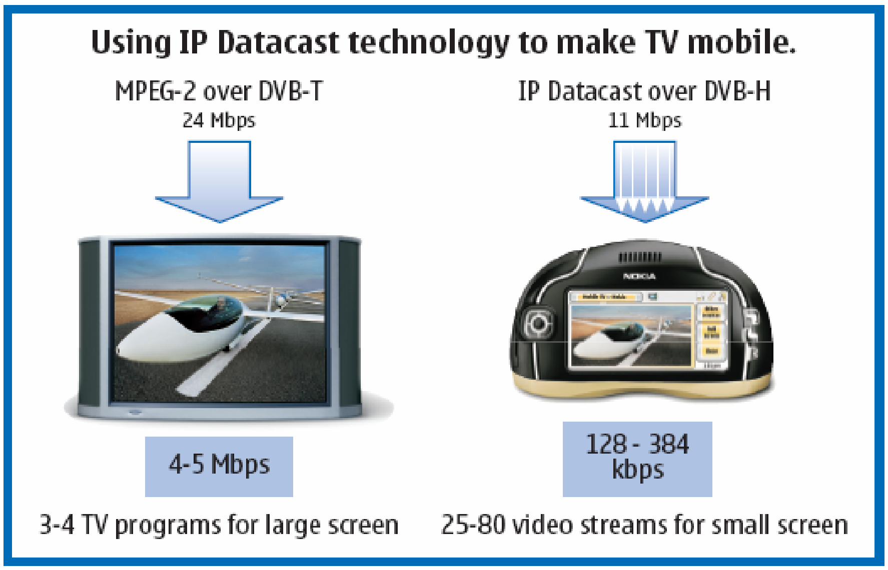


Figure 12 IP Datacast Technology Optimizes Mobile Broadcast [5]

IP Datacasting networks support indoor coverage for devices with small, in-built antennas, and enables a longer battery lifetime for mobile devices through bursting content in high-speed. Between bursts the receiver shuts down to save power, a "time-slicing" technology which is supported by DVB-H [36].

The key elements in the convergence of broadcast and mobile systems can be found in the Nokia IP Datacasting whitepaper [5] and are as followed (listed directly from the paper):

The future of broadcast is digital, digital broadcasting will enable new mobile distribution channels and methods for media. The broadcasting paradigm will come to the mobile environment with the handset as the convergence point. IP Datacasting technology can provide the needed capacity, broadcast capability, and consumer experience to make Mobile phone TV and other mobile broadcast multimedia commercially successful. Suitable digital content for mobile broadcast already exists, it can be billed for, and there is consumer demand for it.

IP Datacast also offers End-To-End solution [5] for Mobile phone TV where the IP Datacast End-To-End solution encapsulates the video and audio stream using an IP Encapsulator, which is the gateway that enables broadcast delivery of IP content. The stream is then formatted and transmitted using multicasting.

The Service System that is in charge of the broadcasts as well as the end-to-end protection generates the Electronic Service Guide (6.1.3) which generates and broadcasts service meta data so that available services can view service descriptions as well as buy new services. The solution also offers service protection through end-to-end control of encryption and decryption through the E-Commerce System which takes care of DRM for viewing and purchase of content as well as the production of charging data for purchase transactions for billing purposes.

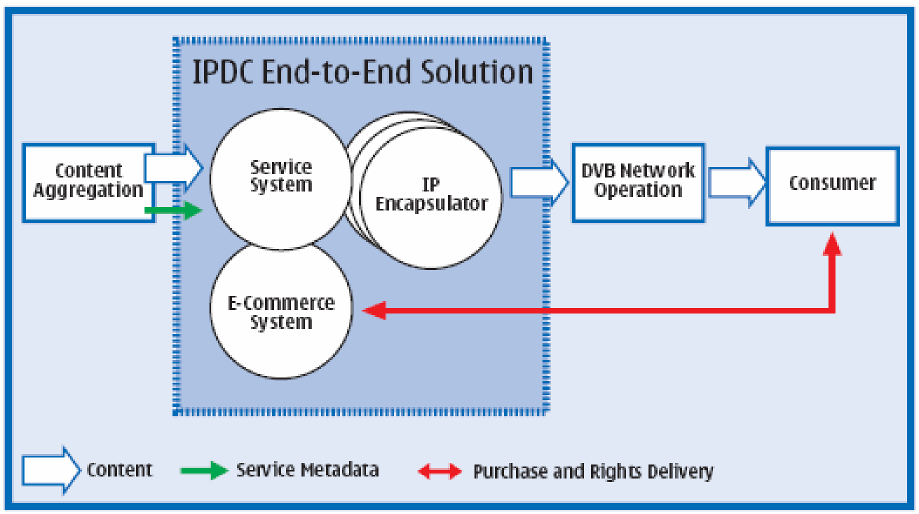


Figure 13 IPCD End-to-End Solution [5]

IP Datacast is based on subscription-based content delivery with encryption due to which the data service providing the content should be implemented by the content providers (media companies, TV broadcasters and mobile phone and network operators). This implementation requires that handsets, network infrastructures as well as service management systems are capable of IP Datacast.

Roles of the participating actors in the Mobile TV IP Datacast system can be summaries as follows:

Content Providers (CP) will offer the same offers to the customers but will now be able to expand the content provision through IP Datacast;

- CP can Reuse existing or re-purpose content over the new distribution channel
- CP can now reach out to a completely new mobile audience
- And reach out to the existing customers in new ways
- CP can offer small additional costs for the use of the distribution channel

Content Broadcasters (CB) will offers their existing services but will additionally

- Reach out to customers that are on the move through IP Datacasting.
- CB will purchase content from CP
- CB will sell content through different channels based on the distribution rights
- Content will be delivered to customers based on their subscriptions

Datacast Service Provider (DSP)

- DSP operates and controls the IP Datacast distribution capacity available on the digital broadcasting networks
- DSP can sell the distribution capacity straight to CB
- Various deals between DSP and CB can be made regarding to the content distribution, such as;
    - How the content will be sold and at what capacity
    - For which periods of time the content will be delivered
    - The agreements also include things like necessary content protection, billing services, e-commerce and refund services.

Electronic Service Guide (ESG)

- A guide containing information about the available services and when they are offered. ESG is broadcasted parallel with the content
- ESG is covered in chapter 6.1.3
- DSP also makes sure that the broadcasted content is protected from illegal viewing and tells the consumers where to purchase rights for the content they are interested in.

Datacast Network Operator (DNO) owns and operates the digital broadcast infrastructure.

- DNO can hold frequency licenses
- DNO operates the transmitters and the site connections
- DNO sells broadcast broadcast capacity and coverage

Cellular Network Operators (CNO) own and operate the cellular network infrastructure.

- CNO holds frequency licenses
- CNO sells network capacity to service providers
- CNO can act as distributors or retailers of IP Datacast services for their own customers who provide e-commerce
- CNO offers two-way access for content purchase as well as use of other interactive services related to the IPDS
    - Advanced and cost-effective billing
    - Customer relationship management
    - Customer support
    - Electronic shop (e-commerce)

The figure below illustrates the IP Datacast system; IP Multicasting).

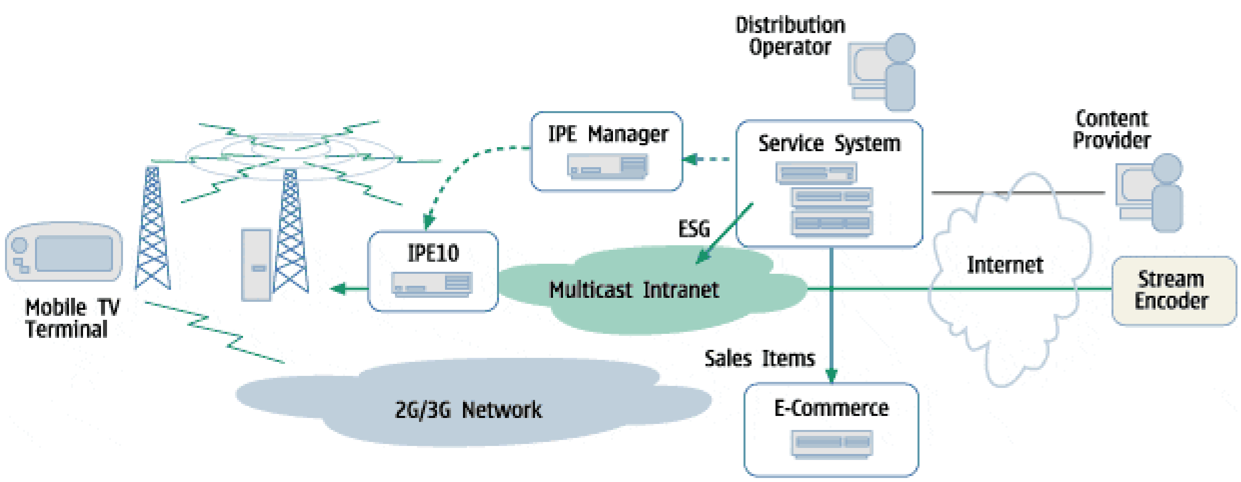


Figure 14 IP Datacast system; IP Multicasting [36]

In short, IP Datacast can be summed up to be a very cost-effective way of delivering multimedia content through mass broadcasting. IP Datacast offers safe, branded and high value content delivery through a digital channel and is a technology with inherent abilities for charging for content purchase as well as services. IP Datacast compatible Mobile Phones will free the consumers from time and space, allowing them to view TV and explore new types of content and services. IP Datacast is the future and the best technology for mobile communication and Mobile TV today [36].

6.1.3 Electronic Service Guide

The electronic Service Guide (ESG) provides the consumer and client applications on mobile devices with information about the services offered by the digital mobile broadcast services which the consumers may buy. In Nokia Mobile Broadcast solution, ESG interacts with the middleware in mobile phones enabling service look-up from the DVB-H stream as well as playback with the correct client software and codec's. ESG prompts the user to make purchases as well as making sure the user is aware of all the new and existing services available. The user can also interact with the broadcaster services through ESG as well as receiving multiple video and audio streams and dynamic links during live-streaming making the Mobile TV experience more interesting.

The ESG protocols used in the Nokia Mobile Broadcast solution 3.0 are defined in Open Air Interface 1.0 Specification, last updated march 2006 []37[. The ESG solution in Nokia IPDC is to be found the ESG short specification [38].

ESG is also used by NSD in the mVideoGuard solution [39].

6.1.4 IP Unicast & Multicast

The unicast technique offers asynchronous person-to-person and client-server communication for delivering of on-demand content (e.g. streaming & download). Unicast can be separated in
real-time and non-real-time and enables a feedback channel for interaction between users. Vodafone Live! Is a good example for real-time unicasting, where the TV Content is 'steamed' live over the 3G network by the provider. Non-real-time unicasting such as used by Verizons [40] offers the customers non-real-time access to media such as news, pictures, music. However, unicast transmission consumes more network resources the more it is used and the amount of consumers for the services are growing rapidly. Despite the 3G network being able to handle the traffic at the moment, increase in Mobile TV usage and viewers will definitely cause problems in the future, especially since the current speed of the 3G networks is only between 150 and 220Kbps [41]. Though users in full working networks can expect performance up to 384 kbit/s, which in the future will certainly not be enough. Faster transmissions can be realized by using High Speed Downlink Packet Access (HSDPA), that in theory can offer transmission speed of 14,4 Mbps, however this may still not be good enough in areas where the number of mobile TV viewers are big. One of its biggest weaknesses is the encounter of a bottleneck if same information is delivered to many end-points at the same time.

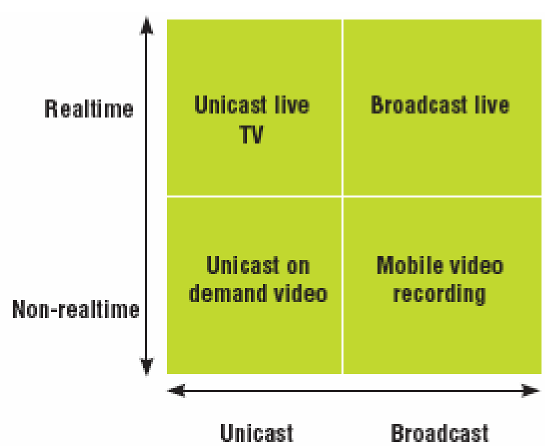


Figure 15 IP Unicast and Multicast

An alternative technique for video streaming over mobile networks is Multicasting, also known as Multimedia Broadcast and Multimedia Services (MBMS).
Multicasting offers one-to-many communication, delivering the same information to many users simultaneously but does not offer any feedback channel for interaction. Since multicasting addresses groups instead of single hosts it is the most efficient transmissions of an IP datagram to a set of zero or more hosts identified by a single IP

destination address. Only one packet is send by the server to a group of receivers replacing only the necessary packets [43].

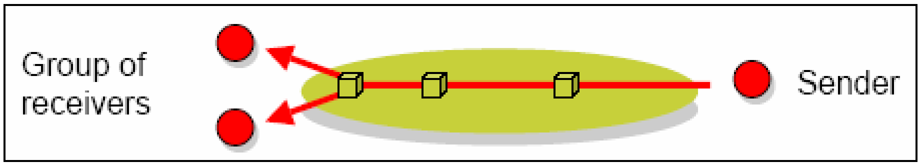


Figure 16 IP Multicast [42]

MBMS is a technology developed as part of the 3GPP standards (Release 6) [43]. MBMS is only suitable for a small quantity of video streaming over UMTS, even though very efficient, the broadcast is only enabled over one to three TV channels. As we already see, commercial products based on this standard are already under development and should be expected sometime during 2007 [44]. Mobile Video Recording offers non-real-time video broadcasting to users anytime anywhere.

Originally the 3G networks were never intended for real-time TV broadcasting since multicasting services over 3G is an inefficient use of limited network capacity. Dr Markus Lindqvist, director of rich media services at Nokia says; “For a three-minute video clip to be delivered to 100,000 users over a 3G network, it could take 1.5-2 days. With a broadcast network it could take a matter of seconds.” The same transmission using broadcasting networks would only take a few seconds because the mobile phone picks up the signal directly from the ether. All around the world, there are now carried out trails with complementary broadcasting networks, to solve the capacity problems in mobile networks.

6.1.5 High Speed Downlink Packet Access (HSDPA)

HSDPA is based on the WCDMA evolution and is a part of the 3GPP specification [45]. HDSPA offers high-speed downlink data capabilities to UMTS networks, allowing high-quality video and audio streaming, quick browsing through heavy-graphic websites, as well as the running of advanced applications on mobile handsets. HDSPA enhances data rates and spectrum efficiency from the radio access network to mobile terminals using Adaptive Modulation and Coding (AMC) as well as fast scheduling and retransmission (based on fast Hybrid Automatic Response request (HARQ) techniques. HDSPA also introduces a new transport channel type namely High Speed Downlink Shared Channel (HS.DSCH) that handles packet data burst and takes advantage of valuable radio frequencies. The transport channel shares multiple access codes, transmission power and the use of infrastructure hardware between users. The downlink channel enables operators to provide new, better and faster services such as on-demand video/audio streaming, high-resolution interactive gaming, music videos, “push-to-watch” services and access to huge email attachments at speeds nearly three times faster than today’s commercial 3G UMTS networks. HSDPA is an evolution of existing WCDMA networks that provides a powerful boost in data rates and an increase in overall

network capacity. The latest HDSPA release "boosts downlink speeds from the current end-user rate of 384 kbps (up to 2 Mbps according to standards) to a maximum value according to standards of 14.4 Mbps. Real life end-user speeds will be in the range of 2 to 3 Mbps." This indeed is fast! [46]The new release also provides for two delivery systems, where one has a maximum downlink speed of 3.6 Mb/s while the other has a maximum downlink speed of 14.4 Mbps. The first one is mandatory while the second is optional.
(Alcatel [47] for example offers both modulations, to provide maximum speeds for everybody). With the increase in data capacity, HSDPA-enhanced UMTS networks will increase 2.5 to 3.5 times of what they have been so far and hence offer extremely cheap network costs for data services for mobile operators. Since HSDPA offers the unique combination of spectral efficiency, high speed and low latency, it is the perfect mobile broadband for the mass-market [48].

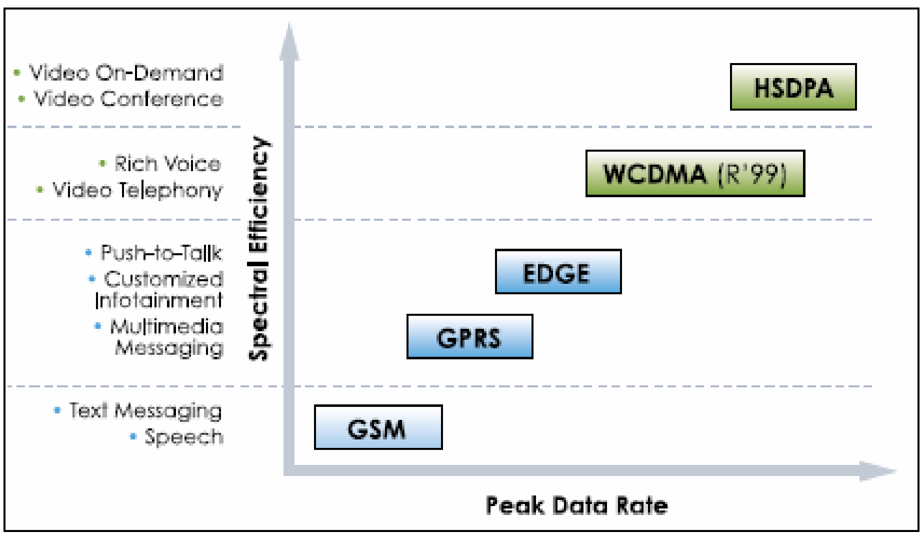


Figure 17 HSDPA enables new and more efficient applications [48]

HDSPA enables new and more efficient applications

Performance improvements in HDSPA are predicted in the future and involve equalization and advanced Multiple-Input-Output (MIMO) techniques.

6.2 Downstream Technology

As mentioned briefly at the end of chapter 4, there are many ways to transmit TV content to a mobile phone, and like all new technologies, there are several different standards for Mobile TV around the world such as GPRS, UMTS, MediaFLO, DAB/DMB, MBMS, 3GPP, DVB-T and DVB-H. Of the ones mentioned here, the most used open standards are DMB, DVB-H and ISDB-T and 3G. MediaFLO™ is not an open standard and used in US only while ISDB-T most likely will never become available in Europe [49].

	DVB-H	MediaFLO™	ISDB-T	DMB	3G
Standard	Open	Proprietary	Open	Open	Open
Regions	US, Europe, parts of Asia	US	Japan	Korea, expanding to other countries	Worldwide
Air Interface	OFDM	OFDM	OFDM (sub-banded)	OFDM	
Service Availability	Mid 2005, open US spectrum nationwide today	2006 (locally through analog TV channels)	Early 2006	Today	Today expanding rollout
Handset Availability	Today from several OEMs	2006	2006	Today from several OEMs	Today from serval vendors

Table 5 The spread and availability of downstream technologies [50]

6.2.1 Mobile Technologies

	Telecom / Cellular			Broadcast Systems	
	GSM	**GPRS**	**UMTS**	**DAB/DMB**	**DVB-T / DVB-H**
Spectrum Bands	900 MHz - 1.8 GHz	900 MHz - 1.8 GHz	2-2.5-3 GHz	1.5 GHz VHF L-Band	VHF, UHF
Regulation	Telecom Licensed			Broadcast Licensed	
Typical Regulation	14.4 kbps	30kbps	30-300kbps	1Mbit/s	DVB-H 10Mbit/s DVB-T 5-25Mbit/s
Transfer Mode	Circuit	Packet	Circuit/Packet	Broadcast	
Primary Applications	Voice	Data	Voice & Data	Audio, Still Image, Push Image, Traffic Information	Audio & Video, Push data
Mobility Support	High			High	Medium to High
Coverage	Wide		Local to Wide	Wide	
Deployment Costs	Network Exists	Incremental	High	Basic Network exists in many countries	

Table 6 Comparison of technical features between broadcast and mobile telecommunication systems [51]

6.2.1.1 GSM

(Global System for Mobile communication)
A digital mobile and most widely used wireless telephone technology in the world along with TDMA and CDMA. Used in Europe as well as other parts of the world. GSM uses a variation of time division multiple access (TDMA) and digitizes, compresses and assigns the data to separate timeslots before it sends it down a channel with two other streams of user data (also in separate slots). Along with together technologies, GSM is part of the wireless mobile telecommunication system that includes High-Speed Circuit-Switched Data (HCSD), General Packet Radio System (GPRS), Enhanced Data GSM Environment (EDGE), and Universal Mobile Telecommunications Service (UMTS).

6.2.1.2 GPRS

(General Packet Radio Services also known as “2G”)
GPRS is a mobile data service based on the GSM system. GPRS can also be described as the technology between 2G and 3G and provides moderate speed data transfer, by using unused TDMA channels in the GSM network. GPRS is integrated into GSM standards releases starting with Release 97 and onwards. First it was standardized by ETSI but now that effort has been handed onto the 3GPP.

6.2.1.3 UMTS

Universal Mobile Telecommunications System is a 3G technology which uses W-CDMA and 3GPP standards for streaming TV content to Mobile phones [52]. UMTS offers mobile operators significant capacity and broadband capabilities to support mass data and voice transmissions to customers, providing the maximum 2Mbps throughput in urban areas, while offering average data rates of 220-320 kbps and very low latency elsewhere. UMTS also provides higher data rates at lower incremental cost than GPRS by taking advantage of radio spectrum in bands identified by the ITU for 3G IMT-2000 mobile services and subsequently licensed to operators. The system is unicast and expensive to mobile phone users. More information about UMTS can be found in UMTSWORLD [53]

6.2.2 Broadcast Technology

6.2.2.1 MBMS

Multimedia Broadcast Multicast Service (MBMS) is an IP packet based broadcasting service that can be offered via existing GSM and UMTS mobile networks.
The infrastructure gives the possibility to use an uplink channel for interaction between the service and the user. This not a straightforward issue in usual broadcast networks, since for example conventional digital TV is only a unicast one-way system. MBMS offers both broadcasting (similar to digital TV) and multicasting including services subscription, join and leave [54, 29].

MBMS has been specified by the 3GPP (Third Generation Partnership Project) in the UMTS Release 6 as a way to transfer multimedia content to multiple users and uses the same radiostream in bort up- and downstream directions as well as offers immediate respose to interactive requests. Unfortunatley use of same radiostream causes lower bitrates that can affect either the frame rate or the quality per frame.

According to Wikipedia, MBMS is believed to currently be under active evaluation by Mobile Network Operators as a means of delivering Mobile Television to the mass market and the first practical network implementations are to be expected by the end of 2007 along with the first functional mobile terminals supporting MBMS.
Competing technologies include DVB-H, DMB and DAB.

6.2.2.2 DAB/DMB

DAB is a terrestrial digital broadcasting standard by IUT (International Union for Tele-communications) and has currently over 300 million users around the globe receiving over 600 different DAB services every day. DAB uses Band III and L-Band frequencies that allow a good quality transmission and eliminates interference as well as the issue with "multipath" while being in a car. DAB also enables distortion-free reception almost at any place and surroundings no matter where. Since it originally was created for mobile transmissions, it can be used for file and multimedia transmissions to portable and fixed receivers with a simple, non-directional antenna with useful bit-rate capacity of approximately 1.5Mbit/s.

According to VDL [55]"DAB programs are broadcast inside a multiplex which is composed of six to ten radio stations on a single frequency", allowing DAB to broadcast more programs than FM. DAB can also transmit PAD (Program Associated Data) or NPAD (Non Program Associated Data) such as text, pictures, data and videos. The last is referred to as DMB (Digital Multimedia Broadcasting).

DAB is however not a good choice for video transmissions due to the problems related in its hardware (up to 1.3 seconds transmission delay) aswell as the limited spectral efficiency and channel coding. The channel bandwidth of DAB is 1.5 MHz. This fits, for example, three times within a UMTS channel (5 MHz) [56, 57].
and can deliver about 3 video services in one DAB multiplex with about 1,2Mbit/s. DAB is currently available in Europe by VHF-or in L-band transmissions.

DMB (Digital Media Broadcasting) is a technique for transmission of packets of binary data (any type including mono and stereo audio, video or WebPages) over radio waves. The reason for converting to digital systems was to enable higher fidelity, greater noise immunity, mobile services, and new services, althoughgh in practice the audio quality is a lot worse than on FM.

DMB is based on the established Eureka 147 Digital Audio Broadcast (DAB) standard and is essentially DAB with additional error correction. DMB is developed in the Asia Pacific region and offered in Korea. Today DAB has been implemented partially or totally in Scandinavia and UK among other countries.

6.2.2.3 DVB-T

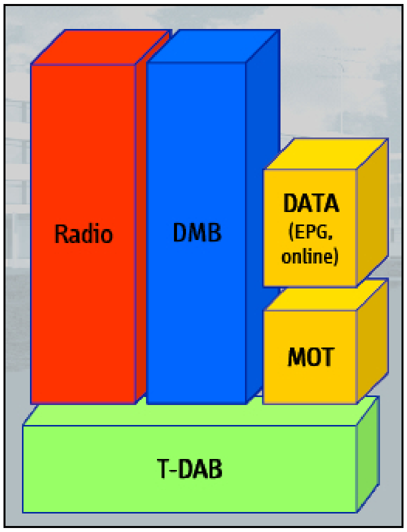


Figure 18 DMB:Digital Multimedia Broadcast [35]

Digital Video Broadcasting – Terrestrial (DVB-T), is the current standard in digital video broadcasting and was designed for TV reception and mainly targets stationary and in-car receivers. Unfortunately DVB-T is not adapted for receiving on handheld terminals due to power consumption issues, lack of processing power on such terminals as well as poor indoor coverage (use of rooftop antenna). However, as antenna technology improved, DVB-T mobile services became feasible, leading to extensive commercial trials making it possible to receive Digital TV reception on the move.

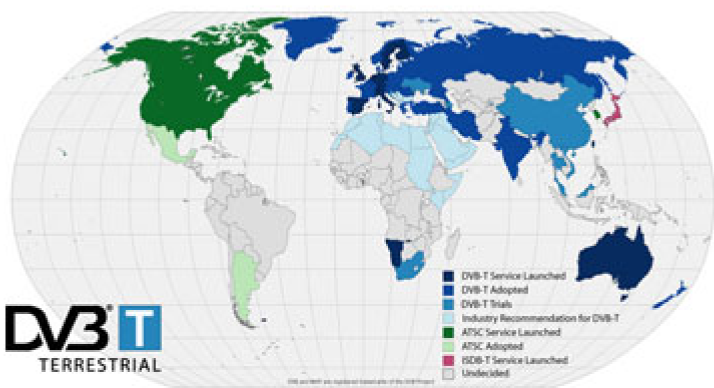


Figure 19 DVB Standard World overview [58]

Due to the low battery life in today's handheld devices, there is not enough power to make DVB-T reception a viable option for consumers, therefore a new solution had to be developed; DVB-H.

