Diplomarbeit

Performance Evaluation of Mobile Video Delivery Technologies

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Abstract

The aim of this diploma thesis was to evaluate the performance of the state-of-the-art mobile video delivery technologies. For this purpose two technologies, HSDPA and DVB-H, were chosen. The performance parameters for "high quality" streaming were investigated and all measurements were provided in live network.

A video service over HSDPA is an *On Demand* approach, which means that users are requesting any media at any time they want. DVB-H is a *broadcasting* approach like standard TV usage scenario, where all users are receiving the same media at the same time.

Using these technologies, video sequences were recorded in several situations - scenarios. Additionally, streaming of the most promising video content types was investigated. An objective video quality metric PSNR was used to compute the quality of received video streams at the application layer of the OSI model. Moreover measurements at the network layer were provided.

In the HSDPA case a streaming server was used to provide these measurements. However, this was not possible in DVB-H case, where we did not get access to the content provider. Therefore an alternative solution was chosen.

This thesis proves the ability of mentioned technologies to provide high quality video services.

List of Abbreviations

2G	Second Generation
3G	Third Generation
3GPP	Third Generation Partnership Project
ADT	Application Data Table
AMC	Adaptive Modulation and Coding
ARIB	Association of Radio Industries and Businesses
ARQ	Automatic Repeat Request
ATIS	Alliance for Telecommunications Industry Solutions
AVI	Audio Video Interleave
C/I	Carrier to Interference
CC	Chase Combining
CCSA	China Communications Standards Association
CDMA	Code Division Multiple Access
CDP	Content Delivery Protocols
CN	Core Network
CQI	Channel Quality Indicator
CRC	Cyclic Redundancy Check
CS	Circuit Switched
DMB	Digital Multimedia Broadcasting
DSCH	Downlink Shared Channel
DSS	Darwin Streaming Server
DVB	Digital Video Broadcasting
DVB-H	Digital Video Broadcasting-Handheld
DVB-T	Digital Video Broadcasting-Terrestrial
EDGE	Enhanced Data rates for GSM Evolution

ES	Elementary Stream
ESG	Electronic Service Guide
ETSI	European Telecommunications Standards Institute
FEC	Forward Error Correction
FFT	Fast Fourier Transformation
Fig.	Figure
FPS	Frames per Second
GERAN	GSM/EDGE Radio Acces Network
GPRS	General Packet Radio Service
HARQ	Hybrid Automatic Repeat Request
HS-DPCCH	High-Speed Dedicated Physical Control Channel
HS-DSCH	High-Speed Downlink Shared Channel
HS-SCCH	High-Speed Shared Control Channel
HSDPA	High Speed Downlink Packet Access
HSUPA	High Speed Uplink Packet Access
IEEE	Institute of Electrical and Electronics Engineers
IMS	IP Multimedia System
IMT - 2000	International Mobile Telecommunications - 2000
IP	Internet Protocol
IR	Incremental Redundancy
ISDB	Integrated Services Digital Broadcasting
ITU	International Telecommunication Union
LTE	Long Term Evolution
MAC-hs	Media Access Control high speed
MBMS	Multimedia Broadcast Multicast Service
MCS	Modulation and Coding Schemes
MediaFLO	Forward Link Only
MPE	Multi-Protocol Encapsulation
MPE-FEC	Multi-Protocol Encapsulation Forward Error Correction

MPEG	Motion Picture Experts Group
OFDM	Orthogonal Frequency Division Multiplex
PC	Personal Computer
PCM/CIA	Personal Computer Memory / Card International Association
PDA	Personal Digital Assistent
PDF	Probability Density Function
PID	Program Identifier
PS	Packet Switched
PSI/SI	Program Specific Information/Service Information
PSNR	Peak Signal to Noise Ratio
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase-Shift Keying
QT	QuickTime
RAN	Radio Acces Network
RF	Radio Frequency
RNC	Radio Network Controller
RS	Reed Solomon (code)
RTP	Real-time Transport Protocol
RTCP	Real-time Transport Control Protocol
RTSP	Real Time Streaming Protocol
SAW	Stop and Wait
SDP	Session Description Protocol
SF	Spreading Factor
SI	Service Information
SNR	Signal to Nosie Ratio
SPP	Service Purchase and Protection
SSH	Secure Shell
TD-CDMA	Time Division - Code Division Multiple Access