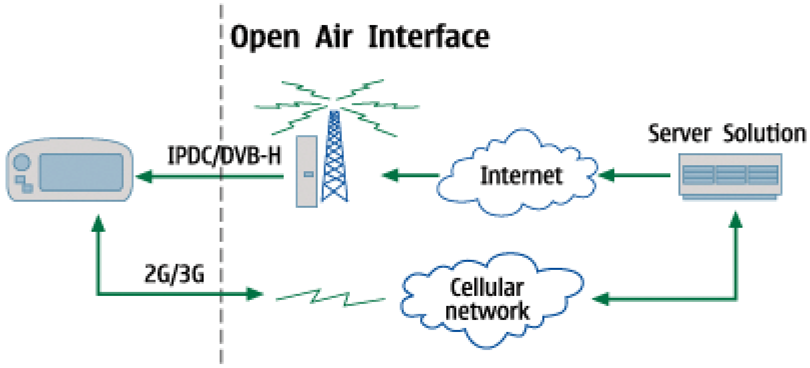


Figure 20 Nokia Open Air Interface [59]

6.2.2.4 DVB-H

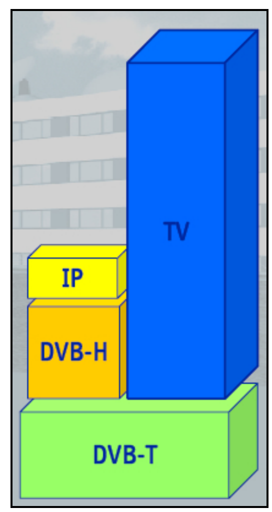


Figure 21 DVB-H [35]

Digital Video Broadcast for Handheld Devices is an ETSI (European Telecommunications Institute) approved standard for handheld devices. DVB-H benefits from existing DVB-T infrastructure components, which reduces initial investments and battery life time for handheld devices up to 90% due to time-slicing technology DVB-H provides the best user experience in the mobile environment, offering the user an energy-saving handset that is only 'on' 10-25% of the time, program guide, soft loss-free handover and in-building coverage [59].

DVB-H also offers an excellent-quality both picture and audio and the quality of the stream can easily be adjusted to the content and in such manner enable optimization of audiovisual quality vs. number of content channels for maximum revenue gains. DVB-H offers maximum compatibility with DVB-T

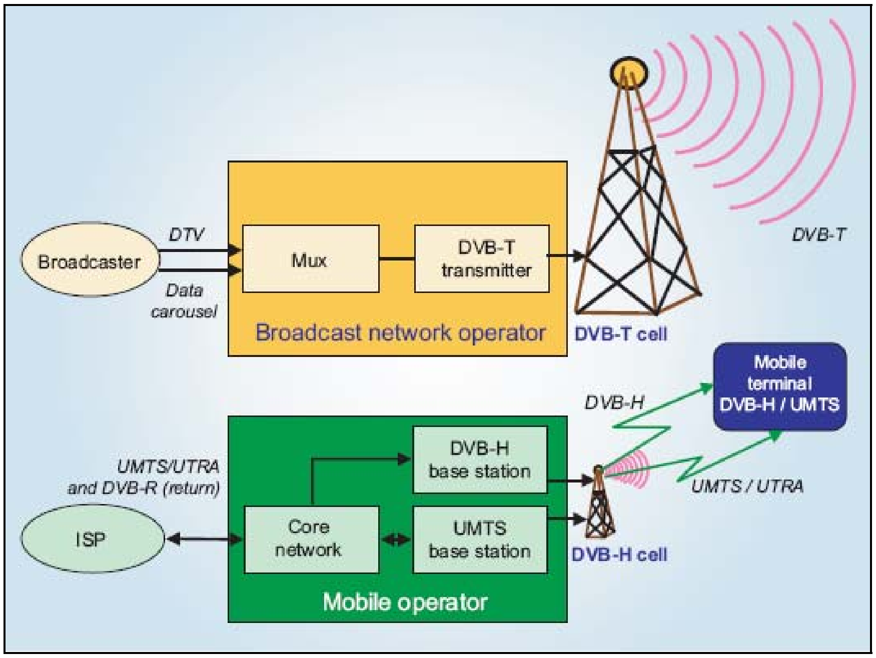


Figure 22 Mobile Broadcasting [70]

Depending on the configuration, DBV-H can offer up to 55 channels at 256 kbits/s scalability and is considered to provide the best user experience in the mobile environment. With excellent picture and reduced battery consumption DVB-H is the best mobile broadcast delivery system currently available. DVB-H is also supported by publicly available interface specifications and include end-to-end control of stream encryption, decryption key generation as well as delivery of keys to consumers in a billing-integrated way [61].

DVB-H also offers interactivity through allowing consumers to interact seamlessly upstream with the server, generating additional revenue streams enhancing the technology value for the consumers.

Worlds first DVB-H mobile phone was launched by LG Electronics [62] on April 28th 2006. Hutchison Italy, the country's largest 3G telecommunications provider has already bought the exclusive rights to the World Cup 2006 which started June 6th. And with that, the first DVB-H broadcasts have already started…

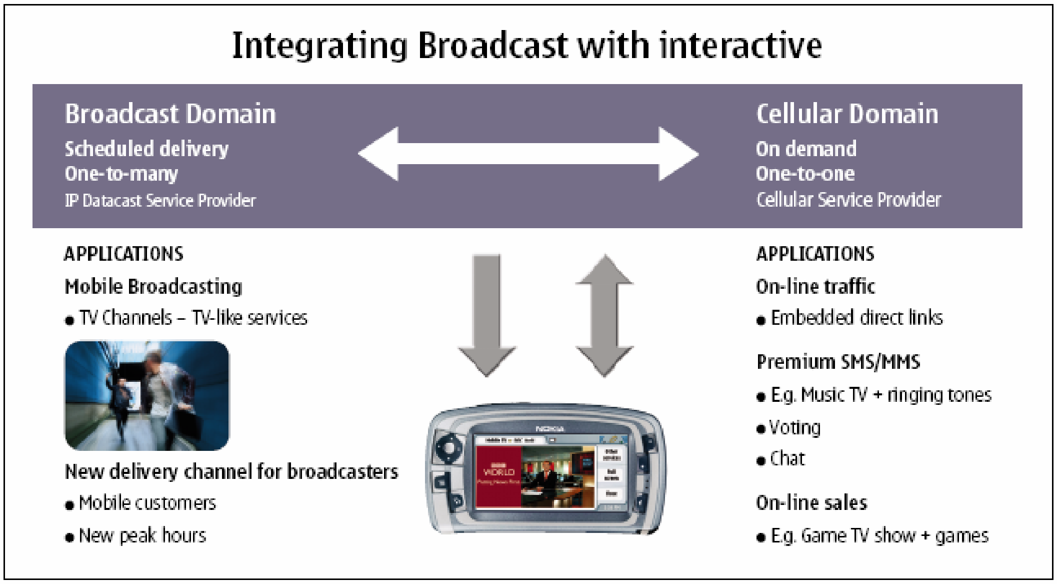


Figure 23 Integrating Broadcast with interactive [63]

6.2.3 Conclusion of Downstream Technolgies

Considering the future developments in today's broadcasting technology and based on the information about standards mentioned earlier, DVB-H is definitely the best downstream technology for the multimedia service in this Thesis due to the following features:

1. The Standard is DVB-T Compatible:
    a. DVB-H can co-exist with DVB-T. For example, an operator could choose to run 2 DVB-T services alongside a DVB-H broadcast within one DVB-T multiplex.
    b. DVB-H can be used in 6, 7, and 8 MHz channels, but also adds the non-broadcast use of 5 MHz
2. Designed to stream both audio and video content and is already in use (Nokia among others)
3. Mobile phones that the desired service is to run on, already exist
4. Downstream channel can be based on IP, using IPDC offering capacity over 300Kbps
5. The standard offers Time-Slicing:
    a. DVB-H transmits chunks of data in bursts, allowing the receiver to be "switched off" in inactive periods.
        i. The result is power savings of up to 90% - and the same inactive receiver could be used to monitor neighboring cells for seamless handovers

6. The standard offers 4K-Mode:
    a. DVB-H offers a good compromise between high-speed, small-area 2K SFNs and the slower, larger-area 8K mode which provides great flexibility to network design.
7. The standard offers Forward Error Correction:

8. Nokia uses the DVB-H standard.

However not all countries support or even consider yet to implement the DVB-H standard. DAB is one of the broadcasting technologies already implemented for example in Scandinavia and UK. 3G can be used for both up and downstream and is currently available all over the world. Since the service needs to be out on the market as fast as possible, 3G is chosen as the current downstream technology which in time will be changed to DAB (already available in Scandinavia and UK according to Kristen Rekdal, NTNU). In the future the service should be able to be broadcasted using all 3 technologies; 3G,DAB and in the future; DVB-H.

6.3 Upstream Technologies

There are many ways a user can interact with a content provider (SMS/MMS, WLAN, WiMAX, GSM, EDGE and UMTS) though the cellular technology is only choice for a return channel as the others have either too short range or are not available inn all global areas [64, 65].

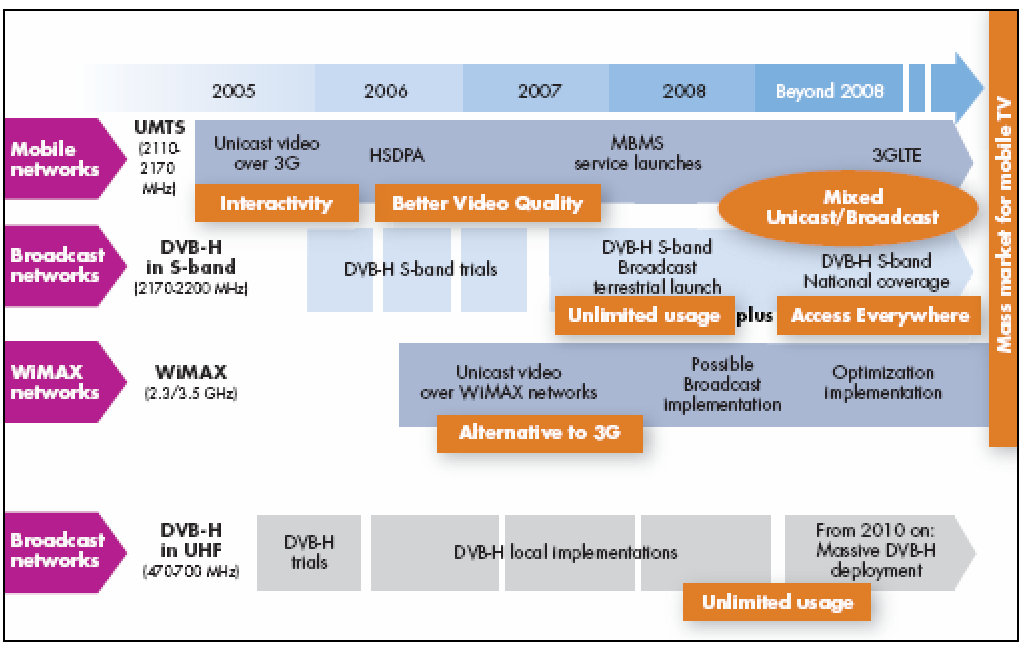


Figure 24 Mobile TV timeline [65]

6.3.1 GSM, EDGE & 3G

Third Generation Mobile Technology (3G) is an ITU specification of mobile communications technology. 3G promises increased bandwidth, up to 384 Kbps for handheld devices regardless whereabouts, 128 Kbps in a car, and 2 Mbps in fixed applications. 3G works over wireless air interfaces such as GSM, TDMA and CDMA. EDGE also called 2,5G was developed specifically to meet the bandwidth needs of 3G. The current service platform will be upgraded to deliver all types of content (live TV, VoD, podcast, etc.), using unicast (2G, 3G, WiMAX, etc.) or broadcast (DVB-H based) modes, depending on the customer location and the requested service or channel.

6.3.2 WiMAX

The IEEE 802.16e, or Mobile WirelessMAN (Mobile WiMax) standard got approved in December 2005 and is a standards-based technology enabling the delivery of last mile wireless broadband access. It provides fixed (not possible to use the service from more than one location), nomadic, portable (basic mobility withought soft handoff) and, eventually, mobile wireless broadband connectivity without the need for direct line-of-sight with a base station.

The WiMAx offers only Voice over IP (VoIP) and is a viable choice for later DVB-H systems if the service will become available all around the globe. Depending on the coverage, there may be need of 3 radio interfaces to support WiMAX, voice and TV content. WiMAX systems are expected to deliver capacity between 15Mbps and up to 40 Mbps per channel, for fixed and portable access applications within a typical cell radius of three to ten kilometres which is enough bandwidth to support hundreds of businesses with T-1 speed connectivity and thousands of residences with DSL speed connectivity at the same time.

WiMax Forum is supporting two 802.16 technologies – OFDM 256 and scalable OFDMA. According to Dr. Jonathan Labs; “What WiMax has adopted for fixed is the OFDM 256 mode – and for mobility, they’re pushing the scalable OFDMA mode,” he says. “And they are incompatible technologies, as they exist today.”

There are however differences between the technologies when it comes to how they deal with the channel characteristics as well as compatibility problems between fixed and mobile WiMAX.
Labs says that “In a mobile environment, your channels are going to be rapidly varying, and they’ve designed the PHY around that fact. And the more mobile you get, the more challenging the channel is to deal with. The idea in general is to try and give the service providers a clear migration path from fixed to portable, and then to mobile.” Labs continues; “So we are using the OFDM 256, and adding the features that will allow at least portable operation,” he says. “Portable is mobile, but maybe not as fast – it’s a limited mobility.”

The full mobile WiMax is not expected until 2007 or 2008 and the portable solution is suppose to be build upon the fixed standard and provide limited mobility in the interim, while requiring only a basic update to systems that are already deployed.
It is expected that WiMAX technology will be incorporated in notebook computers and PDAs by 2007, allowing for urban areas and cities to become “metro zones” for portable outdoor broadband wireless access.

The first WiMAX products became available in January 2006 and consist of the three base stations (Aperto Networks' PacketMAX 5000, Redline Communications' RedMAX AN-100U and SEQUANS Communications' SQN2010 SoC) and one subscriber station: Wavesat's miniMAX customer premise equipment (CPE).

In May 2006 the first agreements of providing wireless solutions for cities and their public safety agencies were made by Alvarion [68] and IBM. The two companies intend to offer a new approach to delivering wireless networks that support data, voice and video for both fixed and mobile applications – mobile, in this case, generally meaning inside police cars and other vehicles [66,67].

6.3.2 The key to success

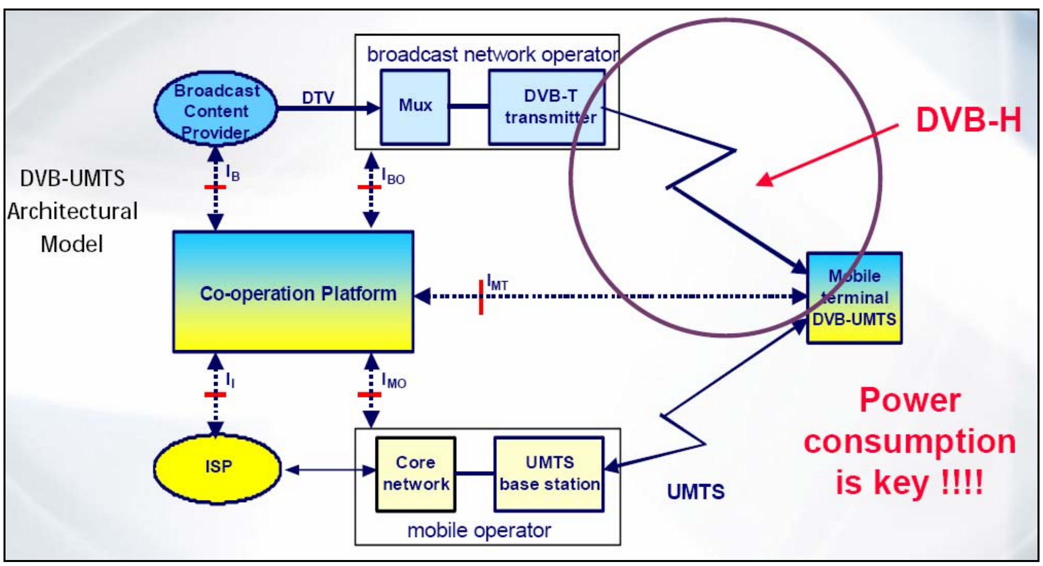


Figure 25 Role of DVB-H in Mobile TV [69]

,

The ultimate key to a successful broadcasting of the multimedia service in this theses is the cooperation between the UMTS and DVB-H interaction channel between the platforms. By sending the content through the UMTS network or the private section of the MPEG packets we can achieve interactivity between the standards.

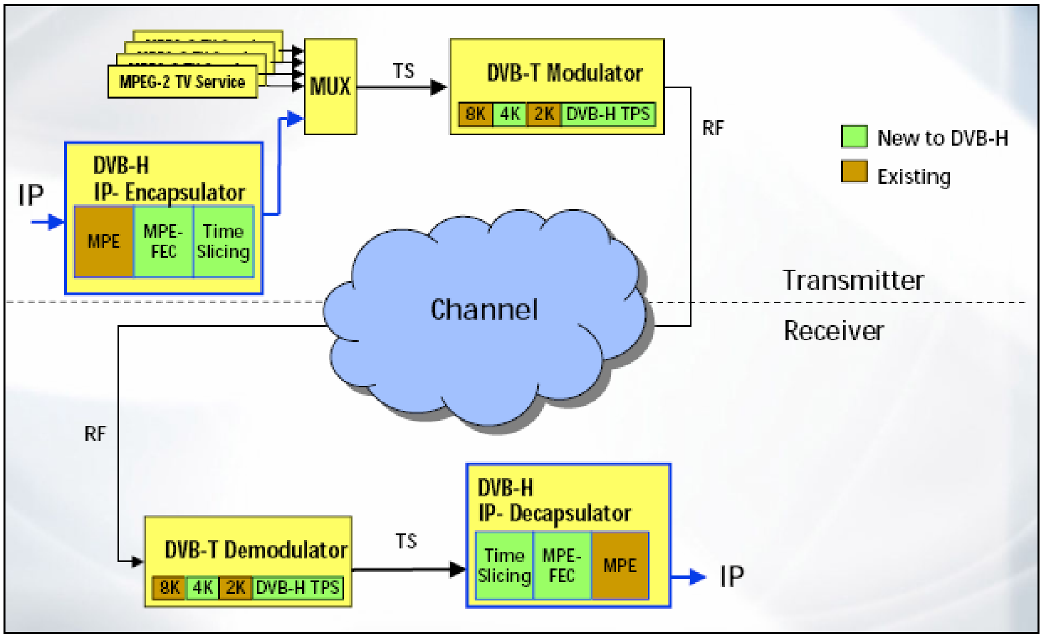


Figure 26 Solution: DVB-H System (When Sharing the Multiplex with MPEG-2)

The platforms that will be cooperating for the broadcast to be successful must be able to send IP signals as input (through UMTS) and return DVB-T signals for multiplexing. In other words the services must be backwards spectrally compatible.

6.3.3 Conclusion Upstream Technologies

There are many different cellular networks to choose among as an upstream technology, both EDGE and UMTS are good candidates due to already being in use across the globe. Currently 3G is the best choice due to its high capacity and availability across Europe at the moment and in the future. Later in time WiMax may be a good candidate, depending on how it is going to be developed and adapted.

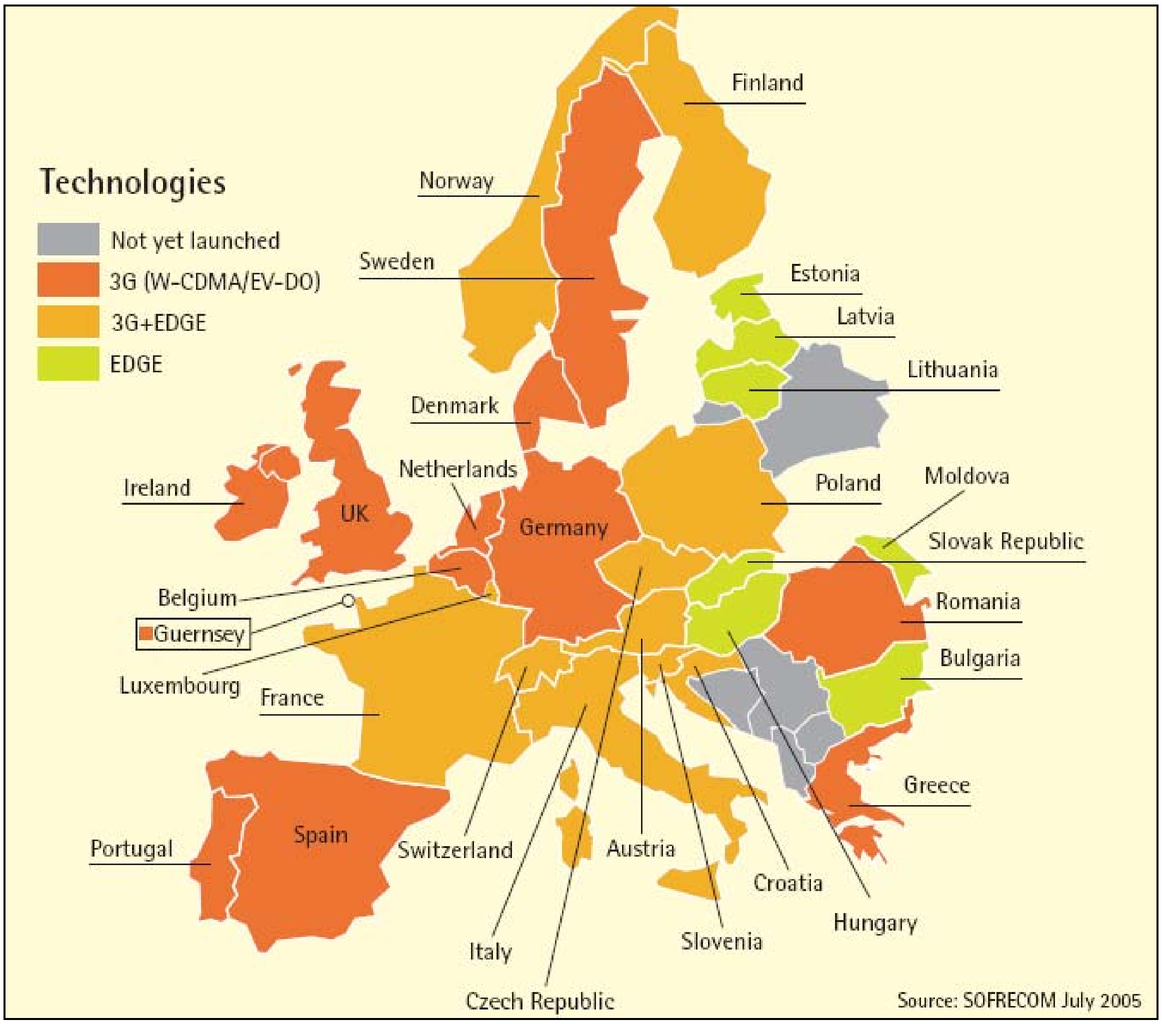


Figure 27 World Technology Overview [70]

7 Broadcasting Standards

7.1 DVB-H

Please see chapter 6.2.2
For full DVB-H specification, please see the DVB Homepage [71].

7.2 MPEG-4

Please see chapter 7.2
For full MPEG-4 specification, please see the MPEG Industry Forum [72].

7.3 3GPP

Please see chapter 7.3
For full 3GPP specification, please see the 3GPP Homepage [73].

Figure 28 Broadcasting Standard Logos

8 Broadcast & Streaming Protocols

The following Network and Transport protocols are used in the Nokia Mobile Broadcast Open air Interface and hence are chosen for the multimedia service in this Thesis.

8.1 IPv6

Internet Protocol version 6 (IPv6) is a network layer standard used by electronic devices to exchange data across a packet-swhiched internetwork and is specified in [74].
Ipv6 is an updated version of IPv4 and has several major advantages compared to it's previous version. IPv6 has a simpler header format and flow labeling. It also has improved support for extensions and options as well as Authentication and Security Extensions.
The size of the IP address is increased to 128 bits providing a simpler autoconfiguration of IP addresses. Multicast routing has been improved by adding a scope field to the multicast addresses and anycast addressing has been added.

The IPv6 packet consists of a header and payload. The header contains source and destination addresses which are 128 bits each, 4-bit IP version, 8 bits traffic class which determinates the priority of the Packet, 20bits flow label (QoS management), 16s bit payoad length, 8 bits next header, and 8 bit hop limit which is the packets time to live. The payload can be up to 64k in standard mode, or bigger with the use of "jumbo payload" option. The sending host takes care of fragmentation and the Next Header field specifies the transport layer protocol used by the packet's payload or the presence of an extra options header following the IPv6 header.
The payload's protocol itself is specified in a field of the options header and is analogous to the handling of AH and ESP in IPsec for both IPv4 and IPv6.

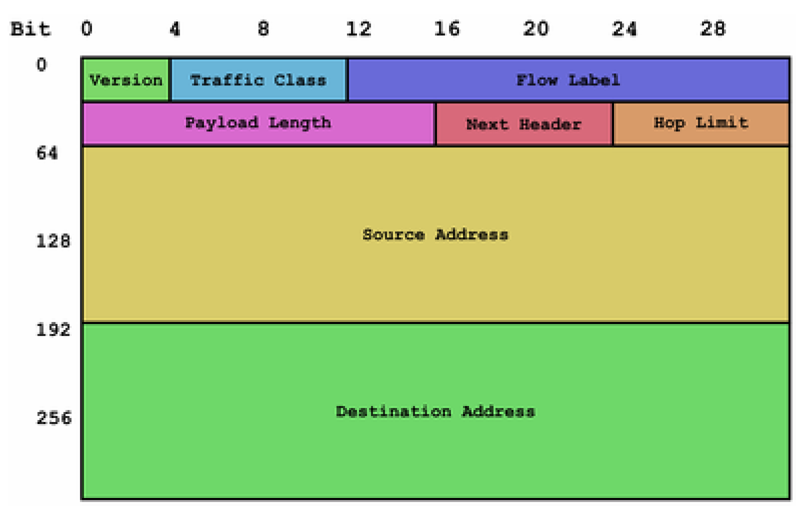


Figure 29 IPv6 Packet Format [75]

By providing more addresses for networked devices, IPv6 can allow mobile phone and other mobile handheld devices to have their own addresses. Compared to IPv4 which supports 4.3×109 (4.3 billion) addresses, IPv6 supports 3.4×1038 addresses, or 5×1028(50 octillion) for each of the roughly 6.5 billion people alive today.

8.2 UDP

The User Datagram Protocol (UDP) is a short message transport protocoll and is one of the core protocols of the Internet protocol suite. It is used to send datagrams (short messages) between computers in a network. UDP is desribed in RFC 768. During the message exchange between computers, UDP attaches source and destination port number fields for the multiplexing/demultiplexing service to the messages that are sent from the application process', adds two more fields, and passes the resulting "segment" to the network layer where the segment gets encapsulated into an IP datagram at the network layer which then makes a best-effort attempt to deliver the segment to the receiving host. If the segment arrives at the receiving host, UDP uses the port numbers and the IP source and destination addresses to deliver the data in the segment to the correct application process. There is no handshaking between sending and receiving transport-layer entities before sending a segment, which is also why UDP is said to be connectionless.

+	Bits 0 - 15	16 - 31
0	Source Port	Destination Port
32	Length	Checksum
64	Data	

Figure 30 UDP Header Format [76]

The UDP header consists of 4 fields (the pink ones are optional). The Source port field (16bits) is used to identify the sending port and is asumed to be the 'reply to' port if needed, else it is set to zero. The Destination port field is required and identifies the destination port. The
Length field (16 bits) specifies the length of the whole datagram (header and data) and can be minimum 8 bytes as that is the length of the header. The Checksum field is also 16 bits and is used for error-checking of the header and data using checksum. More information about UDP can be found in FRC 768.

Unlike TCP, UDP does not provide reliability and ordering guarantees, datagrams may arrive out of order or get lost withought notice. UDP does not check if every packet actually has arrived and is terefore faster and more efficient for many lightweight or time-sensitive purposes. UDP is required for broadcasting as wel as multicastinng and is brilliant for servers that have to answer small queries from huge amounts of clients. According to Wikipedia [76], UDP is used in 20 percent of the Internet traffic compared to TCP (75 percent) as of 2006.