TDM	Time Division Multiplex
TDtv	Time Division TV
TPS	Transmission-Parameter Signaling
TS	Transport Stream
TTA	Telecommunication Technology Association
TTC	Telecommunication Technology Committee
TTI	Transmission Time Interval
UE	User Equipment
UHF	Ultra High Frequency
UMTS	Universal Mobile Telecommunications System
UTRAN	UMTS Terrestrial Radio Acces Network
VHF	Very High Frequency
W-CDMA	Wideband Code Division Multiple Access
WLAN	Wireless Local Area Network

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Chapter 1

INTRODUCTION

The idea of communication was always understood as a process that allows people to exchange thoughts by one of several methods. By the means of technical progress, these methods came to a higher level. The main communication drivers nowadays are television and mobile communication.

Television became commercially available in the 1930s [TVhist]. Since then it captured people all over the world and became the biggest media. However, probably the highest qualitative step was made in recent years by introducing digital broadcasting. It was supported by former deployment of digital technologies like personal video recorders, DVD technology and video streaming technologies over internet. These triggered enormous interest of consumers.

Mobile communication itself is an extension of the basic telecommunication service - telephony, which started already at the end of nineteenth century. Worldwide employment of mobile services began in the 1990s. The step into a new millennium was also a step to another generation of mobile communication technology. Third Generation (3G) networks provide new services and features, where the most significant are higher capacity and data rates.

The rapid evolution in these two environments and huge user demand of both, leads to a logical ambition of merging them into new types of services - *mobile* *video services*. As a matter of course, this carries some new challenges which can be approached from the side of either of the communication drivers.

- From the *digital broadcasting (TV)* point of view; broadcasters were used to bring TV services to home. Their TV receivers have generally great screen resolution and are supplied directly from mains. This will not be the case if they would like to broadcast to mobile handsets, because these equipments have generally small screens, limited battery and are in motion. The difference is shown in the Table 1.1.

	Broadcasted (digital) TV	**Mobile TV**
Data rate	 4-5 Mbps	 128-384 kbps
Display	Large and medium TV screen	Small, mobile phone screen
Antenna	Roof top, desktop or car antenna	Internal
Power Supply	Fixed, continuous	Battery powered, limited
Reception mode	Fixed, indoor portable or automotive	Mobile handheld

Table 1.1: The attributes of standard TV and mobile TV [Rhode]

- From the point of view of *mobile communications*; service operators are well experienced in issues related to mobile environment. However they were used to offer their services to single users, called unicast. Broadcast services, how we understand them by means of television is the great challenge for this approach [Alcat 06].

There is also another way to look at the growing inquiry for mobile video services. Looking from the providing perspective (the way of distributing the service), there are basically two approaches. One is called *Video Broadcasting* and it is actually the same scenario as television broadcasting. This means, that service providers broadcasts some content, which is delivered to consumers at the same time. Another approach is called *Video On Demand*. This is a scenario which we know from internet. Users are requesting some content whenever they want. They are either downloading some multimedia, or requesting for streaming. (see Figure 1.1).