8.3 RTP & RTSP

The Real-time Transport Protocol (RTP) defines a standardized packet format for delivering audio and video over the Internet and is defined in RFC 1889.

RTP was originally designed as a multicast protocol, but is today used in several unicast applications such as streaming media systems (in conjunction with RTSP) as well as videoconferencing and push to talk systems (in conjunction with H.323 or SIP), making it the technical foundation for Voice over IP. RTP is build on top of UDP and goes along with RTCP. Applications which use RTP are less sensitive to packet loss and very sensitive to delays.
RTP does not provide any Quality of Service and hence RTCP is also used.

RTCP stands for Real-time Transport Control Protocol and is defined in RFC 3550 [77]. It provides out-of-band control information for an RTP flow and helps RTP deliver and pack multimedia data. However it does not transport any data itself and is used periodically to transmit control packets to participants in a streaming multimedia session. Its primary function is to provide feedback on the quality of service which is provided by RTP by colecting statistics on a media connection as well as information such as sent bytes and packets, lost packets, jitter, feedback and round trip delay. This information is called QoS report and may be used by applications to increase their quality of service. RTCP does not provide any flow encryption or authentication, that is taken care by IP Security Encryption using AES-128 [78].

In the Nokia Mobile Open Air Interface, the RTP is ues for two major purposes; transport of H.246/AVC content [RFC 3984 [79]] and transport of generic MPEG-4 content [RFC 3640[80]]
However some restrictions are applied to the RFC specifications and they can be found in the Nokia Mobile Broadcast Open Air Interface v.1.0

8.4 SDP

Session Description Protocoll (SDP) is a format for describing streaming media initialization parameters and is described in RFC 2327 [82].
SDP is used for describing multimedia sessions such as session announcement, session invitation, and other forms of multimedia session initiation and is used in conjunction with RTSP, SIP or as a standalone format for describing multicast sessions.
All content streaming, regardless type, in the Nokia Mobile Broadcast Solution is done with multicasting and the terminals know what application to use through the SDP files in the Electronic Service Guide.

8.5 ACL & Flute

The Nokia Mobile Open Air Interface uses Flute (File Delivery Over Unidirectional Transport) [RFC 3926] with Asynchronous layered Codeing (ALC) Protocol Instantiation [FRC 3450] for File delivery for Mobile Broadcast.

9 The Return Channel

The return channel for the service has to be IP based. IP Datacast is chosen due to the one-to many broadcast service with an interactive return channel. IP Datacast offers the customers the possibility to interact with Mobile Network Operators and Broadcasters through voting, service subscription, purchase, data and web-based services etc. Through the Return Channel the viewers can order products and services, or be sent to a web site or other digital content. IP Datacast also links with the billing and e-commerce systems used by the Mobile Network Operators and provides the Interactive User Guide (IUG), which moves viewers seamlessly from between the broadcast channels and environments and the Mobile Network Operators' cellular services.

The most common ways of interacting between users of services offered on mobile phones and the service provider are listed below;

- SMS
- WAP
- SIP - Session Initiation Protocol, defined in RFC 3261.

According to Wikiedia [83] SIP is designed to enable IP-based peer-to-peer connectivity in the core of the network.
SIP is meant to be used for voice/video/messaging/gaming and push to talk.

9.1 SIP

SIP is a HTTP-like protocol using two types of messages; request and response and can with some extensions be used for presence and instant messaging.

Proposed extensions:

- Notify (examples);
    - A user can ask to be notified about a live event he is subscribed to watch at a certain day and time, for example1hour and 5 minutes before the live event is about to begin.
    - A user may want to be notified which of his friends are watching the event with him

- Subscribe (examples);
    - A user can subscribe to certain artists or type of live events, music etc.

- Message (examples);
    - A user may want to PM (send Private Message) to his friends, or favorite artist, report something to the administrators etc.
    - Chat with multiple user while viewing an event.

Video and messaging are the most important aspects of the system as well as SIP being chosen by 3GPP for signaling in UMTS. SIP can identify users through SIP URI's which are similar to an email address. Unique identifier for a user is in the following format:

sip:first.last@operator.com [84]

The general SIP URI format looks like this:

sip:user:password@host:port;uri-parameters?headers [85]

"Note: A sip URI with username@hostname:5060 is not the same as Username@hostname. If the port number is given, a DNS gethostbyname is used to find the host. If there is no port number, the hostname is looked up with DNS SRV. This hostname can point to one or several SIP proxy servers." [85]

The format specifies the subject, media type, or urgency of sessions initiated by using a URI on a web page or in an email message. Additional parameters can be added using ':' and is used to route SIP requests.

IP Multimedia Subsystem (IMS) users can extend the format to include his or hers mobile phone number by using the following format;

sip:+4792048839@miss-jarea.com;user=phone

This format is also more appropriate when used by the system for PSTN interaction. The SIP URI can also define sips, which means that the transfer is safe using Transport Layer Security. This can be done by changing the sip with sips in the last format listed;

sips:+4792048839@miss-jarea.com;user=phone

9.1.1 IP Multimedia Subsystem

The IP Multi-Media Subsystem (IMS) is an IP multimedia and telephony core network defined by 3GPP. IMS is access independent since it supports IP-based peer to peer connectivity in the core network. IMS is standardized reference architecture and consists of session control, connection control and an applications services framework along with subscriber and services data. IMS enables new converged voice and data services, while allowing for the interoperability of these converged services between subscribers [87].

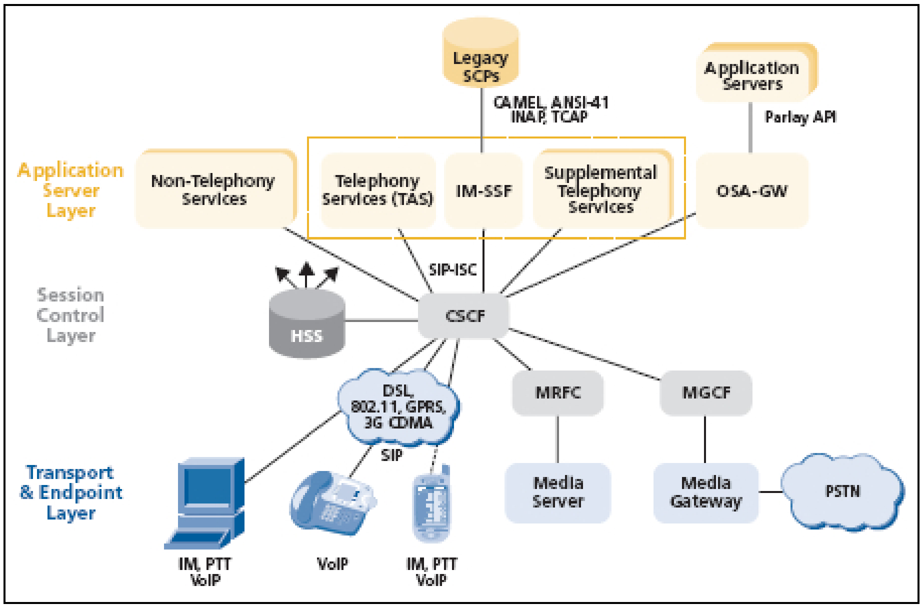


Figure 31 Simplified View of the IP Multimedia Subsystem (IMS) [86]

<u>IMS functionality includes:</u>

- IP connectivity-based development
- Access-independent processing – CDMA2000, GPRS, WiFi, 3G
- QoS for IP multimedia
- Policy control for efficient use of media resources
- User and data security and authentication using SIP
- Charging capabilities

9.1.2 IMS Message Format

SIP for IMS is defined in the RFC 3428 [87].

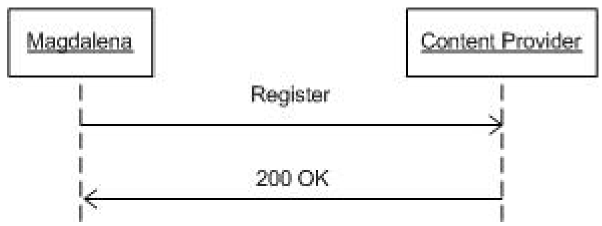


Figure 32 Example of message exchange using SIP

Optional message body:

```
MESSAGE sip:ContentProvider@domain.com SIP/2.0
Via: SIP/2.0/TCP MagdalenaPC.miss-jarea.com;branch=z9hG4bK776sgdkse
Max-Forwards: 70
From: sip:Magdalena@miss-jarea.com;tag=49583
To: sip:ContentProvider@domain.com
Call-ID: asd88asd77a@1.2.3.4
CSeq: 1 Register
Contact: sip: Magdalena@miss-jarea.com
Expires: never
Content-Type: text/plain
Content-Length: 43

You are now a member of 3G services, enjoy.
```

ContentProvider replies to the Registration

```
   SIP/2.0 200 OK
   Via: SIP/2.0/TCP proxy.domain.com;branch=z9hG4bK123dsghds;
                                            received=192.0.2.1
   Via: SIP/2.0/TCP MagdalenaPC@miss-
jarea.com;;branch=z9hG4bK776sgdkse;
                                               received=1.2.3.4
   From: sip: Magdalena@miss-jarea.com;tag=49394
   To: sip:ContentProvider@domain.com;tag=ab8asdasd9
   Call-ID: asd88asd77a@1.2.3.4
   CSeq: 1 MESSAGE
   Content-Length: 0
```

The format would be:

```
   Content-Disposition: session
   Content-Type: application/sdp
   Content-Length: 200
```

When local proxies are used for transmission of outbound messages, proxy authentication, as specified in RFC 3261, is absolutely recommended due to spoofing and spamming prevention at the original network as well as verifying the identity of the originator. The user agent can encrypt the message end-to-end, meaning that the proxies do not need to parse the message body to route the message, which means that the proxies have no idea what kind of session is established.

9.1.3 Instant Messaging

Instant Messaging also known as IM is one of the most common services offered on 3G today. It resembles the already well known services MSN / ICQ / AOL etc. This is used by millions of computer users today. IM is going to offer the users the possibility to interact with other users as well as the service providers.

The instant messages will most likely be short real-time messages and will not be stored in intermediate nodes.

IM is specified in A Model for Presence and Instant Messaging RFC 2778 [88] and in Common Profile for Instant Messaging (CPIM) RFC 3860 [89].

IM uses *im* URI with the following syntax which follows the existing mailto: The URI syntax is specified in RFC 2368 [90].

```
IM-URI         = "im:" [ to ] [ headers ]
to             =  mailbox
headers        =  "?" header *( "&" header )
header         =  hname "=" hvalue
hname          =  *uric
hvalue         =  *uric
```

An example of an *im* URI can be: Magdalena@miss-jarea.com.

Instant messaging can e divided into two modes; pager and session. Pager is used for stand-alone messages and is preferred when a single message is sent for example from the user to the service provider requesting a service or content provider to request a type of content. Session mode would be preferred for example during chatting sessions, voting etc and hence is very useful for the multimedia service in this thesis.

SIP proxies however have the disadvantages of being able to change the transport protocol from for example TCP to UDP without congestion problems. So messages sent should be kept under Maximum Transfer Unit minus 200bytes.

Larger messages can be sent using session based messaging where SIP INVITE is used to establish a connection between users, users and artists or content providers etc. For messaging MSRP (Message Session Relay protocol) can be used [91]. MSRP runs over transport protocols like TCP,SCTP and TSL, which do offer congestion control. MSRP also runs over the media plane and hence does not bother SIP proxies with large messages.

Larger messages can be stored on a server and then accessed using HTTP by opening the link with the URI with points to the message to obtain it from the server. This can be used by the users to download files like tracks, ring tones, music videos and wallpapers etc. during live sessions or on demand.

9.1.4 XML

The Extensible Markup Language (XML) is a general-purpose markup language for creating special-purpose markup languages and is very similar to HTML. XML is used to describe different kinds of data as well as being able to contain the data. XML is a simplified subset of Standard Generalized Markup Language (SGML) with the purpose to facilitate the sharing of data across different systems, particularly systems connected via the Internet. Languages based on XML, such as XHTML, SVG, and MusicXML among many others are defined in a formal way, allowing programs to modify and validate documents in these languages withought prior knowledge of their form. Moree informaytion about XML can be found in the XML Tutorial [92]

The figure below shows a XML example taken from Mobile Broadcast Open Air Interface v1 and illustrates the messages which are exchanged between the Moile Broadcast Client and serve using the Open Air Interface 1.0. The example asumes that the two items available for purchase; “sports channel” (itemID 101) and “entertainment channel” (itemID 102) are defined and received by the Moile Broadcast client via the service guide.

Example of the start of a purchase of a continous subscription to the “sports channel” on January 15th is shown on the next page;

```xml
<?xml version="1.0" encoding="UTF-8"?>
<purchaseRequest interfaceVersion="mbs-oai-1.0" >
 <user>
   <userID>24403123456</userID>
   <lang>en</lang>
 </user>
 <device>
   <deviceID>0044005817853</deviceID>
   <riDeviceID>vXENc+Um/9/NvmYKiHDLaErK0gk=</riDeviceID>
 </device>
 <serviceOperatorCentre>
   <socID>1</socID>
 </serviceOperatorCentre>
 <purchaseItemList>
   <purchaseItem>
     <itemID>101</itemID>
     <purchaseOption>1</purchaseOption>
     <price currency="EUR">5.00</price>
   </purchaseItem>
 </purchaseItemList>
<signature>h8twx3rvEPO0vKtMup4NbeVu0f1O</signature>
</purchaseRequest>
```

Figure 33 Example of XML Syntax [81]

XML documents must have the correct syntax (be well formed). XML documents get validated against a Document Type Definition (DTD) or a Schema which define the building blocks of an XML document. For this purpose XDS schemas are used.

10 Video & Streaming

This chapter will cover Video and Streaming for Mobile phones.

10.1 Streaming

Streaming is the transmission of real-time data like audio, video as well as other multimedia from a server to a client. Streaming has been available for computers for many years now and with the development of faster mobile networks such as GPRS, EDGE and UMTS, it is now possible to stream data to mobile phones as well.

A user that requests a media file can view or hear its content after a short buffering period. The data sent from the operator is encoded and the content file is split into small packets which are streamed in a continuous flow to the mobile phone. At the receiving end, the stream is decoded and becomes viewable for the user. Like streaming from Internet, the user can view the content immediately once the first packets arrive and the users end. Mean while the rest of the stream is transferred while what already has arrived is shown. When a user wants to start viewing some media there will always be a short delay at the start of the streaming. The delay is caused by the time it takes to buffer the first of the data packets and does not affect the rest of the stream. Differences in data rates during streaming can occur but do not affect the content the user is watching at a current time.

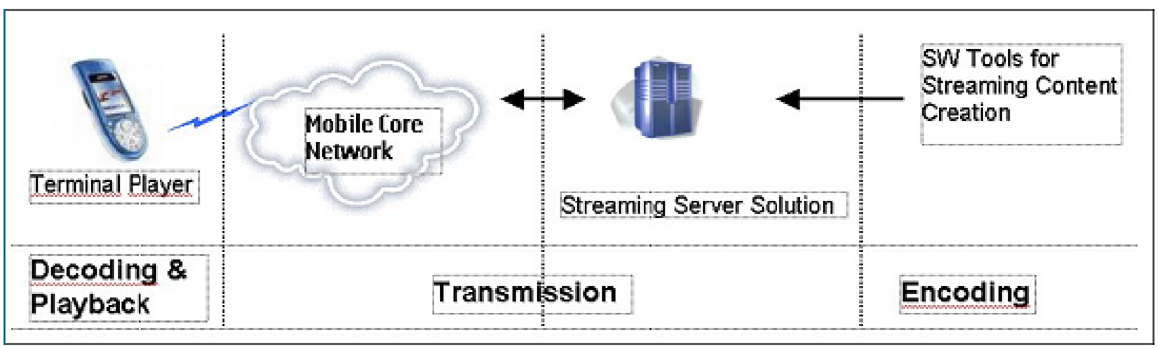


Figure 34 Streaming in Nokia mobile Network [59]

The availability of transfer bandwidth differs from operator network to operator network, all depending on the access technology used and the configuration of the network.

Due to the fact that there is no data stored permanently on the Mobile device, a user can not save nor forward the content to other users, which makes it very profitable for the providers that can and are currently charging per view.

10.1.1 Content Delivery Methods

- Broadcasting
  - Point-to-multipoint transmission by wireless means for general reception
- Datacasting
  - Broadcasting data, typically streams or files (filecasting)
- Streaming
  - Technique for transferring data such that it can be processed as a steady and continuous stream
- Unicasting
  - Point-to-point streaming (live or on-demand content)
- Multicasting
  - Point-to-multipoint streaming (live content)
- Webcasting
  - Live streaming over Internet
- Downloading
  - Transferring data such that it cannot be processed until it has been completely received

10.2 Streaming Applications

The applications for streaming services can be divided into "on-demand" and "live". On-demand applications include video, on-demand-news and on-demand instructional material. Live streaming applications include reception of radio and TV broadcasts.

Delivery method	Downloading	Pre-downloading	Streaming
On-demand content	Yes	Yes	Yes
Live content	No	No	Yes
Max content length	Phone memory size	Phone memory size	Infinite
Content quality	Good (limited by download time)	Good (limited by download time)	Limited by the network bandwidth
Stored in memory	Yes	Yes	No
Content repeat	While stored in memory	While stored in memory	Must be re-streamed
Viewing delay	Viewing starts after download complete	None (when downloaded)	Few seconds
Off-line viewing	No	Yes	No
Server	Web/FTP server	Web/FTP server	Streaming server
Protocol/transport	http (TCP)	http (TCP)	RTP (UDP)

Table 7 Streaming Applications [70]

10.3 Streaming Challenges

The implementation of mobile streaming services faces two big challenges: access network and terminal heterogeneity as well as content protection.

Currently we have access to a variety of mobile terminals with a wide range of display sizes and capabilities. Additionally different radio-access networks make multiple maximum-access link speeds available, but due to the physical characteristics of cellular radio networks the quality and the data rate during the connections vary and hence contribute to the heterogeneity problem. Heterogeneity can be solved by using appropriately designed capability exchange mechanisms that enable the terminal and media server to negotiate mobile terminal and mobile network capabilities as well as user preferences. By doing so, the server sends multimedia data already adapted to the user's mobile terminal and the network. This approach can for example let a user accessing a specific service via a WCDMA network to get the content delivered at a higher bit rate than other using general packet radio service or GSM network.

Similarly, users using mobile phones with a built-in low-quality speaker will get an automatic upgrade of the transmission to a high-quality audio stream when they plug in high-fidelity headphones due to a dynamic capability exchange between the terminals and streaming server that recognizes the headset. Efficient delivery of streamed multimedia content over various radio-access networks with different transmission conditions is achieved if the media transport protocols incorporate the specific characteristics of wireless links. Retransmissions of corrupted data packets can be solved by caching data packets and optimizing the data transport over the wireless links to a mobile terminal by proxies [93, 94].

At the application level, all content protection is taken care of by disallowing the storage of received content. This is done by adapting the digital rights management (DRM) concept, which uses techniques such as encryption [Appendix] and conditional access based on usage rules to protect and manage access to multimedia data. Content providers must use the DRM mechanisms to be able to deliver top-notch content over digital networks and at the same time make sure the content is not illegally ripped and distributed.

10.4 Advantages of Streaming

- Instant play
- Deliver long form media
- Deliver live broadcasts
- Multicasts (one stream to many viewers)
- Eliminates the need to download files to the hard drive
- Distribution control
- Can stream individual tracks into a movie from any streaming server anywhere

10.5 User scenarios

As mentioned in chapter 2.2, the service is suppose to offer the following user scenarios;

- Live streaming of music and video (live broadcasting)
- Streaming video on demand (broadcasting of past events)
- Streaming and download of music on demand
- Streaming of news (audio/video) on demand
- Streaming of commercials on demand
- Interactive content
- Invites
- Chatting

10.5 Basic components for mobile video

To be able to stream video content to a mobile, a few basic components are needed;

- encoder
- server
- player

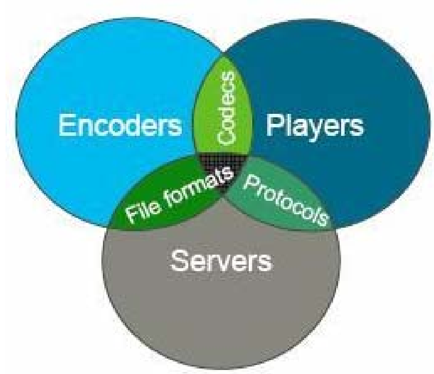


Figure 35 Basic Components for Mobile Video

Standard codecs (video: H.263, MPEG-4, 3GPP), file formats and protocols must be used for interoperability. It is also very important that both proprietary and 3GPP compliant codes, file formats as well as protocols exist.

10.6 Streaming Codecs and File Formats

We can divide the codecs and formats in today's marketplace into three categories: mobile standards, vendor standards, and proprietary codecs.

The mobile standards are specified mainly by 3GPP (3rd Generation Partnership Project see chapter 10.6.3) which provides specifications for most of today's GSM-based networks.
On the providers side, the multimedia 3GPP standard specifies the container for video content, as well as the supported video and audio codecs multiplexed within that container.
The two primary video codecs used in today's phones are: H.263 and MPEG-4. H.263 is mostly used in videoconferencing applications and offers extremely low latency, making it well suited for live applications. Since the CPU requirements for H.263 decoding are extremely light, videoconferencing takes little CPU power. On the negative side H.263 is an 'old' codec and is extremely bandwidth-inefficient.

MPEG-4 (MPEG-4 Simple Profile), is a more modern version of H.263 and has many of the same latency and decoding characteristics, but with greater encoding efficiency. Most content providers have so far been using MPEG-4 as their video codec for 3GPP content. H.264, MPEG-4 Part 10, also known as advanced video coding or AVC, is a digital video codec standard which is noted for achieving very high data compression.

Due to its promises of two to three times more encoding efficiency when compared with MPEG-2, the current broadcast industry standard, H.264 encoding is wanted in everything from mobile devices and HD-DVDs to transmission of high-definition television networks. However considering a phone life cycles, content providers will have to continue H.263 or MPEG-4 video encoding for the foreseeable future.

The 3GPP specification has several audio codecs that provide a good mix of dynamic range and encoding efficiency for the content provider:

- AAC (advanced audio coding) is ideally suited for music content and provides efficient fidelity at low data rates (16 to 32 Kbps).

- AMR (adaptive multi-rate) provides good voice reproduction at ranges of 4.75 to 12.2 Kbps.

- QCELP (Qualcomm code excited linear predictive), based on the Qualcomm PureVoice codec, is included in the 3GPP2 standard for CDMA-based wireless networks.

The two prevalent vendor standards are RealNetworks' RealVideo and Microsoft's Windows Media codecs/formats.

Many of these are based on the J2ME virtual machine and are viable ways of providing video content to customers whose phones don't have native video support. The codecs are optimized for low-bandwidth operations and decoding efficiency. Content encoding in these formats is done with tools provided by the codec vendor. Standards-based J2ME decoders are difficult to implement creating a niche for these optimized codecs and most providers produce their content in most if not all of the codecs and formats in use today.

The following streaming codecs are currently in use; MPEG-4 (* H.264 codec is known also as MPEG-4 AVC.), 3GPP (PSS (Packet Switched Streaming)), RealMedia, Windows Media and QuickTime.

File Format	File Extension	Video Codecs	Audio Codecs
MPEG-4	.mp4	MPEG-4 Visuals, H.264*	AAC
3GPP	.3gp	MPEG-4 Visual Simple Profile, H.263	AMR, AAC
RealMedia	.rm, .ra	RealVideo 8	RealAudio
Winwows Media	.wmv, .wma	Windowd Media Video 8	Windows Media Audio
QuickTime	.mov, .qt	MPEG-4 Visuals, Sorenson Video	AAC, AMR

Table 8 Streaming Codecs and File Formats

10.6.1 Video Coding

A video sequence is a series of still images that have been compressed to reduce the redundant and perceptually irrelevant parts of the video sequence. There are three different types of redundancy; spatial, temporal, and spectral redundancy. Spatial redundancy refers to the correlation between neighbouring pixels. Temporal redundancy means that the same objects appearing in the previous image are likely to appear in the current image as well. Spectral redundancy addresses the correlation between the different colour components of the same image. The current image is sort of predicted from the previous one.

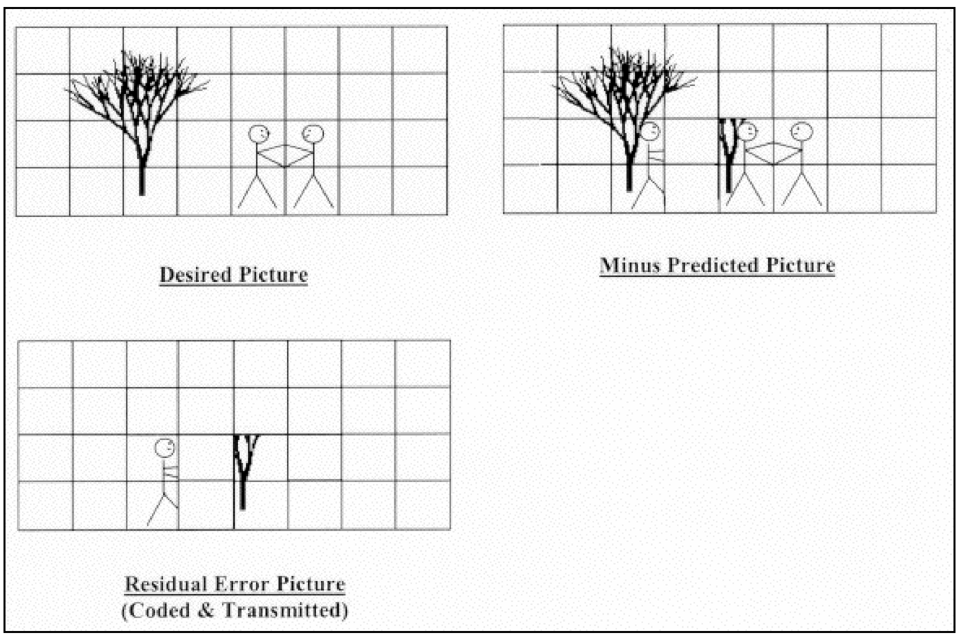


Figure 36 INTRA Decoding

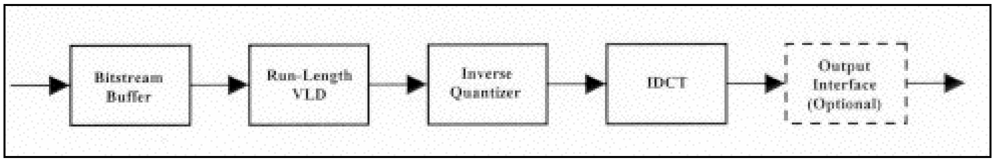


Figure 37 INTRA Encoding

Video compression can be achieved by generating motion-compensation data (which describes the motion between the current and the previous image).
To achieve efficient compression, non-redundant information must be discharged by the video encoders (least important information for the subjective quality of the image). Additionally the redundancy of the encoded bit stream is reduced by means of efficient lossless coding of compression parameters and coefficients.

The mainly used video coding technique is called Huffman coding [95] and uses variable-length codes in place of ASCII codes, and the most common characters, usually *space*, *e*, and *t* are assigned the shortest codes. In this way the total number of bits required to transmit the data can be considerably less than the number required if the fixed length representation is used. Huffman coding is particularly effective where the data are dominated by a small number of symbols.

Video compression methods usually differentiate between images that can use temporal redundancy reduction and images that can not. Compressed images that do not use temporal redundancy reductions methods are usually called INTRA or I-frames, while the temporally predicted images are called INTER or P-frames [96]. The predicted motion-compensated image is rarely sufficiently precise in the INTER frame case, therefore a spatially compressed prediction error image is also associated with each of the INTER frames.
The INTRA technique is limited to processing the video signal on a spatial basis, relative only to information within the current video frame. Considerably more compression efficiency can be obtained however, if the inherent temporal, or time-based redundancies, are exploited as well. Temporal processing to exploit this redundancy uses a technique known as block-based motion compensated prediction, using motion estimation. A block diagram of the basic encoder with extensions for non-intra frame coding techniques known as INTER frame.