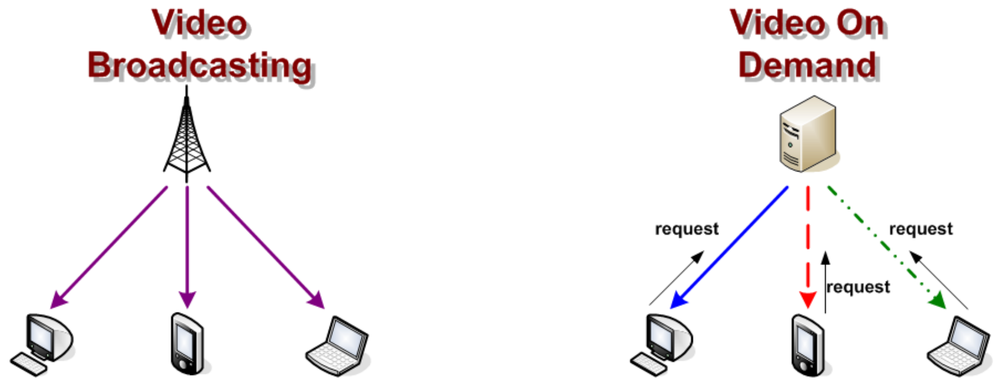

Figure 1.1: Video Broadcasting and Video On Demand scenarios

1.1 Motivation

The motivation of this work is to investigate the state-of-the-art mobile video delivery. Especially evaluate the performance for "high quality" video, which will be assumed as any video with data rate higher than 200 kbps. Several different solutions, which meet the requirements for streaming multimedia content to mobile receivers, already entered the market. These include DVB-H (Digital Video Broadcasting-Handheld), DMB (Digital Multimedia Broadcasting), ISDB (Integrated Services Digital Broadcasting) and MediaFLO (Forward Link Only) [Nadschl 06]. A short overview of these technologies is shown in the Table 1.2.

In addition, there is still the possibility of using the existing 3G network, which is probably the fastest and easiest, but not optimal way to get Mobile TV off the ground. More precisely, technologies like HSDPA (High Speed Downlink Packet

	DVB-H	MediaFLO	ISDB-T	DMB	3G
Standard	Open	Proprietary	Open	Open	Open
Regions	Europe, US, parts of Asia	US	Japan, parts of South America	Korea, parts of Europe, some other countries	Worldwide
Air Interface	OFDM	OFDM	OFDM (sub-banded)	OFDM	CDMA

Table 1.2: Overview of mobile technologies providing TV services

Access) which is an extension of 3Gs UMTS (Universal Mobile Telecommunications System), TDtv (based on TD-CDMA technology from IPWireless) and MBMS (Multimedia Broadcast Multicast Service). None is ideal, as all have drawbacks of one kind or another: spectral frequencies used or needed, signal strength required, new antennas and towers to be build, network capacity required, or difficult business model.

In this thesis, two suitable technologies for two different approaches, as mentioned above, were investigated. For the *Video Broadcasting* approach, it was the DVB-H standard and for *On Demand* video delivery, it was HSDPA. All measurements were done in a live network, using a Darwin Streaming Server and cooperating with one of the biggest Austrian mobile operator.

1.2 Outline of the thesis

In what follows, technologies used throughout this work are explained in more detail. The chapter 2 offers information about HSDPA. In the chapter 3, Digital Video Broadcasting - Handheld technology is explained, mentioning why it is appropriate for mobile environments. Subsequently, information about the test bed, scenarios, measurements and results of HSDPA and DVB-H are presented in chapter 4 and chapter 5 respectively. Finally, in the chapter 6 this work ends with some conclusions.

Chapter 2

HIGH SPEED DOWNLINK PACKET ACCESS

2.1 Historical overview

Mobile telephony systems are undergoing a rapid evolution in recent years. Nowadays the very successful 2G (Second Generation) technology GSM (Global System for Mobile communication), is being altered by its 3G successor, called UMTS (Universal Mobile Telecommunications System).

While the 2G made a giant step from analog networks to digital, the key features of 3G are increased capacity of customers, as well as significantly higher data rates. This opens a variety of new service opportunities, including video telephony or mobile TV. 3G can be based on different underlying radio air interfaces, which are basically related to CDMA (Code Division Multiple Access). Currently, the most common mobile phone technology, UMTS, is based on the use of W-CDMA (Wideband Code Division Multiple Access).