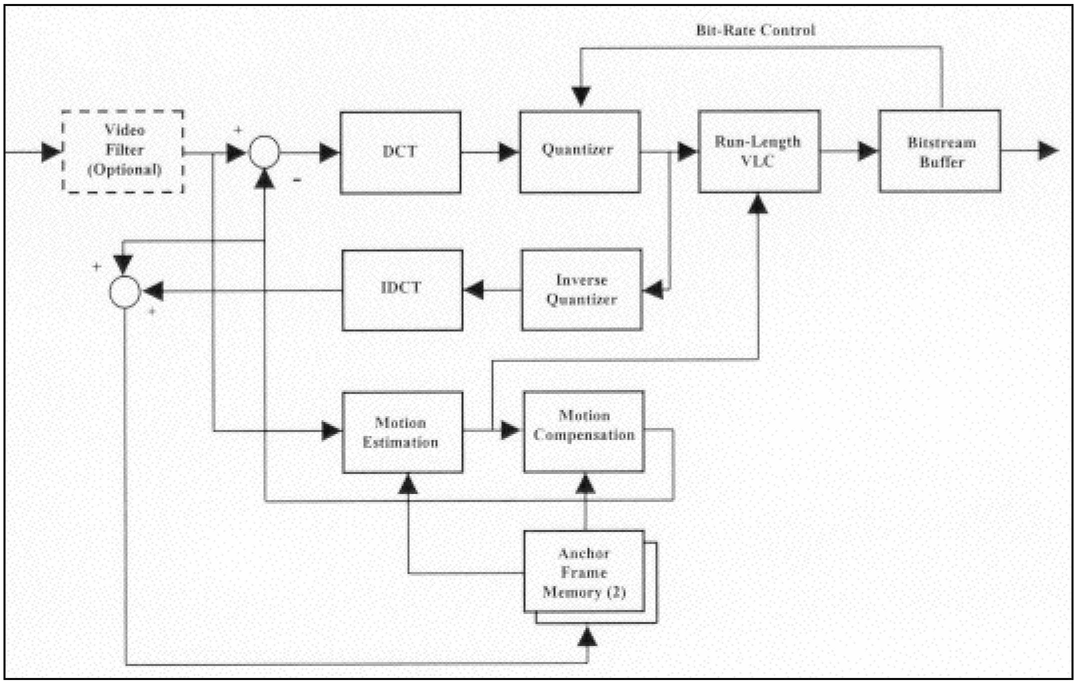


Figure 38 INTRA Frame Coding

There is always a trade-off between bit and rate when it comes to video coding. Since some image sequences can be harder to compress than others (due to rapid motion or complex texture), the video encoder must control the frame rate as well as the quality of images in order to meet a constant bit-rate. The harder the image is to compress, the worse the image quality. In cases where variable bit rate is allowed, the encoder can maintain a standard video quality.

10.6.2 Video File Formats

As one can see from figures 33-35, there is a difference between the video coding format and the video file format. While the coding format refers to the action of a specific coding algorithm that codes content information into a code stream, the file format is a way of organizing video and audio code streams so that they can be accessed for local decoding and playback or streamed over a transport channel.

The following list includes some of the most common video file formats today:

- Microsoft Audio-Video Interleaved (AVI)
- Apple QuickTime file format (.mov)
- MPEG-1 file format (.mpg)
- MPEG-4 / MP4 file format (.mp4)
- 3GPP file format (.3gp)

When it comes to Mobile phones, both Nokia which the service is suppose to be tested on and Ericsson support the 3GPP file format (extension: .3gp) for the storage of video and associated audio (if present). Unlike the Ericsson Video Player, the Nokia Video Player doesn't let a user play back media content inside .mp4 files (MPEG-4 file format), although the file may contain media types supported by the player. If the .mp4 file declares 3GPP file format compatibility, the content will play out as originally intended by the content author, otherwise the player presentation may not reproduce original content. Apple uses a .m4a file extension in its i-Tunes service, but the actual container is still MP4. The video player in Sony Ericsson mobile phones supports local playback and streaming of 3GPP media, mainly for Video On Demand (VOD) and live video applications [97,98].

10.6.3 Video Codecs

10.6.3.1 H.263

The H.263 standard, published by the International Telecommunications Union (ITU), supports video compression (coding) for video-conferencing and video-telephony applications. H2.63 was originally designed to be utilized in H.324 based systems (PSTN and other circuit-switched network videoconferencing and video telephony), but ending up used in H.323 (RTP/IP-based videoconferencing), H.320 (ISDN-based videoconferencing), RTSP (streaming media) and SIP (Internet conferencing) solutions as well.

H.263 was developed from the previous ITU-T standard for video compression H.261 and the MPEG-1 and MPEG-2 standards. Since its first complete version in 1995 providing a suitable replacement for H.261 at all bitrates, it has been enhanced in other projects known as new versions of itself, namely H.263v2 (a.k.a. H.263+ or H.263 1998) and H.263v3 (a.k.a. H.263++ or H.263 2000) [99].

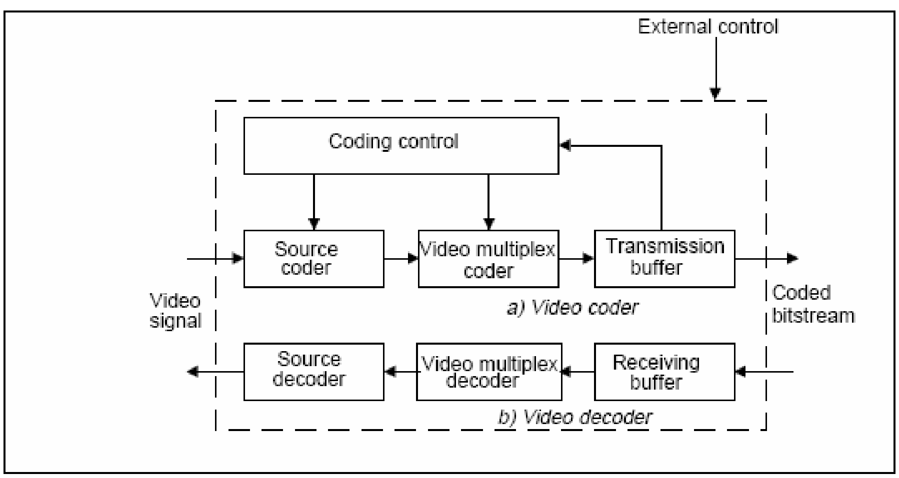


Figure 39 H.362 Outline block diagram of the video codec [99]

The H.263 standard specifies the requirements for a video encoder, decoder and the format and content of the encoded (compressed) stream. The H.263 contains various encoding tools and coding complexities for different purposes as well as under what circumstances they are to be used. The definitions are defined in so-called codec profiles and levels can be found in the ITU-T Recommendation H.263. The H.263 Profile 0, Level 10 has been defined as a mandatory codec for most mobile multimedia services and is the mainstream codec supported in Nokia video players. H.263 reduces the bandwidth by subtracting the previous transmitted frame from the current frame so that only the difference or residue needs to be encoded and transmitted. In other words the areas of the frame that do not change (like for example the background) are not encoded. Next step in reducing the bandwidth is to estimate which areas of the previous frame have moved to in the current frame and then compensating for this movement change. The motion estimation module compares each of the 16x16 pixel macro blocks in the current frame with its surrounding area in the previous frame and tries to find a match. Matched areas are then moved into the current macro block position by the motion compensator module. Then the motion compensated macro block is subtracted from the current macro block. If the motion estimation and compensation process is efficient, the remaining "residual" macro block should contain only a small amount of information.

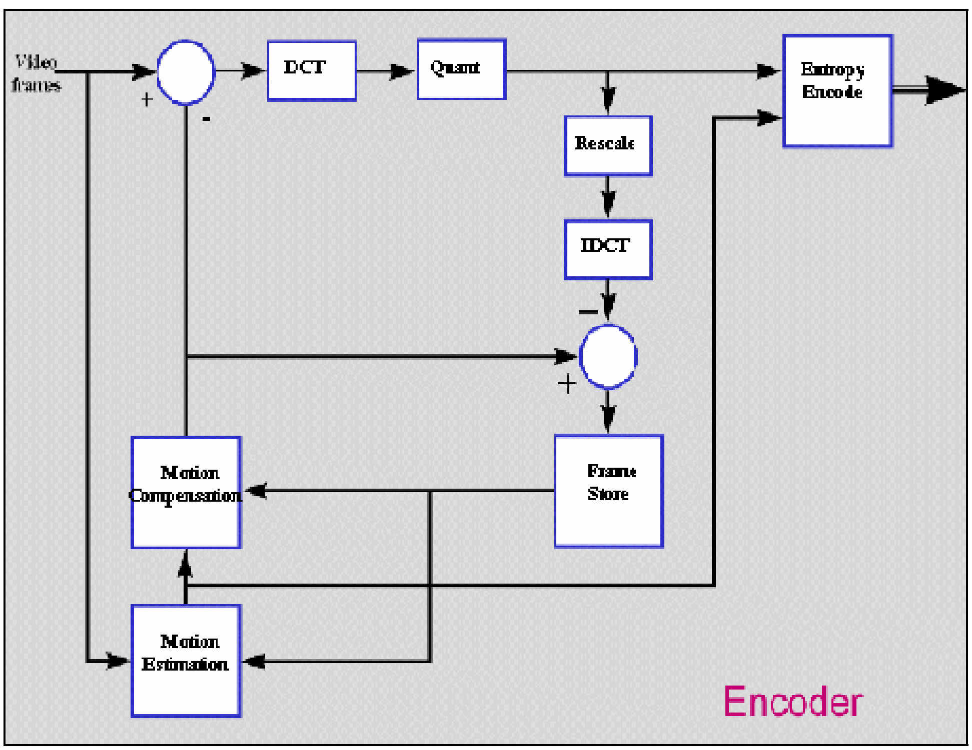


Figure 40 H.263 Encoder

H.263 uses the Discrete Cosine Transform (DCT) to reduce spatial redundancy by converting a block of pixels to coefficients that represent the spatial frequency components of the block. Few of the frequencies appearing in the block have high-amplitude coefficient values, meaning that most of the coefficients are close to zero. If we consider a block that constantly is coloured, it will only have one spatial frequency and therefore be transformed to one nonzero DCT coefficient. The other DCT coefficients will remain zero; hence it is easier to code the DCT coefficient block with run-length codes than the original block of pixels. In order to gain compression, the transformed block is quantized and the coefficients are rounded to certain quantization levels. The fewer possible quantization levels there are, the fewer bits it takes to represent a quantization level. To restore a original block of pixels, the inverse DCT transformation can be used. The more quantization levels used, the better quality of the reconstructed image. ITU-T H.263 allows 31 quantization step sizes that are controlled by the so-called quantization parameter.

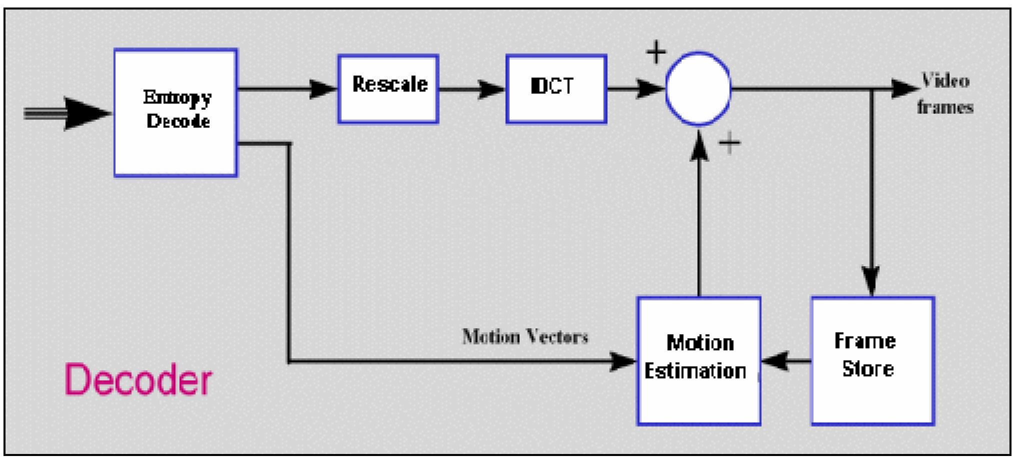


Figure 41 H.263 Decoder

In order to extract the coefficient values and motion vector information, the variable-length codes that make up the H.263 bit stream are decoded.

To reverse an quantization, one must multiply the coefficients by the same scaling factor that was used in the quantizer. However, the rescaled coefficients are not identical to the original coefficients, because the quantizer discarded the fractional remainder during encoding. The reversing transformation is called IDCT, where I stand for Inverse. IDCT creates a block of samples which correspond to the difference values that were produced by the motion compensator in the encoder. The difference values are then added to a reconstructed area from the previous frame and motion vector information is used to pick the same reference area that was used in the encoder, resulting in a reconstruction of the original frame. However the new frame will not be identical to the original due to the loss of information between the transformations resulting in a frame with worse quality then the original. The reconstructed frame is placed in a frame store and it is used to motion-compensate the next received frame.

The next enhanced codec developed by the ITU-T (in partnership with MPEG) after H.263 is the H.264 standard, also known as AVC and MPEG-4 part 10. As H.264 provides a significant improvement in capability beyond H.263, the H.263 standard is now considered primarily a legacy design (although this is a recent development). Most new videoconferencing products now include H.264 as well as H.263 and H.261 capabilities.

10.6.3.2 H.264

H.264 is also known as MPEG-4 Part 10, or AVC, for Advanced Video Coding is written by the ITU-T Video Coding Experts Group (VCEG) and the ISO/IEC Moving Picture Experts Group (MPEG) as the product of a cooperation known as the Joint Video Team (JVT). The ITU-T H.264 standard and the ISO/IEC MPEG-4 Part 10 standard are technically identical and is a digital video codec standard which is noted for achieving very high data compression.

H.264/AVC standard provides good video quality at bit rates that are substantially lower, such as half or less, than what previous standards need (MPEG-2, H.263, or MPEG-4 Part 2). It is easy to implement and possible to apply to a very wide variety of applications, like for example for both low and high bit rates, and low and high resolution video. The H.264 standard works very well on a wide variety of networks and systems such as broadcasting, DVD storage, RTP/IP packet networks, and ITU-T multimedia telephony systems.

H.264 achieves the best-ever compression efficiency for a broad range of applications, such as broadcast, DVD, video conferencing, video-on-demand, streaming and multimedia messaging. And true to its advanced design, H.264 delivers excellent quality across a wide operating range, from 3G to HD and everything in between. Whether you need high-quality video for your mobile phone, iChat, Internet, broadcast or satellite delivery, H.264 provides exceptional performance at impressively low data rates.

H.264 is now the new standard for the HD-DVD and Blu-ray specifications (the two formats for high-definition DVDs) and ratified in the latest versions of the DVB (Digital Video Broadcasters) and 3GPP (3rd Generation Partnership Project) standards. Many broadcast, cable, videoconferencing and consumer electronics companies consider H.264 the video codec of choice for their new products and services. This adoption by a wide variety of open standards means that any company in the world can create devices, mobile phones, set-top boxes, DVD players and more that will work with today's new streaming applications like QuickTime [100].

10.6.3.3.MPEG-4

This is an optional video codec in several recent multimedia standards and it is widely supported in more recent Nokia devices, in addition to H.263. Like H.263, MPEG-4 Visual (also called MPEG-4 Part 2) contains different profiles that describe functionality packages. Nokia platforms support Simple Profile 0. MPEG-4 Visual is a DCT-based compression standard divided into many levels which describes such things like frame size, bit rate and buffer capacity. The Simple Visual profile provides efficient error-resilient coding of video content and is supported among others by the Eriksson mobile phone's video player. The still texture part of MPEG-4 uses wavelet compression. MPEG-4 has been incorporated into the 3GPP specifications for mobile multimedia and supports functionality. More information about MPEG-4 can be found in the MPEG-4 Industry Forum (see chapter 7) [100,101]

10.6.3.4 3GPP PSS

3GPP is the new worldwide standard for the creation, delivery, and playback of rich multimedia for the next generation of mobile devices, such as multimedia-enabled mobile phones and other handheld devices. 3GPP was defined by the 3rd Generation Partnership Project, a group of telecommunications standards bodies, to provide uniform delivery of rich multimedia over high-speed wireless (3G) networks to the new 3G mobile devices. 3GPP is based on the MPEG-4standard for delivery of video and audio over the Internet, and includes specific tailoring for the unique requirements of mobile devices on the UMTS system. More information about the 3GPP can be found on the 3GPP Homepage (see chapter 7) [102,103].

Due to the fact that the 3GPP technology is based on the MPEG-4 (ISO base file) format,
it offers the same high quality, flexibility, and scalability of MPEG-4. Both standards are based on the QuickTime technology and operators who use the 3GPP standard also use the MPEG-4 video codec and AAC but are not directly reliant or dependent on MPEG-4.

Streaming over fixed-IP networks is common in today's applications but no complete standardised streaming framework has yet been defined. For 3G systems, the 3G packet-switched streaming service (PSS) fills the gap between 3G MMS, e.g. downloading, and conversational services.

PSS enables mobile streaming applications, where the protocol and terminal complexity is lower than for conversational services, which in contrast to a streaming terminal require media input devices, media encoders and more complex protocols

With the release of QuickTime 6.3, Apple is the first company to distribute a high-quality, low-cost, standards-based solution for 3GPP content creation worldwide. 3GPP support in the QuickTime-based authoring application in QuickTime 6.3 offers all the tools a content creator needs to be able to produce multimedia content for a whole new audience of mobile multimedia devices.

In January 2003, NTT DoCoMo launched its Mobile MP4 i-motion service, and with it the world's first standards-based 3G network which can stream multimedia content to a phone or between phones. Because the Mobile MP4 service is based on the 3GPP standard, users can receive top-notch multimedia content, create their own local content, and share it among all types of 3GPP-enabled desktop computers and mobile devices.

3GPP should not be confused with 3GPP2, which specifies standards for another 3G technology based on IS-95 (CDMA), known as CDMA2000.

The 3GPP standard is supported by many operators, including Telenor and Netcom and manufacturers like Nokia and Sony/Ericsson (3GPP specification TS 26.234, "Transparent end-to-end packet switch streaming service (PSS)".

10.6.3.5 RealVideo

RealVideo is a proprietary video format developed by RealNetworks, first released in 1997. Currently version RealVideo version 10 is available. RealVideo is supported on many platforms, including Windows, Mac, Linux, Solaris, and several mobile phones (Nokia).

RealVideo is usually paired with RealAudio and packaged in a RealMedia (.rm) container. RealMedia is suitable for use as a streaming media format that is one which is viewed while it is being sent over the network. Streaming video can be used to watch live television, since it does not require downloading the video in advance [104].

The RealOne Player installed on Nokia devices supports the decoding of video in RealVideo 7 and RealVideo 8 formats. Some Nokia devices also support the decoding of video in RealVideo 9 and the new RealVideo 10 format.

Both RealVideo 7 and 8 are similar to MPEG-4 and H.263 in many ways. They both make use of I-frames and P-frames and are block transform-based. They also uses Bidirectional frames or B-frames that are computed based on interpolation between the previous and next I- and P-frames using motion vectors and spatial information. However RealVideo 7 is more efficient than MPEG-4 or H.263 due to the use of B-frames and some advanced coding techniques created by RealNetworks. RealVideo 8 on the other hand is 30 to 40 percent better than RealVideo 7 due to major differences in the specific techniques used for transform coding, motion estimation, and filtering.

RealVideo 10 includes enhancements on the encoder side, so it is compatible with all RealVideo 9 capable players. RealNetworks states that RealVideo 10 provides at least the same visual quality as RealVideo 9, using a 30 percent lower bit rate.

The RealPlayer installed onboard some of the Nokia devices supports the decoding of video in RealVideo 9 and RealVideo 10 formats.

10.7 Video Player

In video streaming, the compressed video data is fetched from a server during playback, while in the local video viewer, the video clip that is to be viewed is stored locally in the memory of the mobile terminal or memory card.

The WAP browser or messaging applications such as SMS and MMS viewers, is used to implement the external session establishment of video streaming service and have the ability to launch external applications such as the player application. In the streaming player application, the streaming links are typically launched from the player's editable play list by recognizing a
" rttp;// " URL or a downloaded definition file (.RAM file).
It is also possible to download a file first and then play. This is considered local playback, since the entire video file has already been stored locally.

There are many different video players for mobile phones, such as RealPlayer, MediaPlayer, Quicktime, PacketVideo, MVideo and many more.

The video player in Nokia uses the same interface for both video streaming and a stand-alone local video player. According to the Nokia Forum, Nokia devices do not support progressive downloading, which means that the entire video clip must be downloaded before playback.

10.8 Audio Coding

Arbitrary sounds can be represented as a sum of waves having different frequencies and amplitudes (any sound is an amplitude waveform as a function of time). By sampling a corresponding waveform with small frequency intervals, the sound can be digitized.
A 44.1 kHz sampling frequency is considered to provide high quality for arbitrary sounds and music, while an 8 kHz sampling frequency is adequate for most speech applications. 16 bits is usually enough to represent one sample. Digitized audio can be compressed in various ways, such as the simple coding method which uses an adaptive step size to quantize audio samples. This technique is used in the IMA ADPCM audio coding standard that reserves 4 bits per sample. Hence, if the sampling frequency is 8 kHz, IMA ADPCM-coded audio takes 32 Kbit/s. Another simple audio coding method is A-law PCM, which uses a logarithmic quantization step size and reserves 8 bits per sample. More advanced audio coding methods take advantage of the human psychoacoustics model where parts of the audio signal are barely audible and can be discarded or compressed. The advanced coding audio methods are usually categorized into generic audio coding and speech coding techniques. The first one uses algorithms that are targeted for music and sound as well as human voices, whereas speech coding algorithms performs relatively poorly when music is coded and is mainly for speech.

10.8.1 Audio Codecs

10.8.1.1 AMR-NB

One of the most advanced speech coding standards today is the Adaptive Multi-Rate (AMR) speech codec, which was developed by the European Telecommunications Standards Institute (ETSI). It includes eight speech coding modes, with bit rates ranging from 4.75 to 12.2 Kbit/s. Some of the modes are the same as the earlier-defined voice telephony codec specified for other standards. For example, AMR at 12.2 Kbit/s is the same speech codec as the GSM Enhanced Full-Rate (EFR) codec. AMR is a mono type codec.

For best listening experience, AMR encoding should be used for content with ought complex, noisy, or critical music content. The limited audio bandwidth of 3.5 kHz means higher frequencies are not reproduced perfectly and are filtered with an anti-aliasing filter.
AMR works best on news and sports coverage as well as light popular music.

10.8.1.2 AMR-WB

AMR-WB represents state-of-the-art technology in low bit rate wideband speech coding. Just like AMR-NB, it is a multi-rate speech codec. AMR-WB technology uses nine bit rates between 6.6 and 23.85 Kbit/s at 16 kHz sampling rate, performing speech processing on 20 ms frames, each representing 320 speech samples. AMR-WB uses MIME type audio / amr-wb and file extension *.awb. AMR-WB payload format as well as MIME type is defined in RFC 3267[1]. AMR-WB was selected by the 3GPP in December 2000 and ITU-T in July 2001. The ITU-T AMR Wideband is now known as G.722.2. The 3GPP has several specifications available related to AMR-WB:

- 3GPP TS 26.190 [2]
- 3GPP TS 26.201 [3]
- 3GPP TS 26.174 [4]
- 3GPP TS 26.194 [5]

10.8.1.3 RealAudio Voice and RealAudio

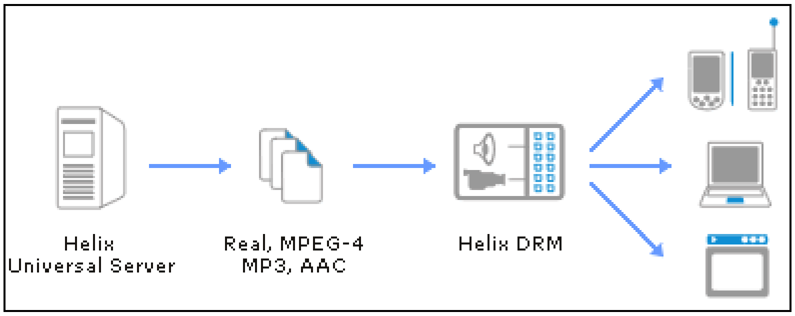


Figure 42 RealAudio and RealAudio Voicec

The Nokia S60 Series support the RealPlayer, Series 80, and Nokia 7710 devices supports RealAudio Voice, RealAudio 7 (known as RealAudio G2), and RealAudio 8 codecs.

- RealAudio 7
  - Internet audio codec used for many legacy contents served by streaming servers.
- RealVoice
  - ADPCM codec used for speech frequency ranges.
- RealAudio 8
  - Audio codec supporting various sampling and bit rates, starting from very low bit rates (3-4 Kbit/s).
- RealAudio 10
  - Marketing name for RealNetworks' latest collection of audio tools.
    - For bit rates lower than 128 Kbps
    - Uses RealAudio 8 codec
    - Employs MPEG-4 AAC codec for higher bitrates (see Section 10.8.1.5 and 10.8.1.6).
- RealAudio 10.5 (Recently released)
- [104]

10.8.1.4 MP3

MP3 formally known as MPEG-1 Audio Layer 3 compression format uses a number of techniques to reduce the size of audio data. The MP3 format is widely supported on some recent mobile devices like Nokia, as well as other digital audio players, CD players, and PC software.

10.8.1.5 AAC

AAC is also known as MPEG-4 Advanced Audio Coding (AAC) and is a successor format of MP3 for audio coding at medium to high bit rates. Comparing with MP3, AAC has a number of improvements in coding efficiency and frequency handling. Simplified it can be said that 96 Kbps AAC encoded audio is claimed to be at least the same or better quality than 128 Kbps MP3. The codec is used in mobile devices Ericsson among others.

10.8.1.6 AAC+ and eAAC+

AAC+ is know as aacPlus and is standardized by MPEG under the name HE-AAC (High Efficiency AAC). AAC+ combines AAC and Spectral Band Replication (SBR) together known as HE-AAC v1 or aacPlus v1. A 48 kbps stream of AAC+ is considered to have a higher quality than 128 kbps MP3. HE-AAC v2 adds support for Parametric Stereo (PS), which improves codec's performance at lower bit rates. This combination is also known as aacPlus and enhanced AAC+ (eAAC+).

10.9 Creation of a video clip

To create a video clip, the following is required;

One or more bit streams (for example, one video bit stream and, if present, an audio bit stream). A bit stream is the output of a video or an audio encoder. See Chapters 10.6, "Streaming Codecs and File Format," and 10.8, "Audio Coding," for more information about stream encoding.
It is also important to use one file format, that will pack the audio visual content into a file, synchronizes it, and provide access to it.

Figure 43 shows a simple illustration of how to create a video stream.

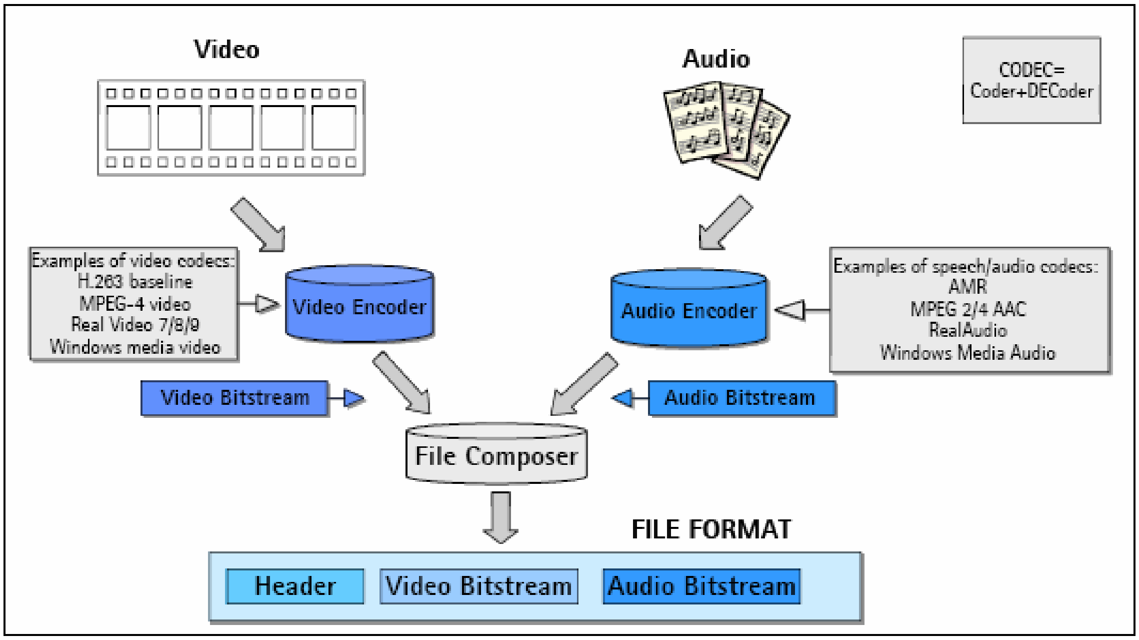


Figure 43 Basic Video Stream Production

Basic procedure in video stream creation:

1. Select any commonly used video editing tool, such as QuickTime, Adobe, etc., and create the video content.

2. Export the content to a compressed format such as MPEG-4, RealMedia or 3GPP and compress the content.

3. Here the 3GPP file may be "hinted" in preparation for the streaming session.

4. The server extracts the media content from the file and sends it to the network, according to the appropriate payload formats.

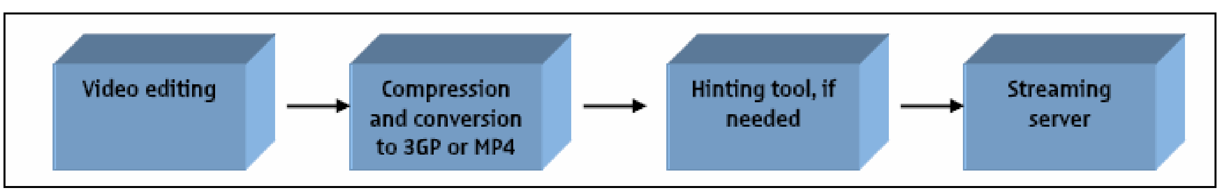


Figure 44 Editing, Compression and Streaming

Depending on the implementations, some of the steps described above can be merged. A server may be capable of doing media transport protocol layer packetization on the fly, which means that it may be able to understand a non-hinted file and simply stream it. Encoding and hinting can also be merged in the same element. This is the case, for example, in the Helix Mobile Producer (for 3GPP media) and the Apple QuickTime Pro tool (version 6.3 and above) with 3GPP support. Since there's no common hinting language for streaming server files yet, all hinting tools today address one specific server.

10.10 Hardware Tools for Video Content Creation and Streaming

Figure 45 Hardware

1. PC: Intel Pentium 4, 2.4 Ghz, 256 MB RAM
2. Server: Powerfull servers to store the content on
3. Capture card: Viewcast Osprey series (Envivio)
4. Mobile Phone
5. Minimum 1 Professional DV Camera (professional!)
6. MPEG-2 Encoder & Multiplexer and DVB-H/T/S modulators (example of equipment is taken from ABE Italy)
7. Digital & Analogue Transposer & Gap-Filler
8. Amplifiers
9. Backup: Portable Hard disks with today's maximum storage capacity (minimum 500G).
10. Envivio Supported OS
11. Windows 2000/ Windows Server 2003/ Windows XP Pro
12. Recommended Hardware ([105])

10.9 Software Tools for Video Content Creation

There are many different manufacturers offering Broadcasting Software these days. As the hype for Mobile TV is increasing rapidly, everybody wants a piece of cake. The basic thing for streaming is to have the right software and set up a server for streaming.

- Helix Universal Server from RealNetworks. This server supports a variety of formats, including RealMedia, Windows Media, Quicktime and MPEG-4.
  - Offers RealProducer Basic a free limited streamer for Windows [107].

- Apple Quicktime Streaming Server, supporting a few formats including MPEG-4 and 3GPP.

- VideoLAN & VideoLAN Client, supports the MPEG-1, MPEG-2 and MPEG-4 standards. Free open source solution [106].

- Envivio 4Caster Mobile Software, supports ISMA Profile 0, 3GPP v2,5,6, DVB-H and outputs 3GPP2, 3GPP5, 3GPP6 single-rate and multi-rate, MPEG-4 file using RTP/UDP and IP unicast and multicasting.

VLC media player (initially VideoLAN Client) would probably be better to try out, as it can be used as a client to receive decode and display MPEG streams under multiple operating systems. VLC can also be used as a server to stream MPEG-1, MPEG-2 and MPEG-4 / DivX files, DVDs and live videos on the network in unicast or multicast.

VLS (VideoLAN Server), can stream MPEG-1, MPEG-2 and MPEG-4 files, DVDs, digital satellite channels, digital terrestrial television channels and live videos on the network in unicast or multicast.

Most of the VLS functionality can now be found in the much better VLC program. Usage of VLC instead of VLS is advised.
This diagram illustrates the VideoLAN solution:

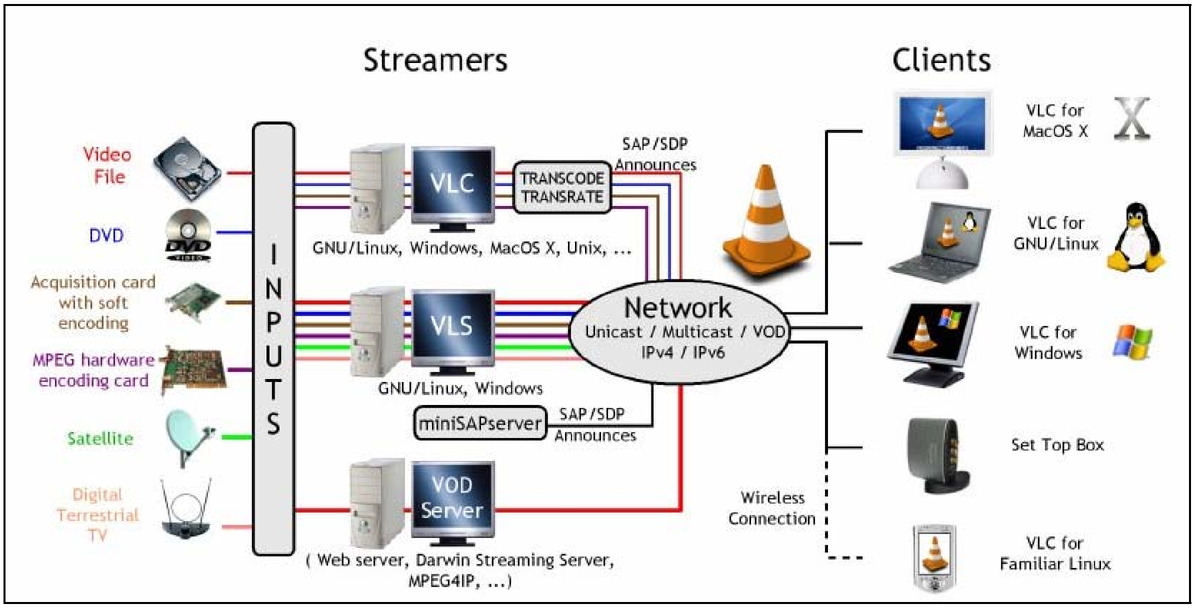


Figure 46 VideoLAN [106]

VLAN testing could not be made in time and the live stream setup using the Quicktime Streaming Server did not work as planned. The preferred system to use is Envivio due to the possibility of real-time MPEG-4 AVC encoding while streaming out to the users.

11 Live Content Production Concicerations

Production of Live Broadcasting Content is to take place at the events themselves and be professionally recorded by the service approved Content Recorder Team. The equipment for broadcasting has to e setup in such ways that the stage where the actual event is happening can be fully recorded and in such way that the stage decorations and lights settings as well as VJ (Video Jokey) animations do not interfere with a good picture.

In the future, the service plans to offer Live Events in 3D view.

12 System Design

This chapter will describe the service and the functions that the system is suppose to provide as well as its overall behavior. It is important to notice that the broadcast can be done in three ways over three different broadcasting networks, namely 3G, DAB and DVB-H. The design proposition tries to take all three broadcasting networks into consideration and leaves up to the buyer of the service to implement the desired one. The thesis will not implement any of the features described in this chapter as the main focus is the services provided.

The design is done using Microsoft Visio and some of the figures may not have the correct connection lines 'theoretical', since this is the first time I am using this program and lack of time prevents me from using it correctly. However I do believe the interactions are explained
good enough with text under each figure. Please take note that there is room for small errors due to the wrong usage of Visio.

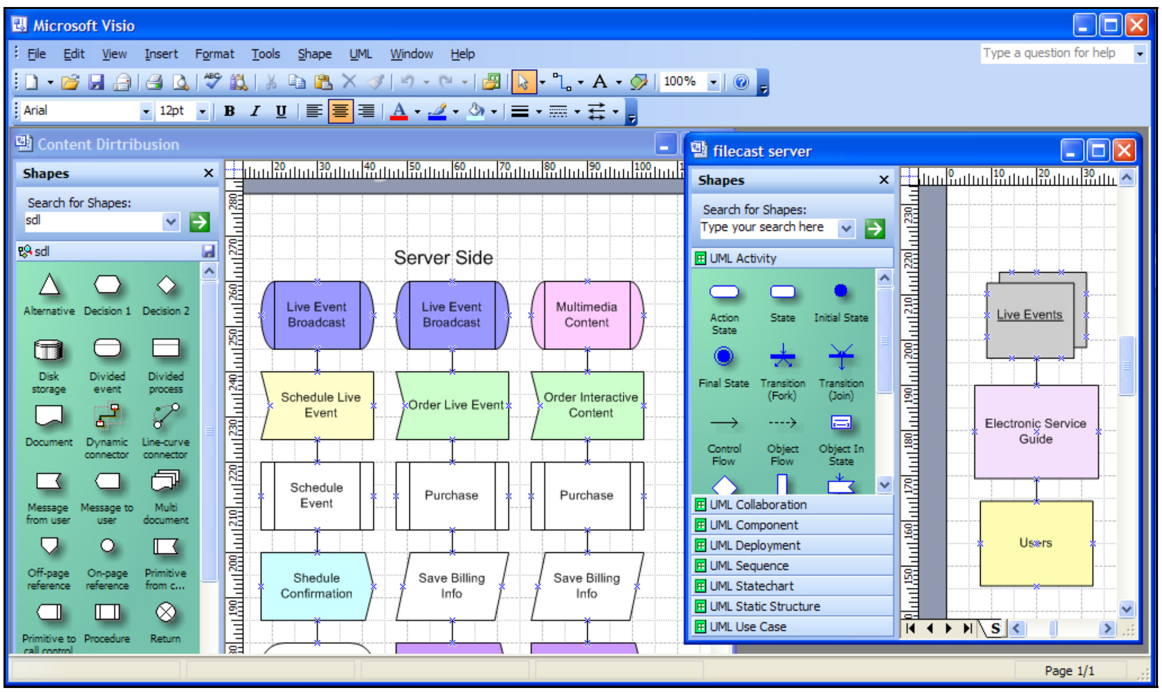


Figure 47 Microsoft Visio

12.1 The System Architecture

The following architecture is proposed for the system and is based on the same solution as the Open Air Nokia Mobile Broadcast Solution.

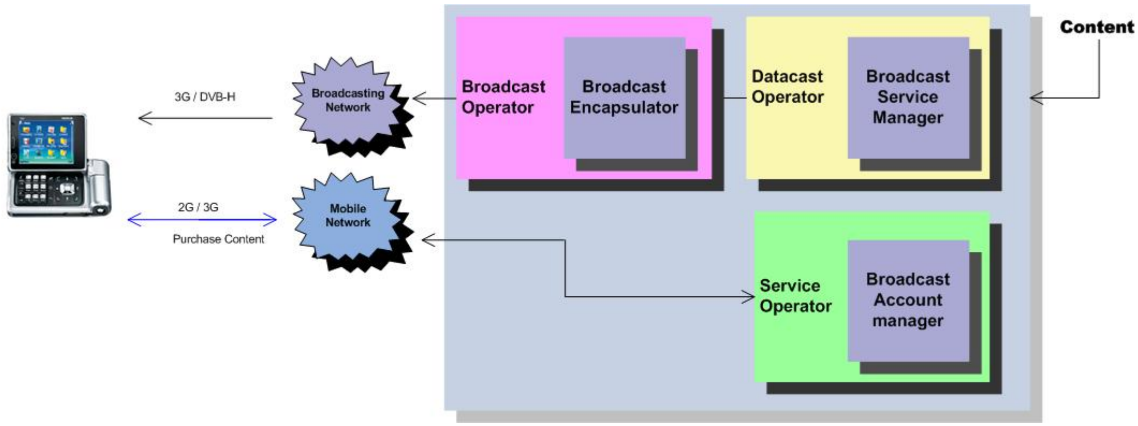


Figure 48 FUMES System Architecture

The Content Provider offer content to the Broadcasting parties. The Service Operator handles all user orders for content, takes care of the charging and approves sales. The Datacast Operator makes sure tat the content which is encoded on the Providers side gets forwarded to the Broadcast Operator who encapsulates the content using IP Encapsulation and send it out to the customer through the Broadcast Network.

12.2 The User Perspective

The following Figure illustrates the system from a user's perspective

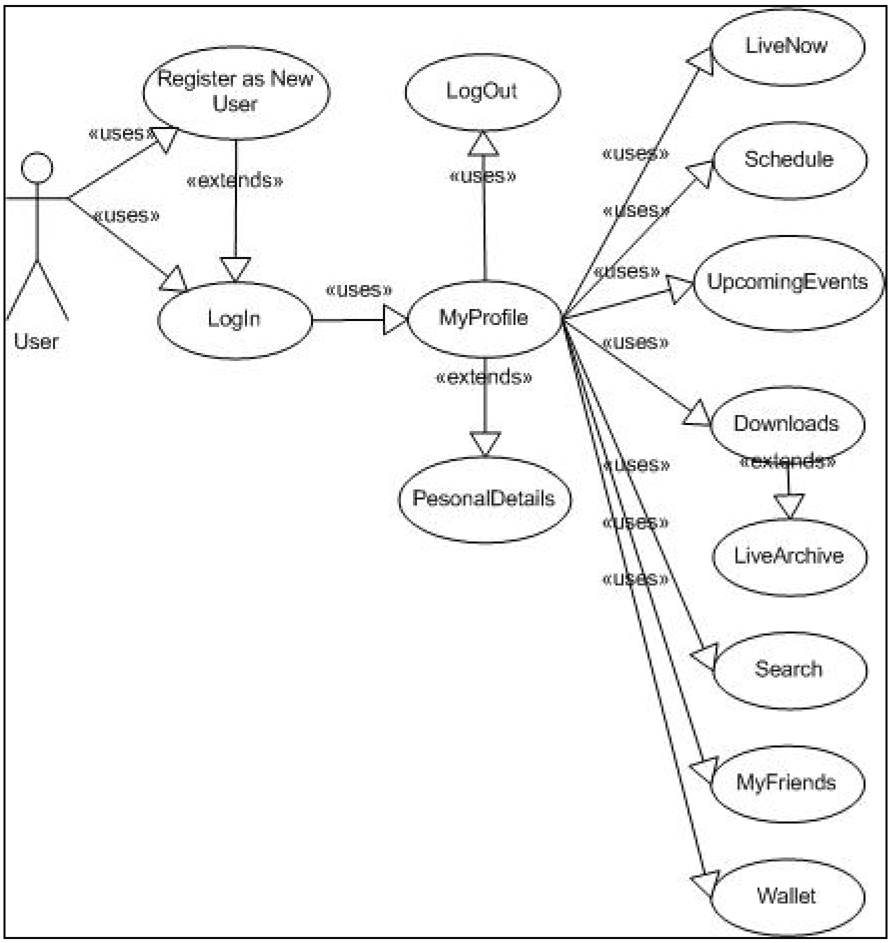


Figure 49 User Perspective

The user logs on or registers first and then logs on to the service which take the user to his Profile. From the profile MyProfile, the user can choose among several services available, listed in the requirements chapter.

12.2.1 LiveNow

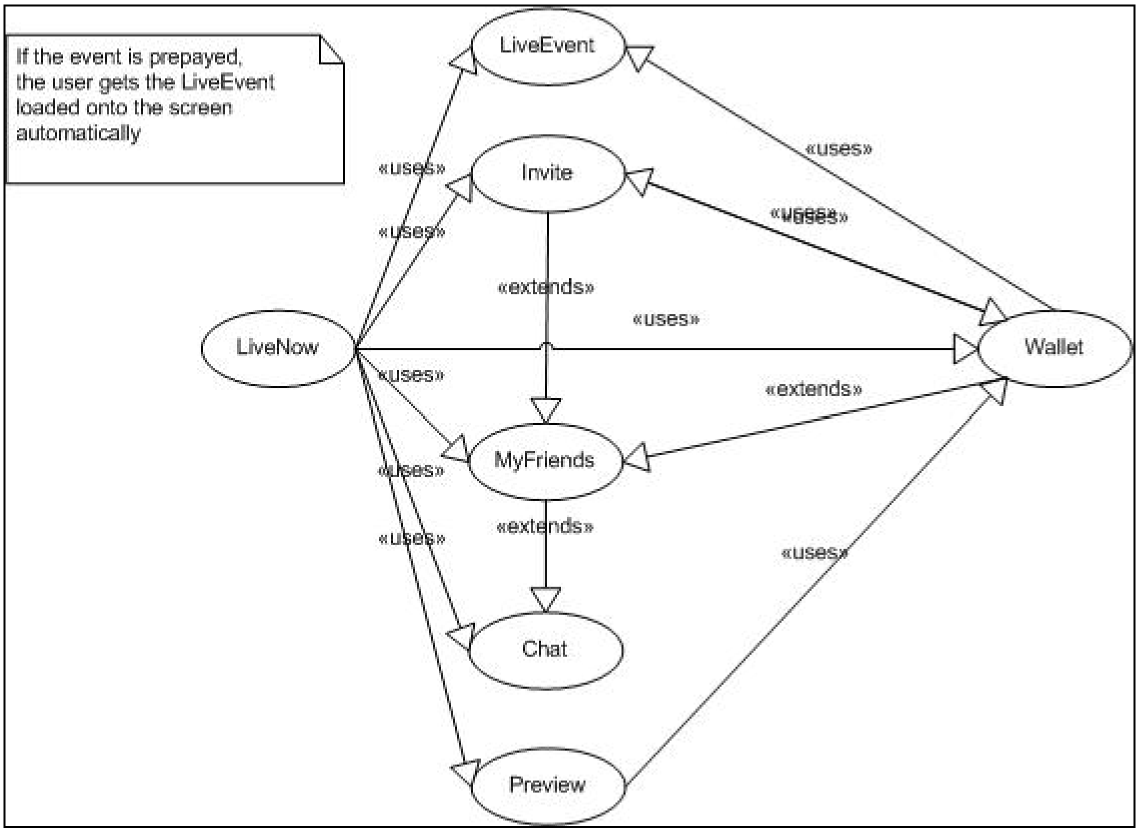


Figure 50 LiveNow

Once the user chooses the LiveNow menu, he will automatically get the LiveEvent loaded into his screen, if he has prepaid for the service. Else a preview of the show will be shown for lets say maximum 10seconds and the user will be asked if he wants to buy viewing of the event, which loads the Wallet menu where the user can pay for the event. While watching the event, the user can chat with his friends and invite them to join the viewing using Invite. The Invite menu allows the user to either send a normal invitation where the friend will be shown a preview and then asked to buy the event, or the user can choose to prepay the event for the friend, For every successful invite which results in a new user buying an event, the inviting part gets 'credits' which are added to the users wallet and can be used for purchase of content or if the user wants, transferred to his friends etc. All 'live event' previews are linked to a users SIM card allowing the user only to see it once (else friends would use the invites to watch the show by multiple invite requests). Purchased content can be viewed at any time after the live event is passed and can be found in the users Archives.

12.2.2 Schedule

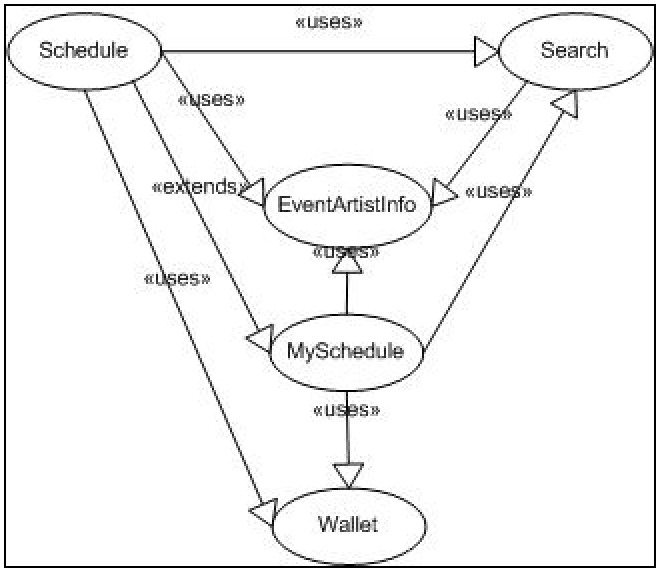


Figure 51 Schedule

The Schedule lists all events scheduled of a day or week (user sets the preferences) and allows the user to add events to the users personal schedule MySchedule. The user can also use the Search menu to look for artists or events, as well as specify in the MySchedule options if he wishes to subscribe to certain events/artists etc. Once an event is added to MyProfile, the user can choose to prepay it and set alarm of notification for when he wishes to be reminded about the streaming. The alarm can be added to the Calendar function in the phone and notify the user by sound or by sending an SMS. If the user has not prepaid the event and not set any alarm, the event gets added to the user's archives and marked unpaid. The user can at a later time go into the archive and buy the event if he likes or remove it from the list. The user can also cancel to view a prepaid event; the money is then translated into Credits and added to the users Credits account for later purchases. Cancellations are only refunded as Credits.

12.2.3 UpcomingEvents

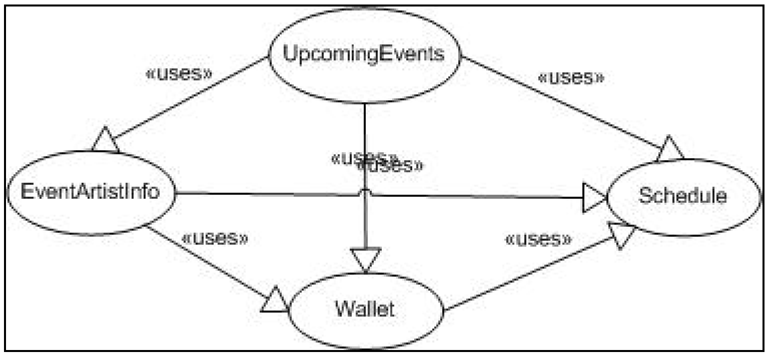


Figure 52 UpcomingEvents

The UpcomingEvents lists all upcoming events that will be streamed live.

The user sets the preferences for how he wants the list to look, for example he can choose to list 10 or 20 events at a time sorted by artist, date or music genre. The service will only list live events maximum 3 moths ahead so that the users check in more often to see what is coming up. When choosing an artist or event, the information about the artist / event gets listed by the EventArtistInfo and gives the user the possibility to add the event to the Schedule first and purchase later, or purchase the live broadcast or an actual ticket to the event.

12.2.4 Downloads

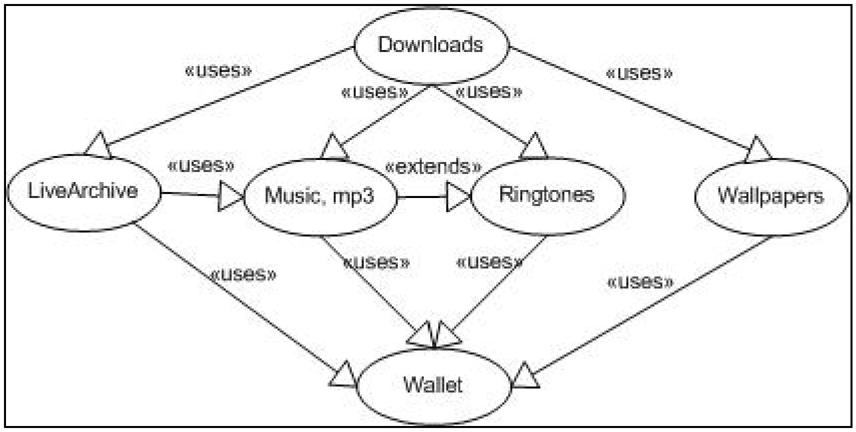


Figure 53 Downloads

The download menu works like an online shop and lets the user purchase, tracks, ringtones, DJ sets, wallpapers and past events from the LiveArchive. After the content is chosen and put in the shopping chart, the user gets directed to the Wallet to make the payment. Live events that have been purchased before can be viewed for free and will have price zero.

12.2.5 Search

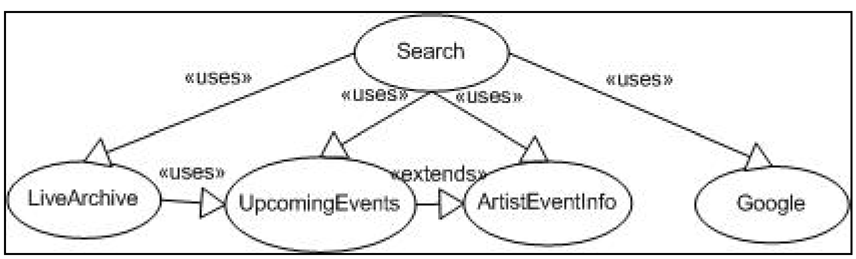


Figure 54 Search

The Search function searches for information within the system using it's own searching tool. Search uses additionally Google.

12.2.6 MyFriends

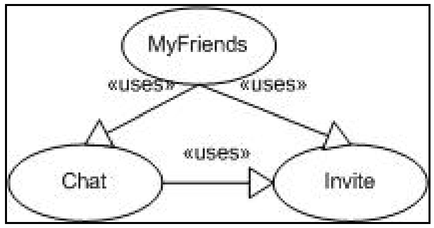


Figure 55 MyFriends

The MyFriends menu works similar to MSN Today and keeps track of all the users Friends. The friends are added to the users MyFriend list in two ways;
Users that are not members of the service, receive a personalized or standard invitation from the user who is identified with full name and mobile number. The invitation is sent by ordinary

Once the new member is added to the service (membership is free), the 'friend' gets added to the users MyFriends list. If a user does not wish to accept the invitation by ignoring it for example or using the decline link, the user gets notified of the decline.

Users already in the system can use the WAP link in the SMS to refuse to be added to someone's FriendLists as well as invitations to view live events and chat sessions.

MyFriends menu is also used to interact with other user and artists through chatting similar to IRC.