2.1.1 3GPP releases

UMTS was standardized by the 3GPP (3rd Generation Partnership Project), which is a collaboration agreement established in December 1998. It is an international co-operation between European ETSI, Japanese ARIB/TTC, Chinese CCSA, North American ATIS and South Korean TTA[1] [3GPP]. The great numbers of 3GPP standards are structured in a so-called *Release*, which introduces new features. Up to now releases are:

Release 98 - specifies pre-3G GSM networks
Release 99 - specifies the UMTS 3G networks [Summary 99]
Release 4 - adds features including an all-IP Core Network [Summary 04]
Release 5 - introduces HSDPA and IMS (IP Multimedia System) [Summary 05]
Release 6 - introduces WLAN (Wireless Local Area Network) integration, HSUPA (High Speed Uplink Packet Access), MBMS (Multimedia Broadcast Multicast Service) and some other features [Summary 06]
Release 7 - (in progress) focuses on QoS improvements and decreasing latency and other features

Plans for the future beyond Release 7 are developed under the title Long Term Evolution (LTE). One of the main objectives of this research is to enable an all-IP network [Am Rel 7].

2.1.2 Release 99 - UMTS

Every 3GPP release is in some way a milestone in the 3G development, but Realease 99 is one of special importance. It meets the original scope of 3GPP,

[1] For convenience are the shortcuts explained only in the List of Abbreviations

to produce globally applicable 3G mobile phone system, which would fulfill the IMT-2000[2] requirements [Wireless]:

- better spectral efficiency;
- higher peak data rates - up to 2Mbit/s - which would result in a choice of channels with a bandwidth of 5MHz instead of 200kHz indoor and 384kbit/s outdoor of GSM;
- supporting multimedia applications, meaning the retransmission of voice, arbitrary data, text, pictures, audio and video, which requires increased flexibility in the choice of data rates;
- backwards compatibility to second-generation systems;

Release 99 (UMTS) is a very mature specification, providing an evolution path for the GSM network and especially for its improvements GPRS (General Packet Radio Service) and EDGE (Enhanced Data rates for GSM Evolution) [Am Evol 04]. The architecture of UMTS in whole is composed of three subsystems:

- radio access network (RAN)
- circuit switched (CS) core network (CN)
- packet switched (PS) core network.

To ensure the backward compatibility, access network of the new system consists of two main components: GSM/EDGE radio access network (GERAN) and UMTS terrestrial radio access network (UTRAN) [Hakaste]. UTRAN architecture is depicted on the Figure 2.1.

UMTS is nowadays commercial deployed in many countries all over the world, provided by hundreds of mobile operators and having more than 30 million

[2]International Mobile Telecommunications - 2000 is the global standard for 3G as defined by International Telecommunication Union (ITU).