12.2.7 Wallet

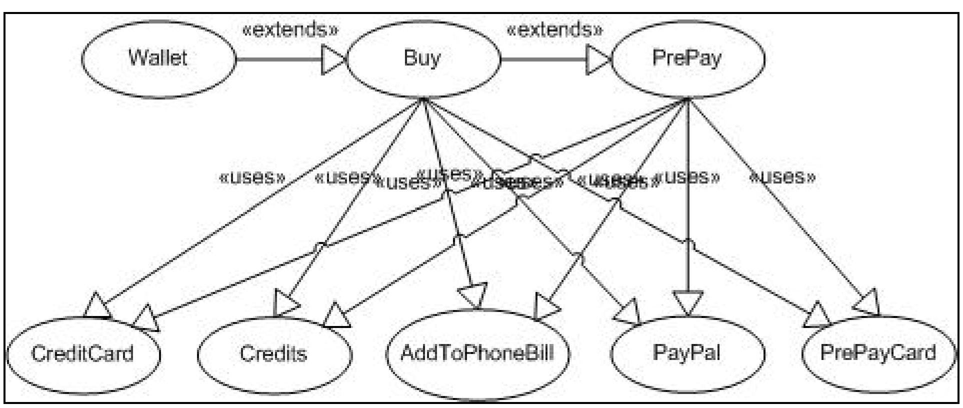


Figure 56 Wallet

Wallet is the payment feature of the system and allows the user to pay for content in several ways. The user can pay with Credit Card (the service allows for storage of the CC details on the users personal profile and considers it to be safe due to the personal profile being linked to the users SIM card along with good encryption), use of credits (obtained through invitations which purchased events and events the user has purchased himself). The purchase can also be added to the user's phone bill. In such cases the system checks with the mobile operator the user uses to see if the users credit has not gone over the limit and makes sure that the purchase also keeps the phone bill at least $1/10^{th}$ from the max credit limit (to allow the user still use the phone for a while after the purchase). Some users use PrePay cards in their Mobile phones. The service checks the credit left on the card and if enough credit, notifies the user of how much credit will be left on the card after the purchase, which the user must confirm he agrees to.

12.3 Content Upload

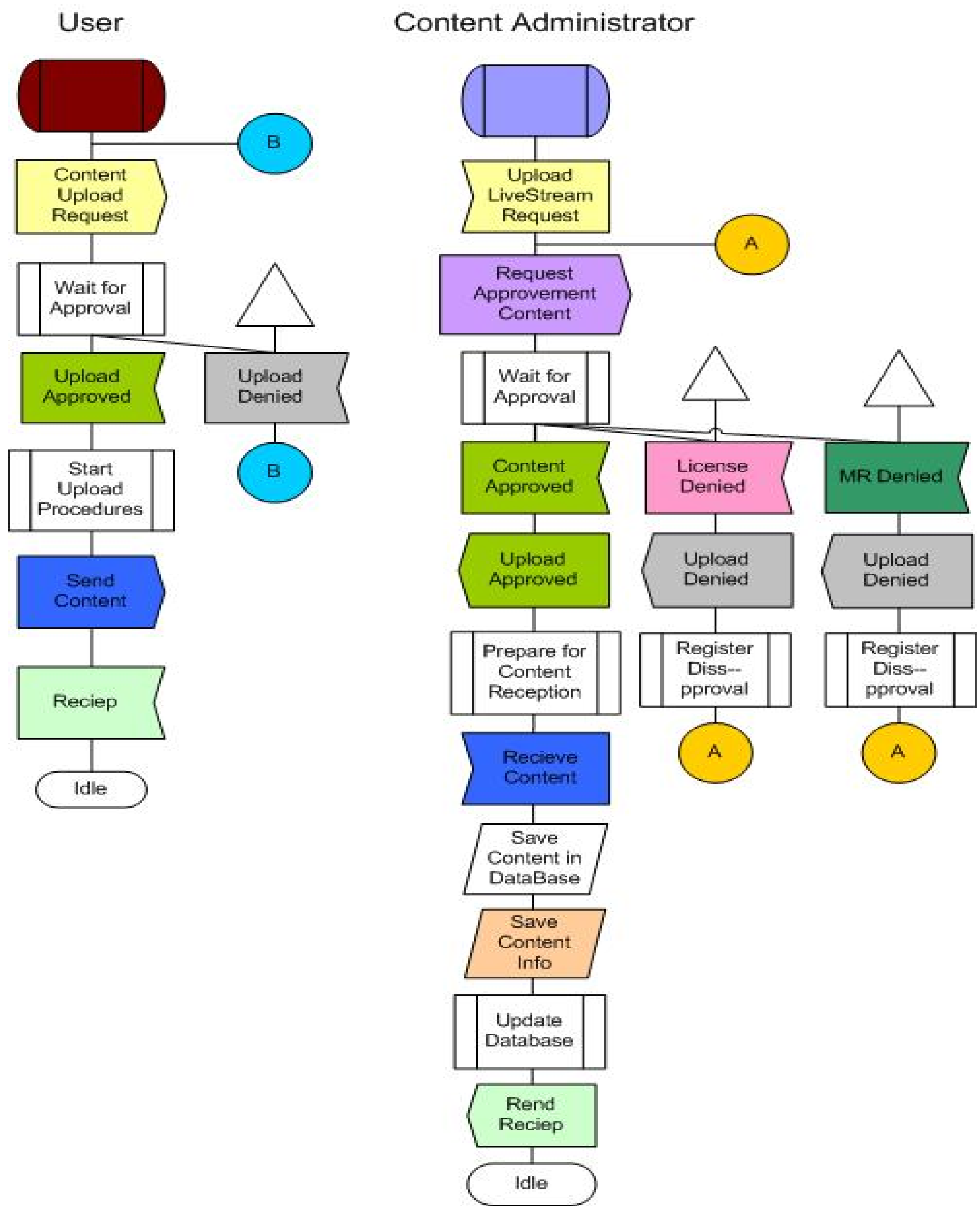


Figure 57 Content Upload part 1

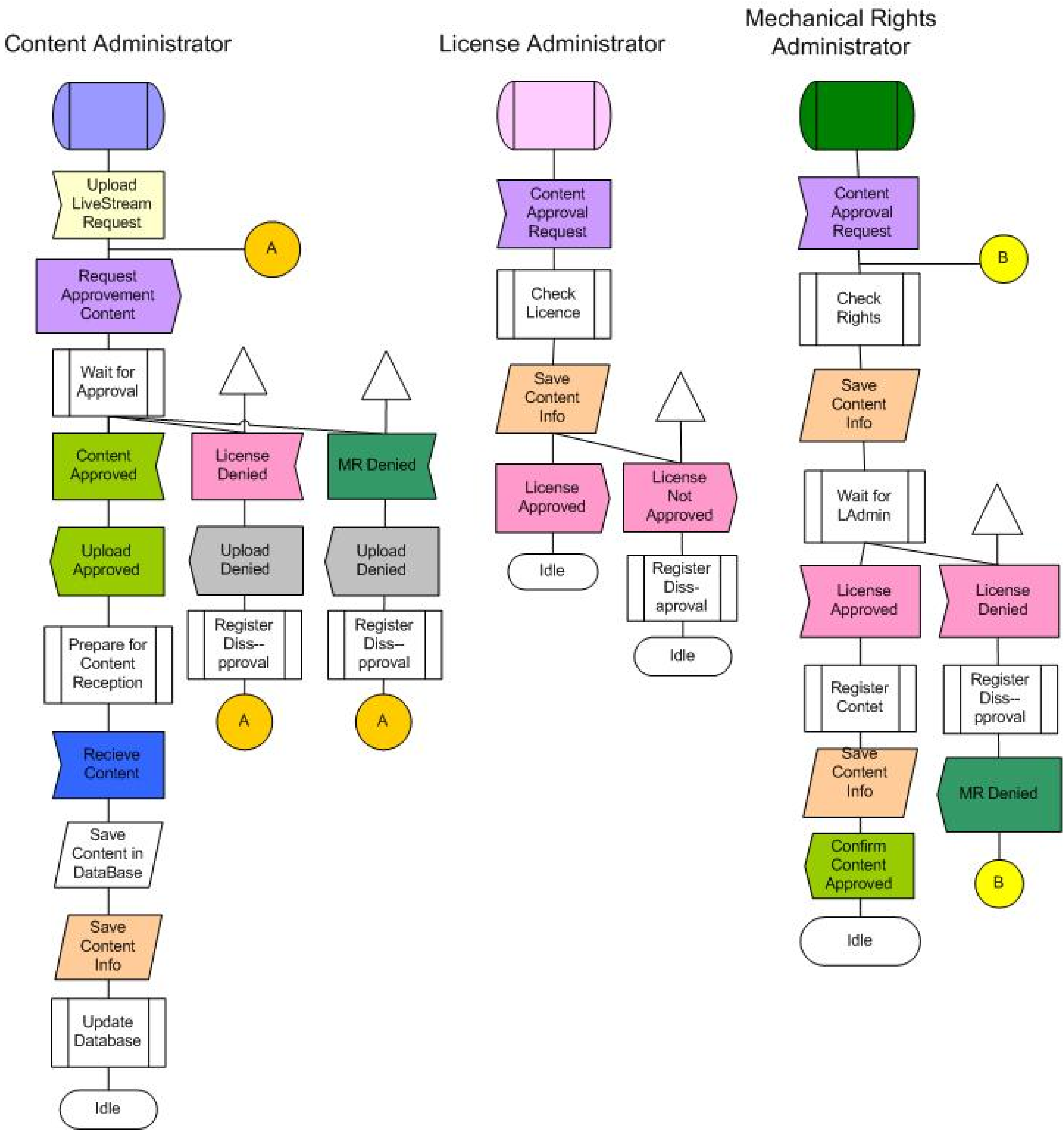


Figure 58 Content Upload part 2

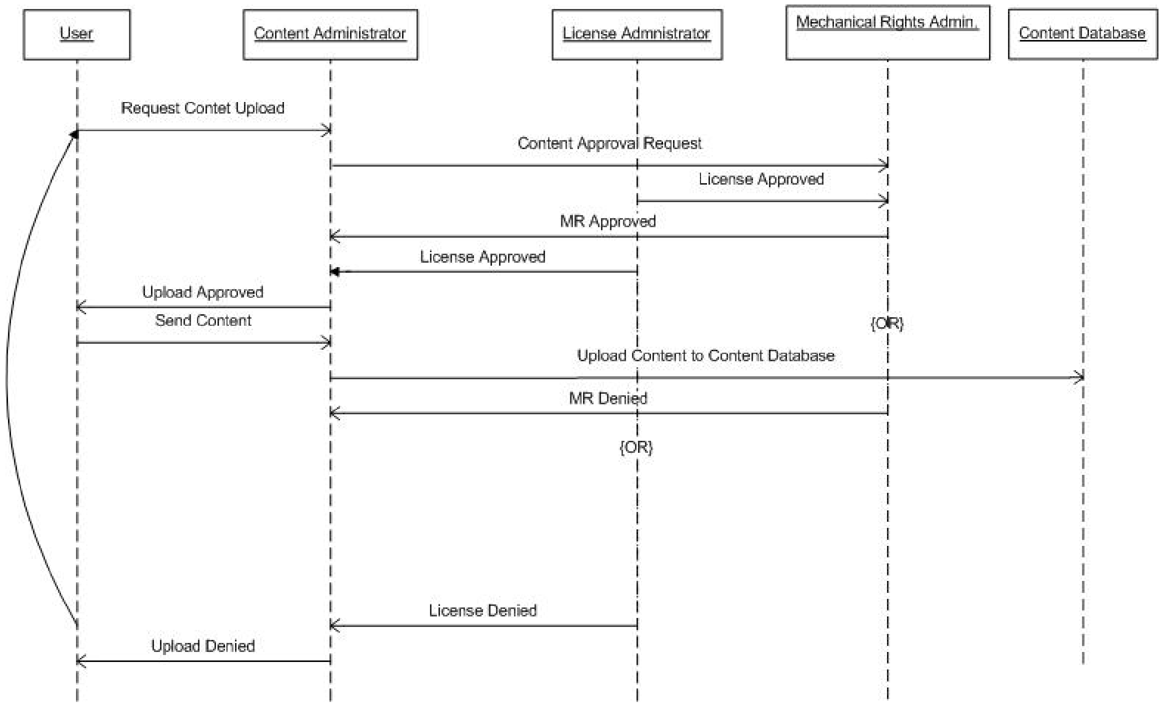


Figure 59 Content Upload Sequence Diagram

1. The User srends a request (including all necessary details about the user and the content and their relations) to upload content to the CAdmin.

2. CAdmin forwards the Request details to LAdmin and MRAdmin
    a. If LAdmin disapproves of the Licenses request, CAdmin refuses the user to upload the content.

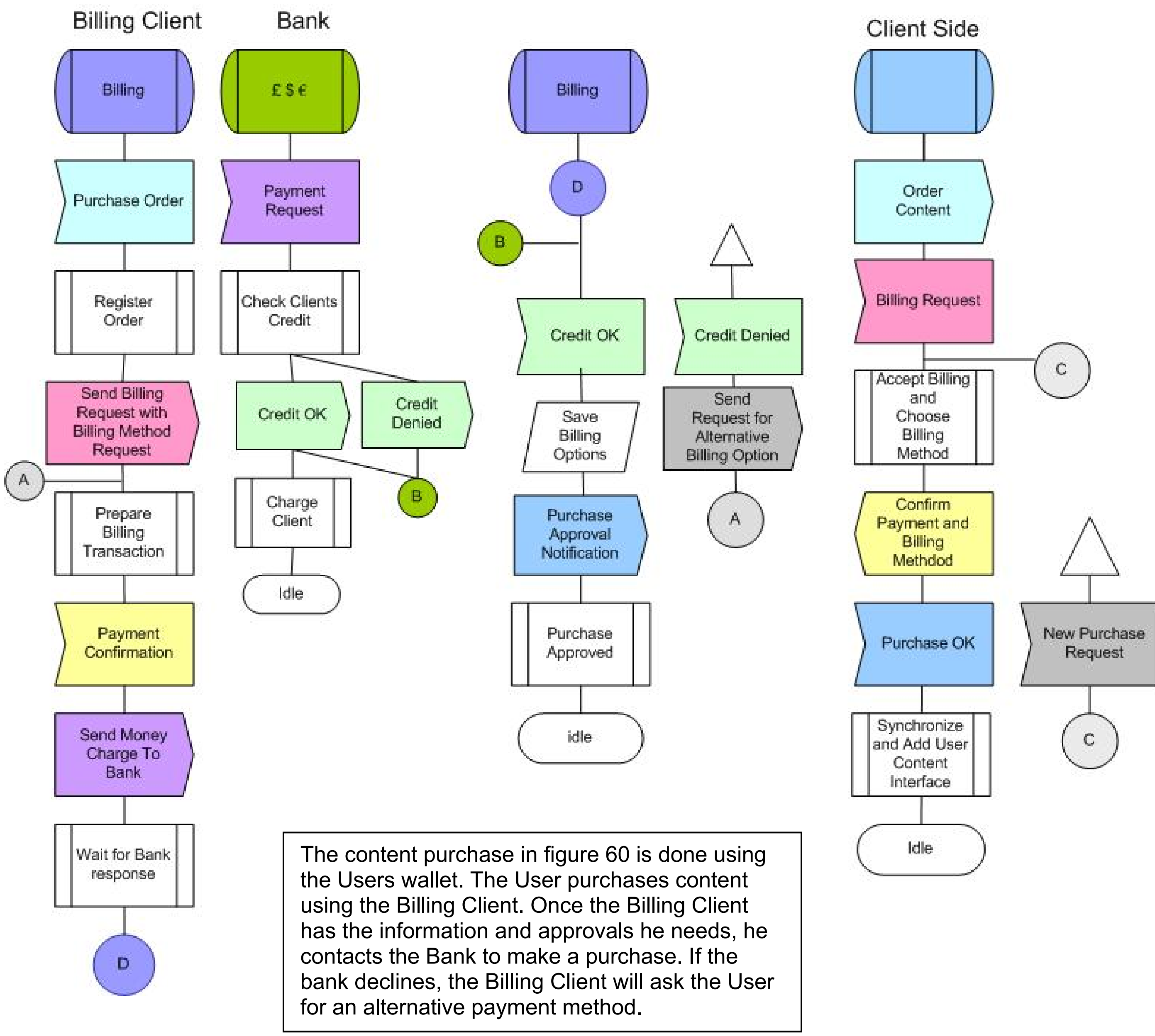
    b. If LAdmin approves the content, he sends a request to MRAdmin to approve Mechanical Rights.

    c. If MRAdmin approves the content, he sends approval response directly to the CAdmin who sends an upload approval to the User.

    d. If MRAdmin disapproves the content, he sends disapproval response directly to the CAdmin who sends upload denied to the User.

3. If the request is successful, the user uploads the content to the CAdmin who adds it to the content database and issues a recipe to the user.

12.4 Content Purchase



The content purchase in figure 60 is done using the Users wallet. The User purchases content using the Billing Client. Once the Billing Client has the information and approvals he needs, he contacts the Bank to make a purchase. If the bank declines, the Billing Client will ask the User for an alternative payment method.

Figure 60 Content Purchase

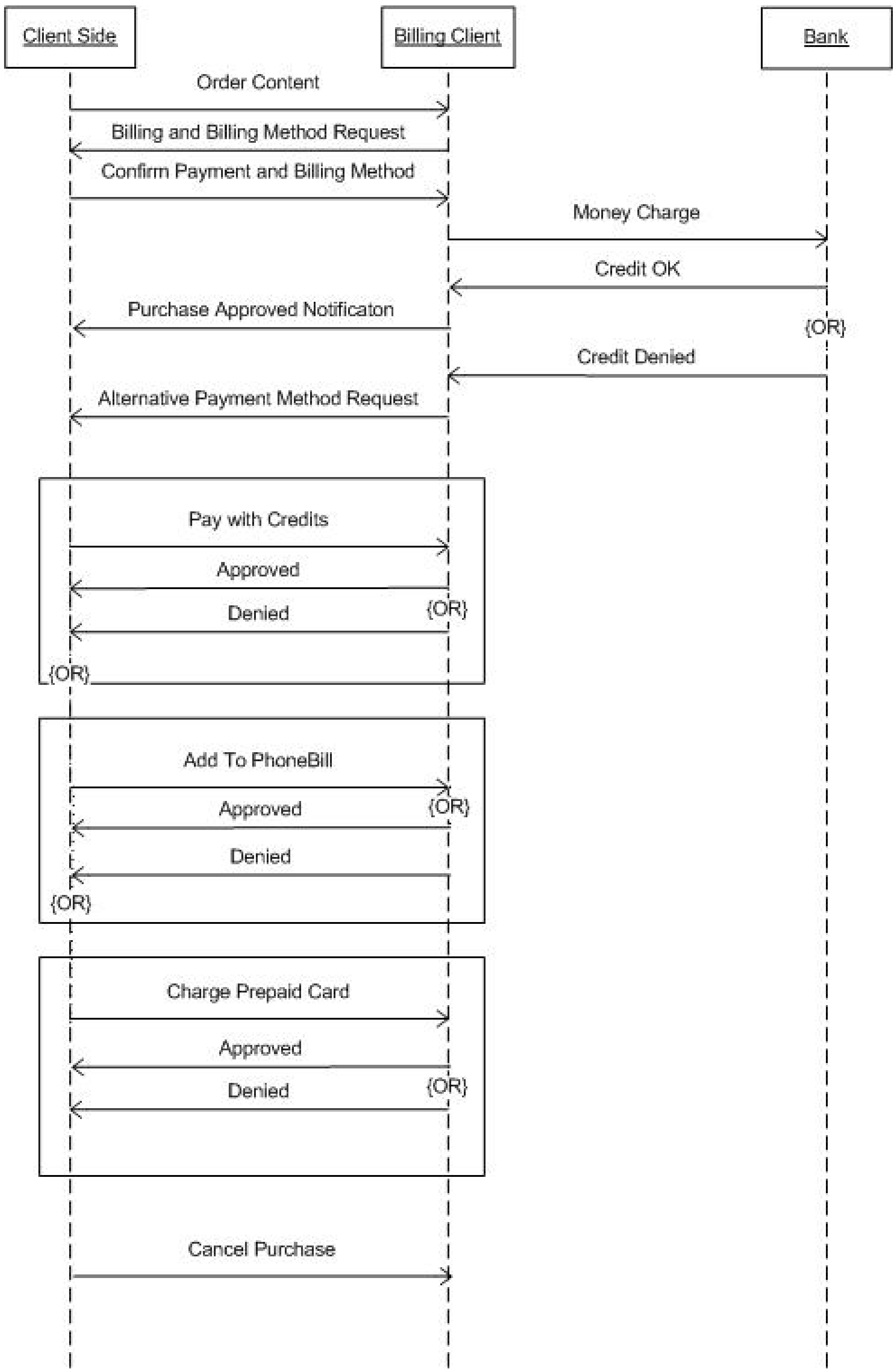


Figure 61 Content Purchase Sequence Diagram

1. The User sends a content order to the Billing Client who then requests to perform the billing and asks for the payment method to be used.

2. If the Bank approves the Credit due to purchase with a Credit Card or PayPal, purchase is made.

3. If the Bank refuses Credit, The Billing Client requests alternative paying methods;
   a. Credits; if the User has enough Credits on his account, he can make the payment

   b. Mobile Phone Bill; the User can add the purchase to his phone bill.
      i. The Billing Client will in such case contact the Network Operator the User is using and request to add the amount to the Users phone bill.
      ii. If the User is too close to his allowed “credit” with the Network Operator, the request will be denied
      iii. Else the purchase will be added to the bill

   c. Prepaid Phone Card; the user can charge the purchase on the prepaid card he uses.

      i. In such cases the Billing Client tries to charge the prepaid card.
      ii. If the amount of money left after a purchase is for example less then 20 NOK, the purchase will not be void.
      iii.

4. The user can also *cancel* the transaction upon alternative billing method request. This is *not* added to Figure 60, only the Sequence diagram in Figure 61.

Content Protection
Except with DRM , the content can be additionally protected by being bound to the user thru his SIM card. Every SIM card is unique and personal. By binding the content to the SIM card with additional information about its owner, the content will not be viewable for others that the owner of the SIM card.

12.5 Content Distribution

The following simplified flowchart illustrates how the Content is distributed in the system. The User (Client Side) can either schedule or order content. The system distributes the content based on the purchase made by the User.

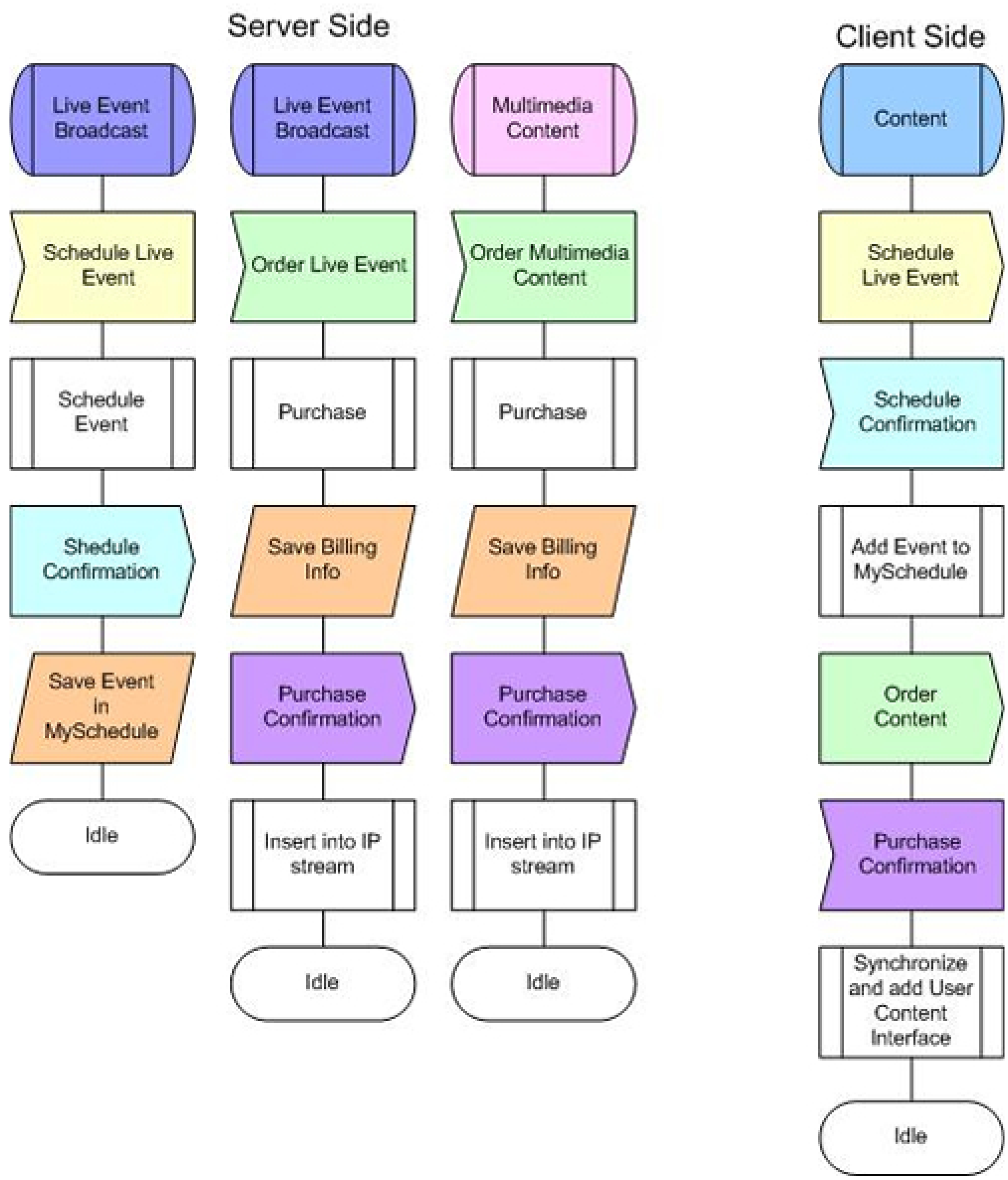


Figure 62 Content Distribution

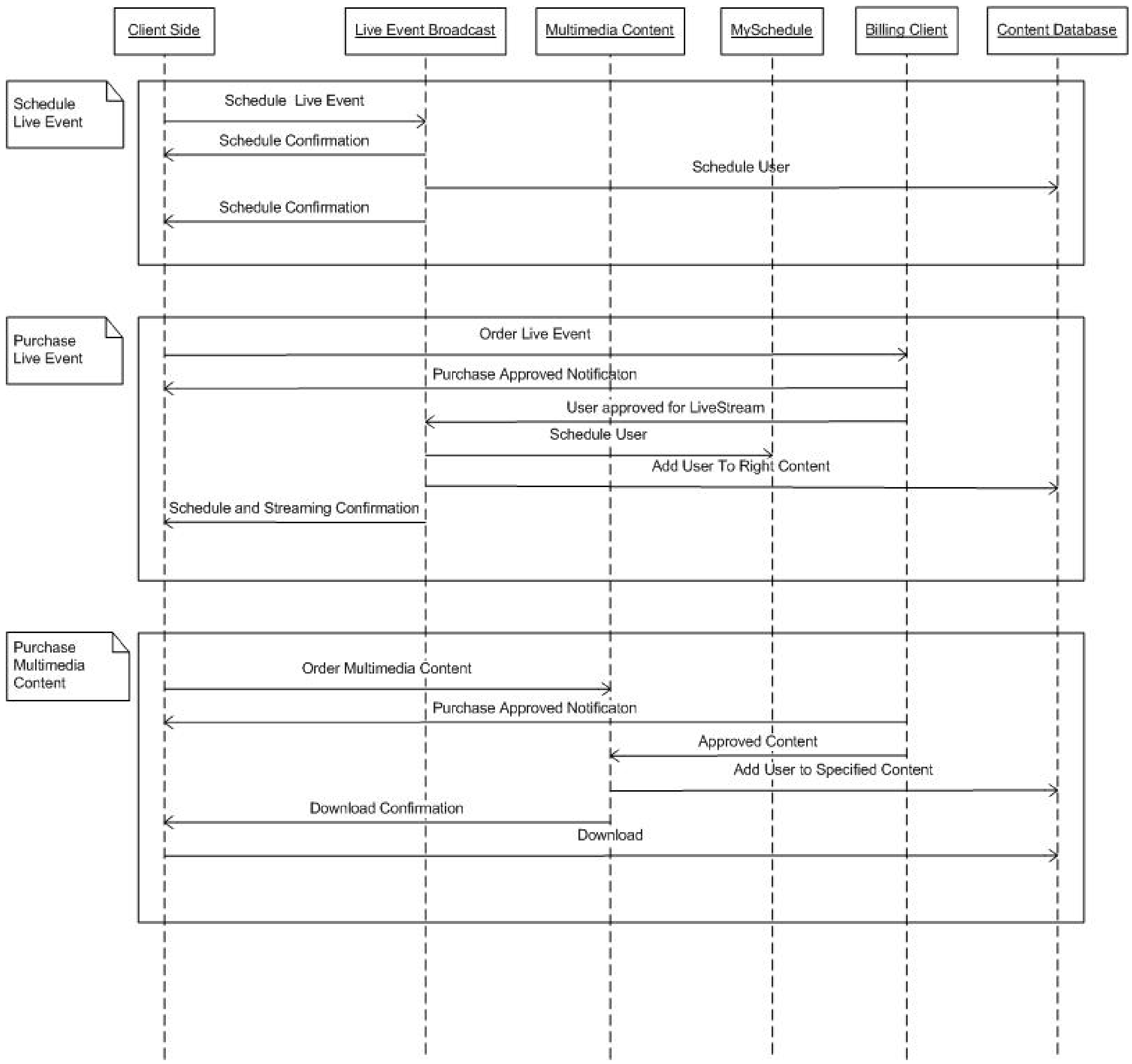


Figure 63 Content Distribution Sequence Diagram

Content is simply distributed as shown in the figure above; The User can schedule or purchase events and multimedia content. Purchased events become scheduled automatically.

12.6 The Subsystem Architecture

The following Subsystem Architecture is proposed for future system implementations.

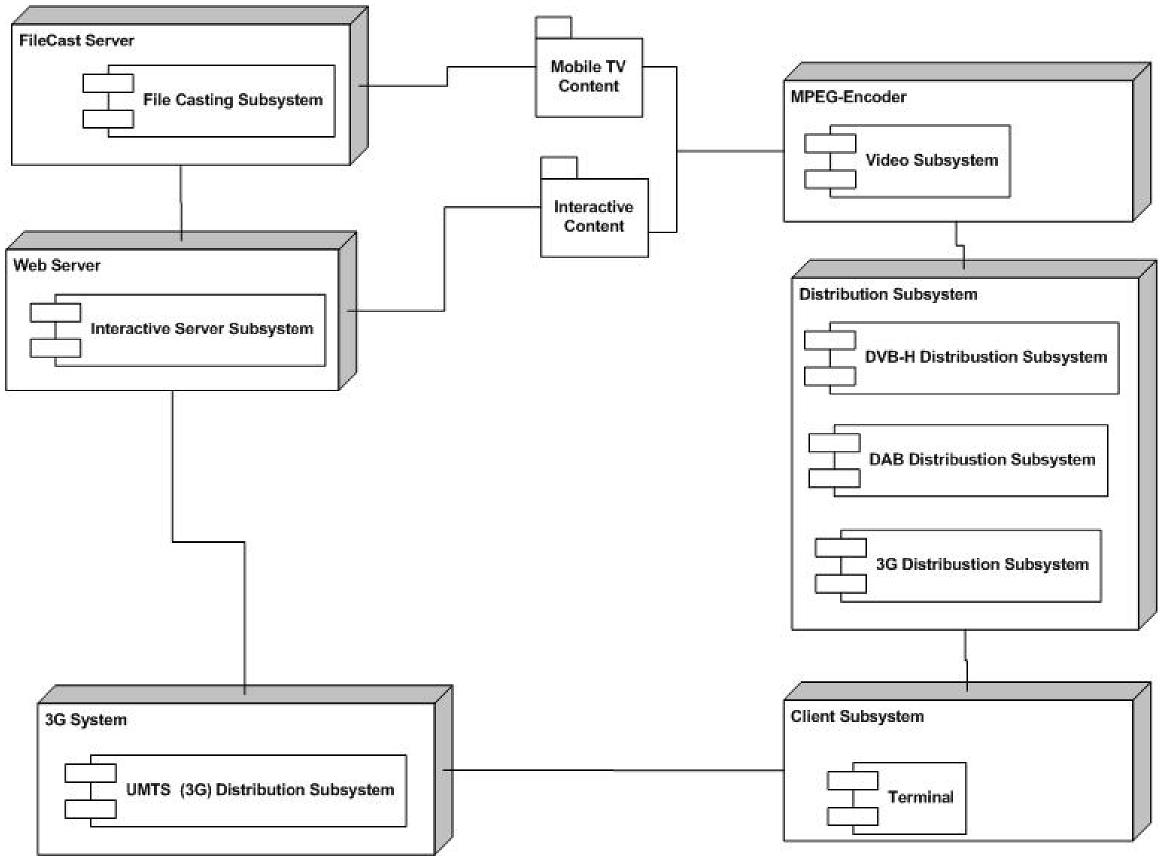


Figure 64 High Level Overall Design of the System divided into subsystems

To get an idea of how the system should be implemented in the future, it is important to know how the system is build up and divide it into smaller subsystems. The overall system is made of 6 Subsystems, each explained in the following subchapters.

12.6.1 The Video Subsystem

The Video Subsystem can offer one or more live video streams preferably MPEG-4 or 3GPP format. (Live content is recorded using high quality DV cameras). The stream is then combined with interactive content such as statistics and information about the content, poll results, and other information which the content provider can take advantage of later.
The stream is then encoded using MPEG-4 and then transported to the Network using the UDP/RTP/RTCS transport protocols. The stream is also recorded on a server as well a

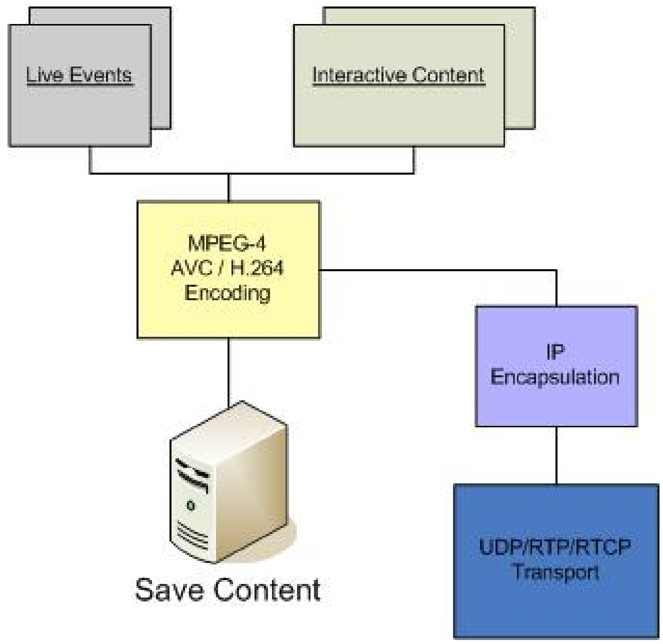
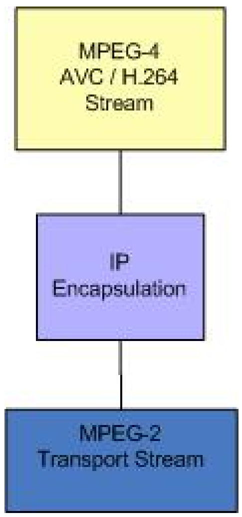


Figure 65 Video Subsystem

The MPEG-4 encoding can be done in both software as well as hardware. Software encoding requires great CPU usage and can take time, which leads to delay problems. Alternatively the encoding can be done in the hardware using for example the MPEG-4 AVC encoder in Envivio which enables encoding while streaming the content out for broadcasting.

12.6.2 The Distribution Subsystem



The Distribution subsystem takes care of all three possible distribution types; current 3G and DAB as well as the currently on trial and future distribution solution DVB-H.

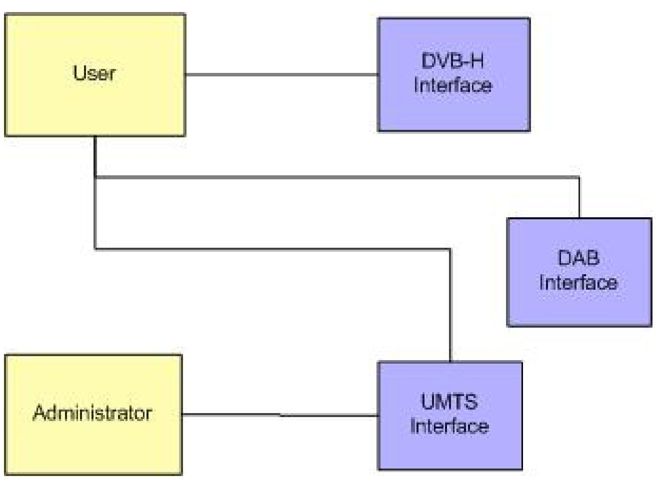
The Distribution subsystem makes sure that the content is encapsulated into IP packets and then MPEG-2 transport stream that delivers the content to the Mobile devices. The encapsulation takes place at the Broadcast Operators end.

It is also possible to add interactive content to the packed streams, if there is free capacity. This can be useful for the Interactive System to for example send software updates.

Interactive Content can be added to the Transport Stream after it has been Encapsulated using an IP Inserter and in that way producing a new stream which includes the Interactive Content.

Figure 66 Distribution Subsystem

12.6.3 Client Subsystem Interaction



The client side of the subsystem must contain both DVB-H and UMTS interfaces. The Live broadcastings are transmitted over the DVB-H interface while all interactive content and interaction between the user and content provider is done over the UMTS interface. This is done to keep track of all the information sent from an individual user so it can be used for other purposes like statistics on the user and his use of the service.

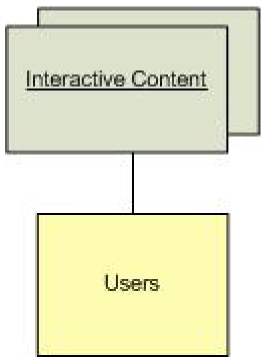
Figure 67 Client Subsystem Interaction

The administrator in the subsystem is not needed for the multimedia channel scenarios, but is good to have incased the system will be expanded to be used for other purposes like seminars, lectures, other educational purposes. In those cases the Administrator could be a professor or lecturer who would like to update a webpage, or interact with students through message-boards etc.

Note that the User Interface of the system must be as simple and easy to use as possible with “idiot proof” functionality. It should be simple and easy to use, like in Figure X below;

Figure 68 “ Idiot proof ” functionality

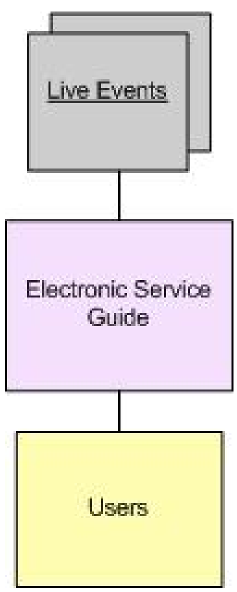
12.6.4 The Interactive Server Subsystem



The Interactive Server Subsystem takes care of all interactive activity from the users as well as interaction between them. The Subsystem also uses the Content Database to provide users with

Figure 69 Interactive Server Subsystem

12.6.5 The Filecasting Subsystem



The Filecasting Subsystem takes care of all multimedia activity between the service provider and the users. It also handles multimedia interaction between the users and makes sure that the Electronic Service Guide is always offering the users new services updates and as well pursuing the users into purchasing new

Figure 70 Filecasting Subsystem

12.6.6 The UMTS Distribution System

The remaining UMTS Distribution Subsystems is self explaining; the 3G system interacts with the users through their terminals and is used for all interactive content as well as multimedia streaming.

13 System Functionality

The Figure on this page show how I imagine the overall architecture can to look like.

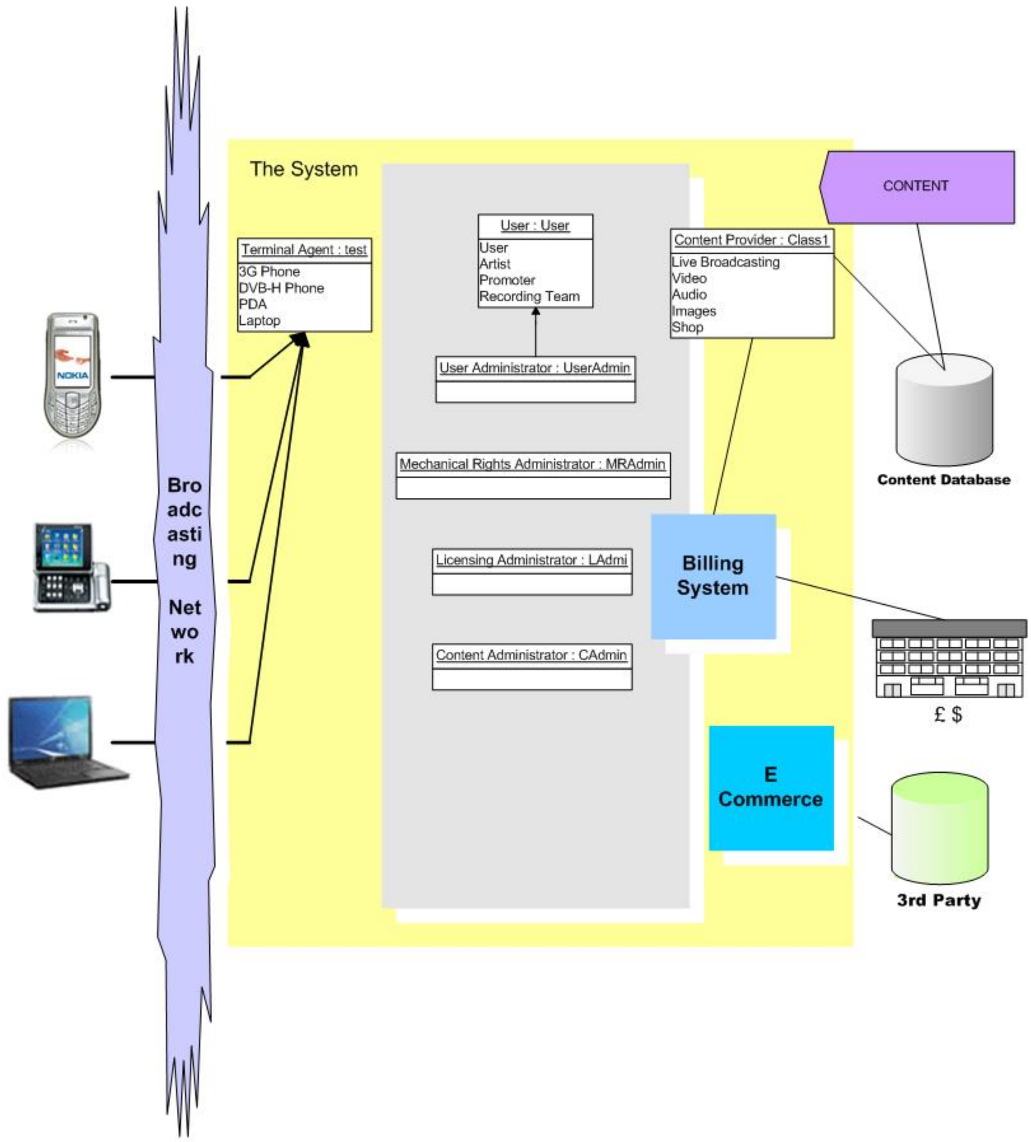


Figure 71System Functionality

14 Conclusion

Only a few years back, corporations like Cisco Systems put together elaborate streaming media systems for their own use. In contrast, during the current era of company budget-slashing, multimedia isn't exactly a mission-critical IT priority for most organizations. At the same time though, demands for streaming video and audio are more rampant than ever inside many departments. When a streaming media project rears its head, network managers and other administrators usually get called upon for advice, if not for actual implementation.

The American Department of Defense for example, is now launching a streaming video application among 20,000 military health care pros, including medical directors and clinical consultants. The application will supply 35 hours of training about smallpox vaccinations.

On the government side, streaming media is being driven partly by the Federal Streaming Alliance, a new American initiative for sharing streaming content among federal agencies.

Against today's overall backdrop of financial uncertainty, end users' interest in streaming media stands out vividly. During panel discussions at the recent Streaming Media show in New York City, a number of attendees pointed to streaming media deployments at their workplaces, either imminent or already underway. On a consumer level, recent statistics show a tremendous increase in online shopping activity, not only in the United States, but even in Norway. There is an increasing demand for online services and it is a common fact that people, especially women, think of computers as big and ugly machines that they like to keep hidden away out of sight of their daily life. These same people all carry a mobile phone that is tuned to their own taste. With the ever increasing power and possibilities of mobile phone devices, slowly but surely these little computers are bound to take over most of the online features that require a stationary computer at this time.

With this development, which is going on at this very time, the platform that is required to launch the concept that I describe in detail throughout this thesis, will be largely established in a matter of no time. With the technology at hand, you might expect that someone would already be working on their own FUMES concept. For many purposes however, live streaming doesn’t add anything at all. In fact, Video on Demand (VoD) makes more sense for many applications. With VoD, you can pre-position a video on everyone's desk, so they can view it at their own convenience. Therefore we can assume that the main international focus where it comes to streaming media, is on VoD applications for corporate training and briefing purposes for some time to come.

With the above in mind it should be concluded that the FUMES concept is a great business opportunity in many ways. It is undeniable that this new method of distribution will be the next leap in mobile media entertainment and with corporate focus still on VoD applications, the chances are high that whoever jumps on FUMES now, will be the first one on the market. I don't need to point out that being the first offers major advantages to a business perspective.

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Appendix

DRM Architecture

Messerges and Dabbish [109] have made a proposition of where the DRM layer could be added in the Operative System of a device used for DRM content. The generic operating system is extended with DRM and security capabilities by adding a DRM manager, trusted application agents, and security agents and hardware. The OS with extensions comprise a trusted system. In their approach, only applications that access DRM-protected content have to be aware of the new DRM extensions. They also state the following example to support their proposition; “when an application tries to access a particular file, a header may indicate that this file is protected. In which case, the application can use the DRM extensions to open the file and render the data. The application will access these extended system services through an Application Programming Interface (API) that is augmented to provide additional DRM-related services”. Figure 69 shows that these DRM extensions include a DRM manager, security hardware, and a suite of trusted application and security agents. They will also run in the same “privileged mode” as the OS and will have access to system data and resources making applications run in “user mode” and have limited access to system data and resources. Example of an application can be a video player. At the GUI level, these applications do not need to worry about DRM content other than to report on the status of the license. Lower-level components of the applications will invoke DRM extensions to process protected content under control of privileged-mode extensions due to the content being decrypted and unprotected.

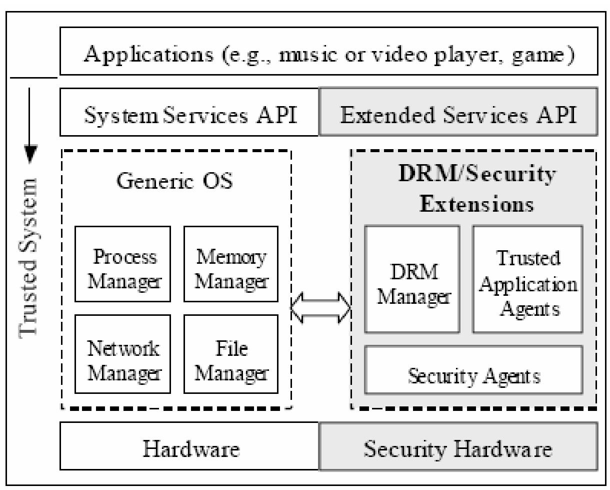

Figure 72 GUI Application [109]

The DRM manager manages the functions in DRM and works with security agents to authenticate licenses and content, parse and enforce usage rules, access a secure DRM database, and provide decrypted content to a trusted application agent.

5.1.1 Actors and Functional Entities

In DRM, an actor is defined as an external entity that is involved in carrying out use cases. There are many possible actors in a DRM system, such as content owners, developers and distributors, content consumers, via network service providers, billing service providers and manufacturers of network equipment and devices. Different actors play different roles in the system, all depending on the deployment scenario.
In the OMA DRM architecture are used to describe specific roles in the DRM system, making it possible to decompose the tasks involved in digital rights management, separately from what actors perform each task in a certain deployment.
Functional entities are not represented by any physical nodes, since they are logical and hence can be implemented by the same or different physical nodes and be operated by the same or different actors, depending on the configuration. Depending on the required functionality in each deployment setting, different deployments may incorporate some or all of the functional entities.

The following functional entities have been identified in the architecture by DRM:

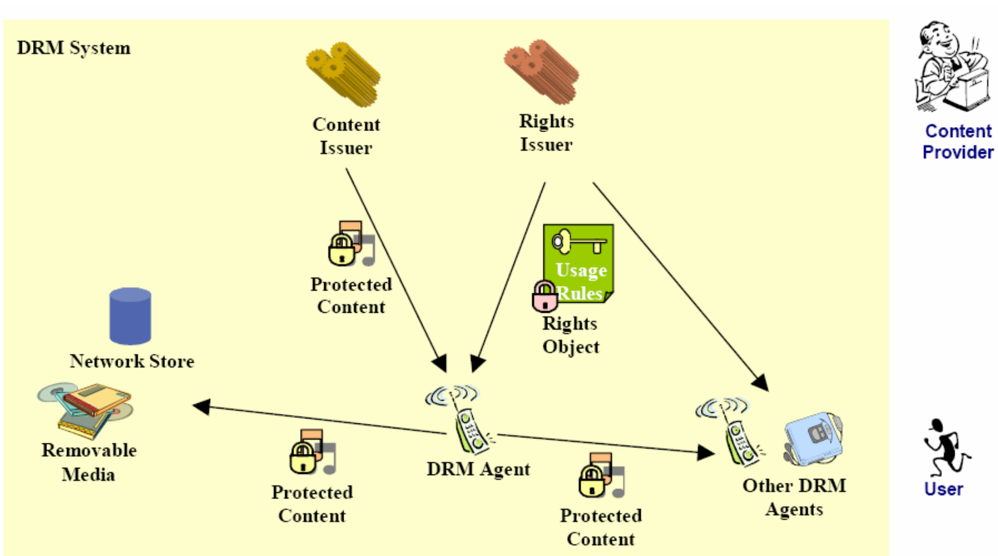

Figure 73 DRM System

1. **DRM Agent**
   A DRM Agent consists of a trusted entity in a device, which is responsible for enforcing permissions and constraints associated with DRM Content, such as controlling access to DRM Content, etc.

2. **Content Issuer**
   The content issuer is an entity that delivers DRM Content to the DRM Agent. The format of the content as well as the way the DRM Content is transported from a content issuer to a DRM Agent, using different ways of transport mechanisms is defined by OMA DRM. The content issuer can either do the actual packaging of DRM Content him/herself, or he/she may receive pre-packaged content from some other source.

3. **Rights Issuer**
   The rights issuer is an entity assigning permissions and constraints to DRM Content, and generating Rights Objects. Rights Objects are XML documents expressing permissions and constraints associated with a piece of DRM Content and govern how DRM Content may be used. They may only be used as specified by the Rights Object. *DRM Content cannot be used withought an associated Rights Object*.

4. **User**
   A user is the human user of DRM Content and can only access DRM Content through a DRM Agent.

5. **Off-device Storage**
   Of-device storage is for example a PC, removable disk that is placed in a secure place. This may be used for example for backup purposes or to free up memory in a device. Since DRM Content is inherently secure, this is a splendid way of storing it. Similarly, Rights Objects that only contain stateless permissions may be stored off-device.

To prevent unauthorized access to any content sent, the content before any content is to be sent, it is decrypted to protect it from unauthorized access. A content issuer delivers DRM Content, and a rights issuer generates a Rights Object. The content issuer and rights issuer embody roles in the system. Depending on deployment they may be provided by the same or different actors, and implemented by the same or different network nodes.

An XML document also called Rights Object, specifies permissions and constraints linked to a certain piece of DRM Content and governs how the DRM Content may be used. DRM Content can only be used with an associated Rights Object, and may only be
used according to the permissions and constraints specified in a Rights Object.

OMA DRM makes a logical separation of DRM Content from Rights Objects. DRM Content and Rights Objects can be requested separately or together, and can be delivered separately or at the same time. For example, a user buys a past concert and receives DRM Content and a Rights Object in the same transaction. Later, if the Rights Object expires, the user can go back and acquire a new Rights Object, withought having to download the DRM Content again.

Rights Objects associated with DRM Content have to be enforced at the point of consumption. This is modeled in the OMA DRM specifications by the introduction of a DRM Agent. The DRM Agent embodies a trusted component of a device, responsible for enforcing permissions and constraints for DRM Content on the device, controlling access to DRM Content on the device, and so on.

All Rights Objects are bound to a specific DRM Agent cryptographically, meaning that only a certain DRM Agent can access the corresponding Right Object that belongs to it. DRM Content can only be accessed with a valid Rights Object, so it can be freely distributed. Because otthis, for example, super distribution can be enabled, as users can freely pass DRM Content between them. Access of DRM Content on any new device requires request of a new Rights Object delivered to a DRM Agent on that device.

A Rights Object may optionally be bound to a group of DRM Agents. This is called a Domain and is only possible if this is supported by the rights issuers. DRM Content and Rights Objects that are distributed to a domain, can be shared and accessed offline on all DRM Agents that belong to that domain. In other words a user may purchase a concert or event for use on both her mobile phone and her PDA.

Specifications of the OMA DRM define the format and protection mechanism for DRM Content, as well as the expression language and the protection mechanism for the Rights Object. The specifications also define the security model for key encryption management and how DRM Content and Rights Objects may be transported to devices using a range of transport mechanisms, including pull (HTTP Pull, OMA Download), push (WAP Push, MMS) and streaming. Interaction between network entities like between rights issuer and content issuer, is out of scope.

DRM Content

OMA DRM is designed to be flexible and support a wide variety of different business and usage models for the DRM content and it's distribution, among these are the Basic Pull Model, Push of DRM Content (in both the content is packaged and delivered as one item), Streaming of DRM content, Domains, Backups, Export and more. In this thesis we will look only at the ones that are most relevant for the multimedia service.

1 Streaming of DRM Content

When streaming DRM content, the content is packetised and delivered as a stream which is protected by encryption. OMA DRM does not specify formats for encrypted streams as this is done by other standards bodies and content streams can be protected with encryption schemes which are different from those specified by OMA for Download. Once the stream has been encrypted a Rights Object is generated, the encryption key(s) to access the encrypted stream is put in the Rights Object just like a Content Encryption Key would, and the Rights Object is then bound to a DRM Agent. Withought the Rights Object, the protected stream cannot be accessed.

2 Domains

As already mentioned the Rights Objects and content encryption keys are bounded to a specific DRM Agent. Domains expand this notion, by allowing a rights issuer to bind rights and content encryption keys to a group of DRM Agents. By doing so, the users can share DRM Content off-line between all DRM Agents that belong to the same domain. This offers the possibility for a rights issuer to provide new services such as enabling users to access DRM Content from several devices that they own. The Domain concept also offers support for Unconnected Devices making it possible to purchase DRM Content and rights via one device such as a PDA and later use the content on another device such as portable player withought wide area network connectivity.
The rights issuer controls and decides if they wish to provide services based on domains, as well as decide what DRM Agents are allowed to join a domain.