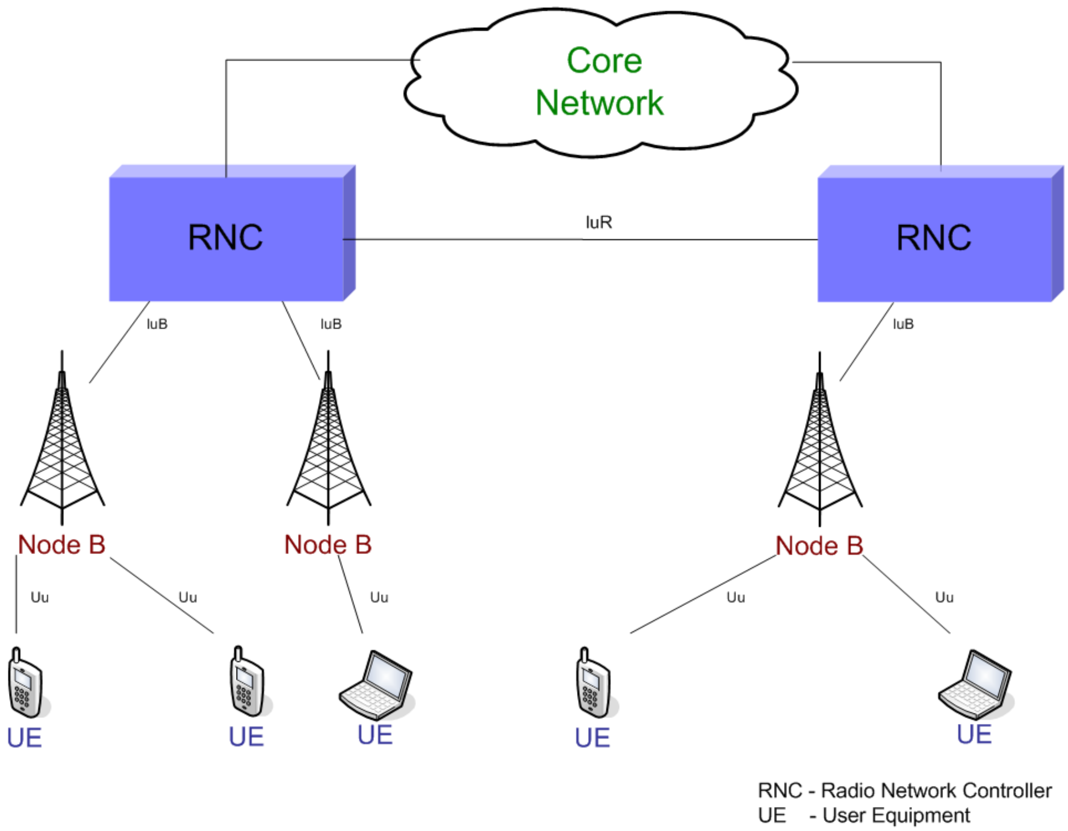

Figure 2.1: UMTS Radio Access Network

subscribers [UMTS Forum]. Data rates considerably exceed the speeds of GSM and reach the limit of 384 kbps. However, these seem not to be enough for some demanding services, like video and music on demand. Hence the potential of 3G had to be further exploited.

Introduction of the new Release 5 aimed to increase the spectral efficiency, improve user experiences and add new services [Nort paper]. One of the key feature of this release is High Speed Downlink Packet Access (HSDPA), which offers much higher capacity and downlink data rates. HSDPA is being described in the following sections.

2.2 HSDPA enhancements

HSDPA provides considerable improvements over Release 99 for the downlink, achieved by employing of some new techniques. Possible peak data rates are up to 14,4 Mbps and sector throughput is increased three- to five-fold, which significantly raises the number of data users on a single frequency [Qualcomm]. Applying some small changes in the UTRAN translates into shorter connection and response times.

Implemented methods like *Adaptive Modulation and Coding (AMC)*, *Hybrid ARQ (Automatic Repeat Request)*, or *Fast Scheduling*, make this improvements possible. Great advantage of HSDPA is also in the very cost effective deployment, since the incremental cost is mainly due to software/hardware upgrade of the Node B and RNC (Radio Network Controller) [Qualcomm]. Mentioned techniques are described in the following subsections.

2.2.1 General Channel Structure

One of the major innovations of HSDPA over Release 99 is the introduction of three new channel types:

- High-Speed Downlink Shared Channel (HS-DSCH)
- High-Speed Shared Control Channel (HS-SCCH)
- High-Speed Dedicated Physical Control Channel (HS-DPCCH)

HS-DSCH

The HS-DSCH is a shared transport channel type very similar to the DSCH in Release 99. As in case of DSCH, the channel is shared among all users in some specific sector. However, while the scheduling in DSCH is done in the RNC, scheduling in HS-DSCH is done closer to the user equipment, in the Node

B. The purpose of the new channel is to be the main radio bearer, offering a best-effort packet data service and supporting the technologies discussed below [Parkvall].