3 Backup

All DRM Content can be stored safely on any storage media and kept safely for backup. The content is stored in encrypted form and can only be accessed by a particular target DRM Agent or in the case of Domains a group of DRM Agents using an associated Rights Object. Also Rights Objects can be stored for backup purposes, but the replay cache mechanisms makes sure that the Rights Objects which contain full state permissions can not be reinstalled as the result of restoring a backed-up Rights Objects. All Rights Objects are protected and can only be accessed by the intended DRM Agent, or a group of DRM Agents in a certain Domain. Even if a Rights Object is stored off-device, it will still only allow the intended DRM Agent or Agents to access associated DRM Content.

2.4 Super Distribution

DRM Content can be safely copied and transferred to other DRM Agents. Users that would like to send DRM content to each other have to go through the Rights Issuer using a link in the DRM Content package, to acquire a Rights Object. The rights issuer controls whether to release a new Rights Object or not to the new DRM Agent. If all is OK, the Rights Object is released and the content is transferred between the users through the Agents.

5 DRM Format

DRM has two different Content Format profiles;

1. Discrete Media Profile (DFC)

DFC is used to package and protect Discrete Media like for example ring tones or applications by being wrapped around the content before it becomes encrypted as a single object completely identical to the contents internal structure and layout. This specification defines the Discrete Media format based on the types of the ISO base media file format [ISO14496-12] hence the DCF format maintains the extensible nature of the ISO format and keeps overhead minimal by using the ISO principles. A Device defined in [DRM-v2] MUST support the DCF format as defined in this specification. In addition, version 1 DCF as defined in [DRMCF-v1] MAY be supported.

2. Continuous Media Profile (PDCF)

PDCF stands for Packetized DCF (PDCF) and protects Continuous Media like Audio and Video in a separate profile. Applications that read and parse Continuous Media are meant to work on the file on a packet-by-packet basis, so to make the playback of protected Continuous Media easier, the storage format needs to be structured so that the packets are protected individually. This type of structured packetization is also required in order to stream Continuous Media. All OMA DRM compliant streaming servers *must* be able to understand the Content Format' structure, this in order to break the content into headers and packets that can be delivered to a client that understands the Content Format.

6 Discrete Media Profile (DCF)

DRM Content Format (DCF) is a secure content package for encrypted content and has its own MIME content type. In addition to the encrypted content it also contains content description such as the original content type, vendor, version, etc. It also contains the rights issuer URI which is a location where a Rights Object may be obtained among others. The extra information is not encrypted and may be presented to the user before a Rights Object is retrieved.

Since a DCF is inherently secure, it can be transported using any transport protocol, just like in an HTTP response or in an MMS message. DCF can also be stored for back-up on any kind of storage, such as removable media or a networked PC and can be copied and sent to another DRM Agent, where a Rights Object may be acquired for use on the receiving device. This technique is called Super distribution.

To unlock DRM Content inside a DCF, en encryption key which can be found within a Rights Object is needed. Meaning it is impossible to access DRM Content withought a Rights Object and the content can only be used as specified in a Rights Object.

OMA DRM also includes a verifying mechanism which allows a DRM Agent to verify the integrity of a DCF to protect against unauthorized modification of the content.

For content download, OMA DRM V2.0 defines the DRM Content Format (DCF), which is specified in [OMA-DCF]. The DCF format, copied from the specification, is shown in Figure 1. A DCF object is encrypted with a Content Encryption Key (CEK) using symmetric key mechanisms. The DRM agent at the terminal, after receiving a DCF object, is supposed to acquire the Rights Object (RO) from the RightsIssuer (specified in the RightsIssuerURL in the Common Headers). To use only the content format in MBMS, no RO is required. Therefore it needs to be specified how to indicate that the object is MBMS protected, in which case the DRM agent should not attempt to acquire the RO, but rather should use the corresponding MTK for decryption.

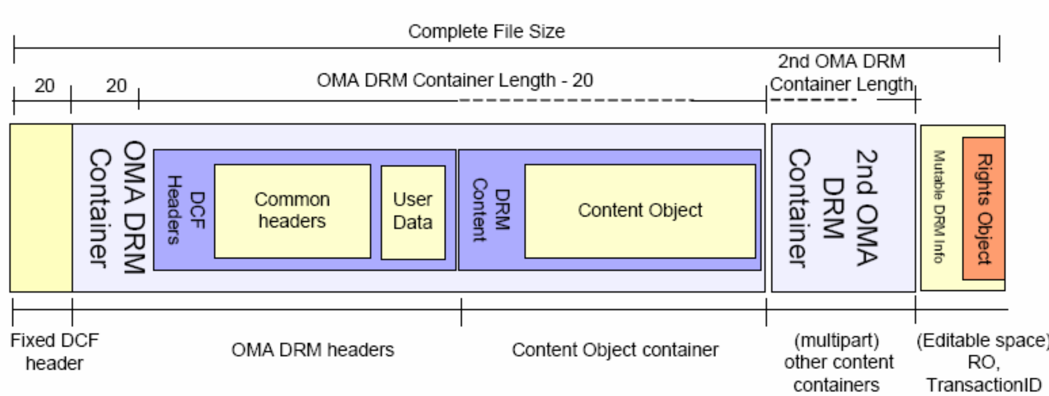

Figure 74 DCF Structure [111]

For integrity protection in DRM V2.0, a hash of the object is embedded in the RO. For MBMS, RO is not required. Integrity protection of the object can be provided by the XML-signature of the FDT as suggested in [S3-040557] or by including the object hash, calculated according to the OMA DRM V2.0 specification [OMA-DRM], in the FDT.

7 PDCF Format

The Packetized DRM Content Format format is an instance of the ISO Base Media File Format [ISO14496-12] that supports encrypted media tracks. PCDF *must* use OMA DRM for key management as well as *must* include the OMA DRM data structures defined in the OMA-TS-DRM-DCF-V2_0-20060303-A specification. The 3GP format [TS26.244] and the 3G2 format [C.S0050] are examples of ISO Base Media File Format instantiations.

The OMA DRM 2.0 specifications define key management functionality which supports Continuous Media and offers optimisation of the protocols and codecs used as well as their architecture. The PDCF format can be used for various purposes, such as downloaded content and for hosting stream able content. OMA DRM specifies common data structures for file formats and additional information on top of streaming services and PDCF support for devices is an optional feature.

The key management part of the PDCF format is specified in the OMA-TS-DRM-DCF-V2_0-20060303-A specification. “In the `ProtectionSchemeInfoBox`, there is space for a “black box”
(SchemeInformationBox) describing the key management governing access to the encrypted
media content. In a PDCF file, this box MUST be the `OMADRMKMSBox`.”

The basic PDCF file format data structures are defined by the corresponding base file format specification OMA-ERP-DRM-V2_0-20060303-A, and this specification only adds OMA DRM specific structures and parameters. Other DRM mechanisms can be used in those file formats supporting encrypted media tracks, but not in PDCF files, as explained in the OMA-TS-DRM-DCF-V2_0-20060303-A specification.

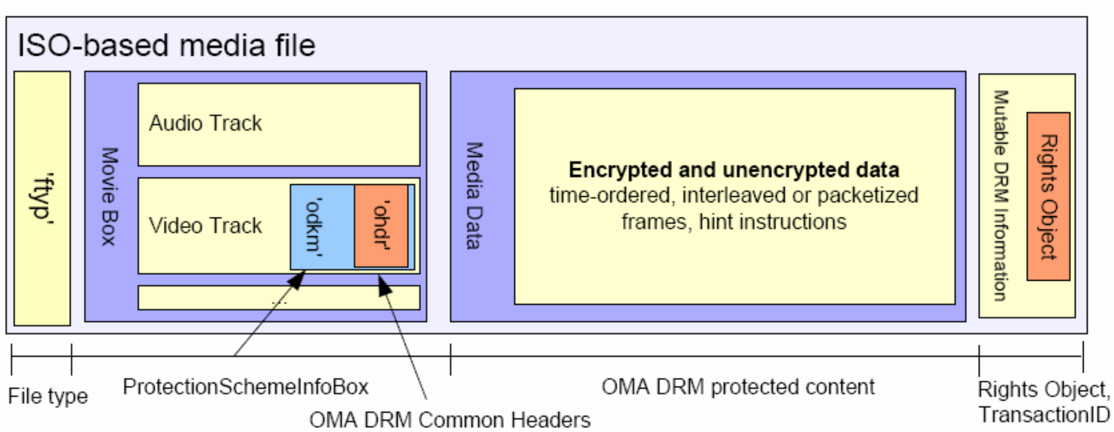

Figure 75 PDCF structure example

The figure suggests how to protect information stored in a PDCF. In this example only the video track is protected by placing a `ProtectionSchemeInfoBox` into the track and specifying the OMA DRM identifier as the key management system. All tracks in a PDCF can be protected with the mechanism.

The PDCF content can be divided into two; stream able and a non-stream able.
A stream able PDCF *has to* conform to the server profile of the file format specification, and the media data is stored as packets. In a non-stream able PDCF the media data is stored as samples. A group of one or more samples is called an *access unit*.

The original packets and samples are encrypted before sending and since the encryption process changes both packet and sample formats from the original plaintext, non-stream able PDCFs *must* have the `OMADRMAUHeader` inserted before each access unit.

Streaming DRM Content is done by using standard streaming protocols. The DRM Content is transferred over a real-time transfer protocol encrypted in packets, include the original payload. The encrypted payload wrapper format can be used in any streaming service which uses RTSP streaming [RFC2326], SDP signalling [RFC2327] and RTP transport [RFC3550].

PDCF streaming support is optional and multimedia streaming sessions can contain both protected and unprotected PDCF tracks. Streaming protected tracks are signalled through SDP parameters, which use the information contained in the sample format entries of the PDCF file. A streaming server derives network packets from a *hint track* in the media file. Streamable PDCF profiles are defined by each service supporting OMA MDR and end-to-end streaming services like [TS26.234] or [C.S0045] specify the RTP payload format used and mechanisms for signalling OMA DRM and encryption parameters.

Parameter name	Purpose
ContentID	ContentID for the protected track
RightsIssuerURL	The RightsIssuerURL for fetching Rights

Table 9 Required OMA DRM specific parameters

Content Distribution

The basic steps for distributing DRM Content can be listed as following [110]:

First the content that si to be distributed has to be put in a secure content container (DCF) and encrypting it with a symmetric content encryption key (CEK). Content can be pre-packing is allowed, meaning that it does not have to happen on the fly. It is also recommended that one should not use the same content encryption key for all instances of a piece of content because it poses a greater risk if a single device was to be hacked and a CEK stored on that device would get into the wrong hands. Therefore different CEK's should be used for different deliveries or different devices.

Second the content has to be authenticated by the DRM Agent authentication. All DRM Agents have a unique private/public key pair and a certificate. The certificate includes additional information, such as maker, device type, software version, serial numbers, etc. which allows the content and rights issuers to securely authenticate a DRM Agent.

Then a Rights Object has to be generated. Right Object is an XML document, expressing the permissions and constraints that are associated with the content. The Rights Object also contains the CEK to ensure that the DRM Content will not be used withought an associated Rights Object. Before the Rights Object is delivered, sensitive parts have to be encrypted with the CEK), and then the Rights Object becomes cryptographically bound to the target DRM Agent. This ensures that only the target DRM Agent can access the Rights Object and thus the DRM Content. Additionally the Rights Issuer digitally signs the Right Object.

Now the Rights Object and DCF can be delivered to the target DRM Agent. Since both are inherently secure, they can be delivered using any transport mechanism such as HTTP/WSP, WAP Push or MMS. Both can be delivered together, in for example a MIME multipart response, or separately.

DRM Content can be safely copied and transferred to other DRM Agents using the Rights Issuer to acquire a Rights Object. The rights issuer controls whether to release a new Rights Object or not to the new DRM Agent. In such way friends can copy DRM content safely among each other.

There are several models for content distribution in DRM, such as Basic Download, Super Distribution, Streaming Media, Domains and Export. Only

1 Basic Download

The Basic Download model lets a client connect to a Content Issuer portal by launching a browser that initiates a browsing session with the Content Portal. The user chooses the content he/she is interested in by browsing the content available. (1) Additionally a payment mode during this session can be selected by the client. Once the content of choice is chosen and downloaded, the client connects to the Rights Issuer portal by looking up the Rights Issuer URL within the DRM Content headers and initiates a connection to the Rights Issuer portal and engages in the Rights Object Acquisition Protocol to get the associated rights (2).

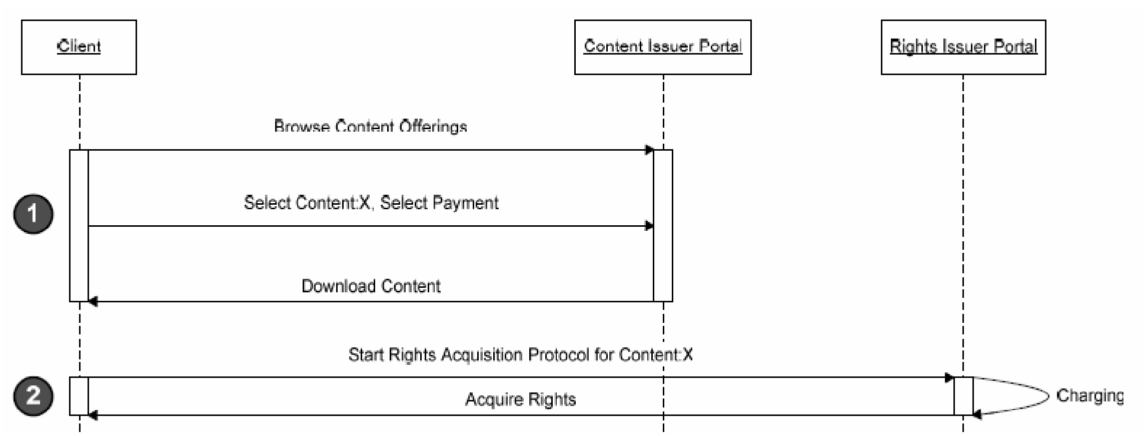

Figure 76 Basic Download 1

Content download can also be done through subscription, where the subscriber gets the DRM Content and Rights Objects pushed to his device on regular interval bases.
After the client has established the subscription and charging agreement with the Rights Issuer, the Rights Issuer pushes the DRM Content and Rights Objects to the clients on a regular interval.

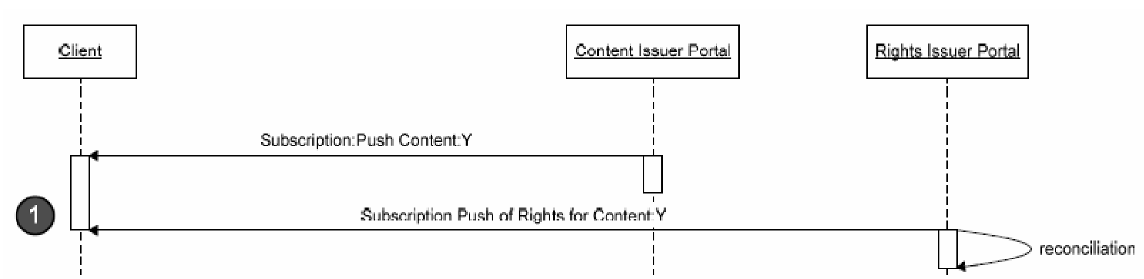

Figure 77 Basic Download 2

The third way to download DRM content is based on subscription where the Device invokes the rights acquisition silently as needed.
This subscription based push initiates a pull of the Rights Object from the Rights Issuer Portal. The DRM Content is delivered with the 'silent' header ("in-advance") and the Client connects to the Rights Issuer to trigger the Rights Object Acquisition Protocol when the content is being received. Once the protocol is completed, the Rights Object is issued to the client.

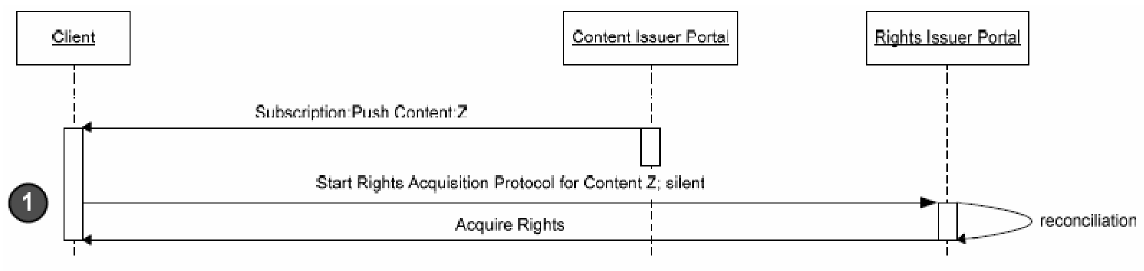

Figure 78 Basic Download - Pull and Push Model

2 Domains

As mentioned in 5.2 it is also possible to distribute content to a group of user devices that are within the same domain. This domain is created, managed and administered by a Rights Issuer and once a domain is formed and the devices are enrolled in it, all content and rights can be shared and distributed to any of the devices within the domain withought connecting back to the Rights Issuer. A device can also join a desired domain on reception of content that is targeted for the domain.

Consider a scenario as illustrated on the next page, where each of the devices D1, D2, and D3 connect to the Rights Issuer, complete the registration and join a domain DM1. Once a device, such as D1 connects to the Rights Issuer, it acquires content DCF1 (DRM Content Format) and the associated domain Rights Object for the DCF, DRM Rights Object1. Now since the device D1 is part of the domain DM1, the content and rights are usable on this device and it can now forward (3) the content and the associated domain RO to the other devices D2, & D3. Since D2 & D3 are part of the domain DM1, the content and associated rights are immediately usable on those devices with ought to connecting to the Rights Issuer (4). The content can also in later time be forwarded to device D4 (5) which is not a member of the domain DM1. As a result, the content is not usable on D4 and the user must connect to the Rights Issuer and join the domain DM1 to gain access to this content. Since the domain management is conducted by the Rights Issuer, the Rights Issuer can explicitly decide on the composition of the domain and whether D4 can become a member of the domain or not.

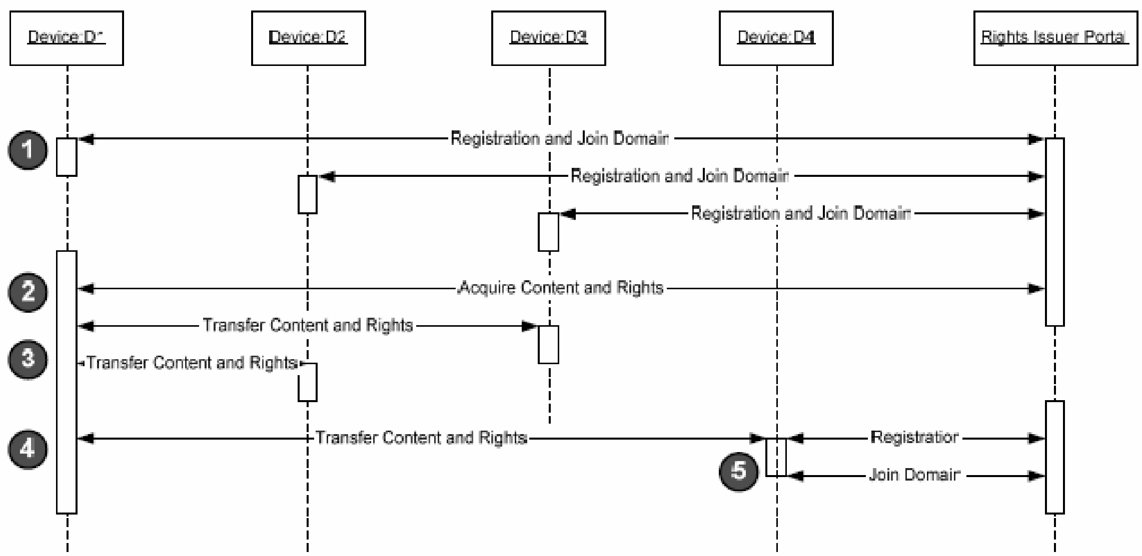

Figure 79 Domain

3 Streaming Media

Pprotected media can also be distributed in streams using a streaming token. A streaming token is a piece of data that the streaming player uses to determine the location of streaming media, possibly also to determine properties of the streaming session or streams, and to set up and start the delivery of streaming media. A SMIL presentation, an SDP session description, or an RTSP URL is examples of streaming tokens in 3GPP.
The streaming token is acquired from the Content Issuer portal and the Rights Object governed all access to the streams.

After receiving the session headers, the client can connect to the Rights Issuer and acquire the necessary Rights Object, which again will provide the necessary information for the client to be able to decode the streams and render the content.

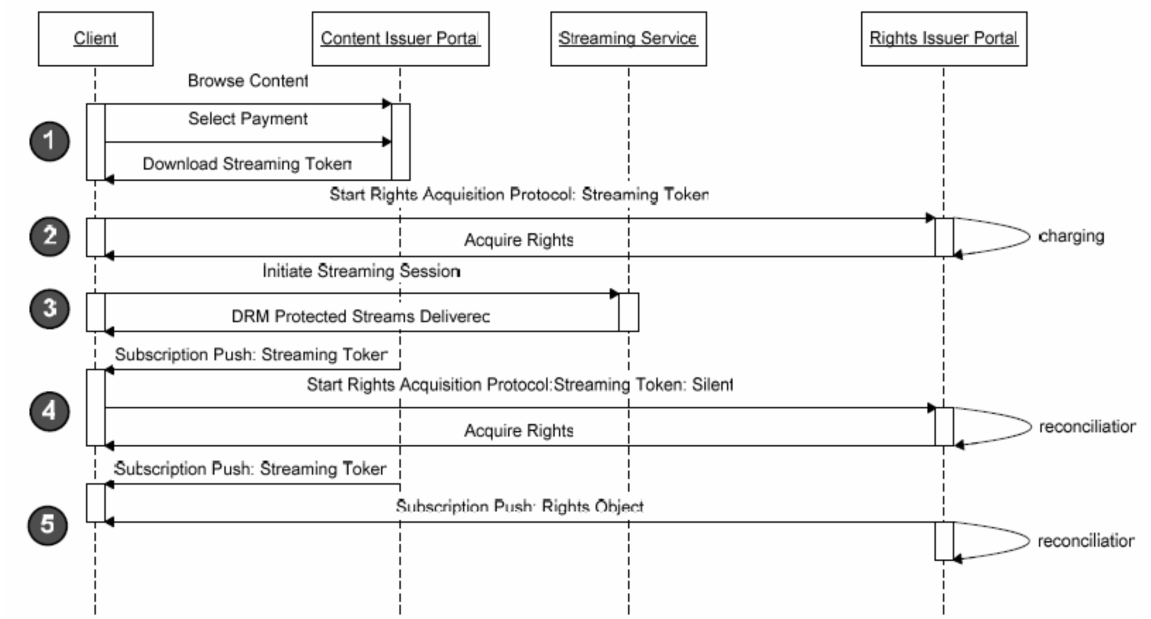

Figure 80 Streaming Media

The session starts with the client connecting to the Content Issuer portal and browsing for content of interest. After the client has made his pick, and decided upon payment agreement, he selects the stream (1) and downloads the streaming token.
In the next step the client has to requests rights by connecting to the Rights Issuer and initiating the Rights Object Acquisition Protocol to acquire the rights for the streamed content (2). Once the protocol has been completed, the client obtains the Rights Object for the streaming service and can now connect to the Streaming server and initiates the streaming session (3). After the stream is initiated, stream properties become available to the client along with the DRM properties. The only exception is for the case of an SDP description token, where the properties are already contained in the token.
The client connects to the Streaming server and receives the Protected Streams. It is also possible to deliver streams with the rights delivered in advance or along side the streaming token. In these cases the client connects to the streaming server and initiates the streaming session. Since then the DRM Agent will have rights, the client will be able to start the streaming session immediately instead of going through step 3.

4 Super Distribution

Any client who has downloaded content from a Content Issuer can distribute this DRM Content to other devices using various networked links as well as removable media. However the DRM Content is not usable by the receiving part because it is encrypted. The only way the content can be of use to the third party, is if the third party acquires the associated rights content. This is done by locating the Rights Issuer URL within the DRM Content header and using this information to connect to the Rights Issuer portal to acquire the rights needed. The interaction diagram below illustrates super distribution of content the related flow of events amongst the significant actors.

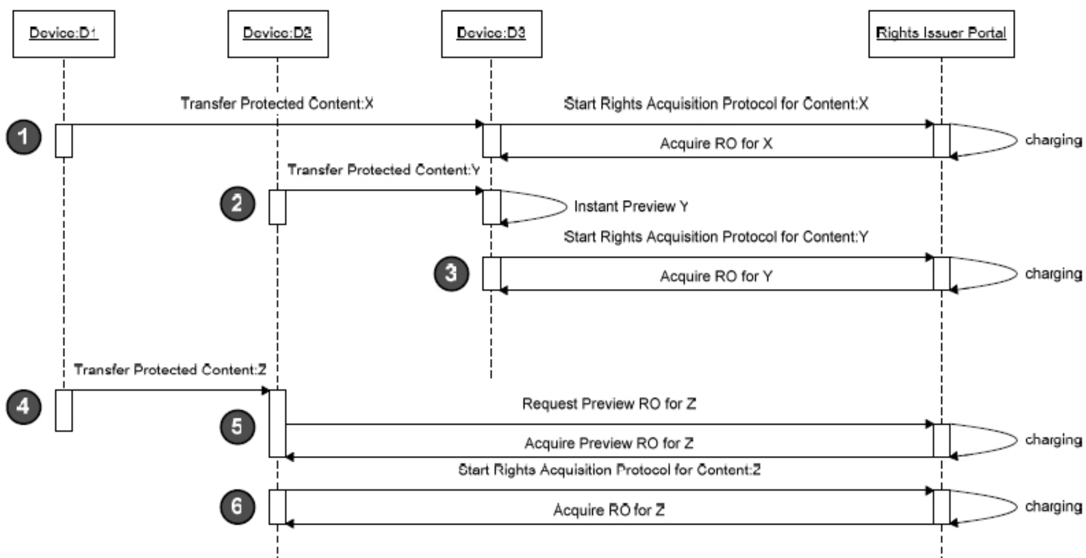

Figure 81 Super Distribution

There are 3 devices used in the example and we assume that the first device D1 already is of possession of stored DRM content. When D1 want to share his content with D3, he first transfers the content to D3 using local connectivity or removable media.
Device D3, on reception of this DRM Content, discovers the Rights Issuer URL from the DRM Content headers and initiates a Rights Object Acquisition Protocol session with the Rights issuer. Once the protocol is completed and payment arrangements are made, D3 gets the Rights Object associated with DRM Content X and is now able to use it (1).

Now D2 transfers DRM Content Y to D3. D2's DRM Content Y has the 'preview' headers and is able to provide an 'instant preview' for the content within it. D3can now offer the 'preview' to its user and the user can decide if he/she would like to buy it or not (2). If the user decides to buy the content, D3 initiates the Rights Object

Acquisition Protocol with the Rights Issuer, completes the protocol and obtains the Rights Object for DRM Content Y (3).

In the (4) D1 transfers DRM Content Z to device D2 which D2 discovers is content with a preview which can be viewed if a preview Rights Object is obtained. As a result, D2 connects to the Rights Issuer and obtains the Rights Object to enable a preview(5). Rights Objects provided are full-fledged Rights Objects with permissions and constraints specified to just enable a preview (Whether this is a feature free of charge depends on the business model). Once the user decides to buy the rights, D2 starts a Rights Object Acquisition Protocol session to acquire rights for content Z and on successful completion of the protocol, the Rights Object for Z is obtained by the device and can be viewed by the user.

As we can see, DRM plays an extremely important role in any media distribution, especially to wireless handheld devise.