Since HSDPA is based on the CDMA technology, channels are divided by the means of Spreading Factor (SF) codes. HS-DSCH is shared in a number of SF 16 codes. In addition it is shared in the time domain. The corresponding time unit is called TTI (Transmission Time Interval). Its duration is fixed and equal to 2 ms [TS 25.308]. Since HSDPA is a best-effort service, all codes can be assigned to one user during the TTI, or may be divided between other users. The corresponding code and time structure is shown on the Figure 2.2.

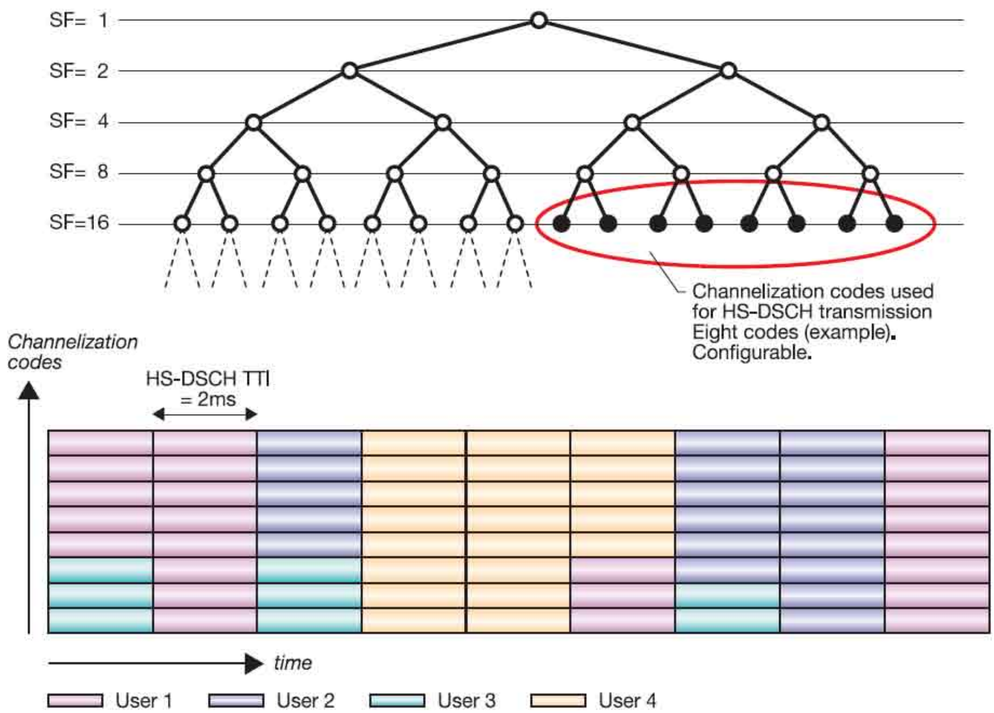

Figure 2.2: The code and time structure of HS-DSCH [Park Eng]

The TTI of 2 ms in HSDPA is significantly lower than the 20 ms, 40 ms, or 80 ms in Release 99. The advantage of shorter TTI is in better short round trip delay between the UE (User Equipment) and the Node B. Furthermore it improves the link adaptation ability and is important for the Adaptive Modulation and Coding (AMC) technique.

HS-DSCH is supported by two additional control channels described next.

HS-SCCH

The HS-SCCH is a control channel used for the downlink signaling between the Node B and the UE. It is accompanied with each HS-DSCH before the beginning of every TTI. Although it is a shared channel, it has a fixed rate of 60 kbps and SF of 128 [Qualcomm]. It carries the information regarding when to receive the HS-DSCH [Hiltunen]. The signaling message furthermore includes AMC and HARQ (Hybrid ARQ) information [DSCH Design].

HS-DPCCH

Up to now, only downlink channels were discussed. For the uplink signaling, a dedicated HS-DPCCH control channel with SF=256 is used. It carries two types of information. Firstly, it is used for the HARQ acknowledgement (ACK/NACK information), to indicate whether the corresponding transmission was successfully decoded. Secondly, it carries the CQI (Channel Quality Indicator), which is used for reasons of link adaptation. Visual representation of described channels is on the Figure 2.3.

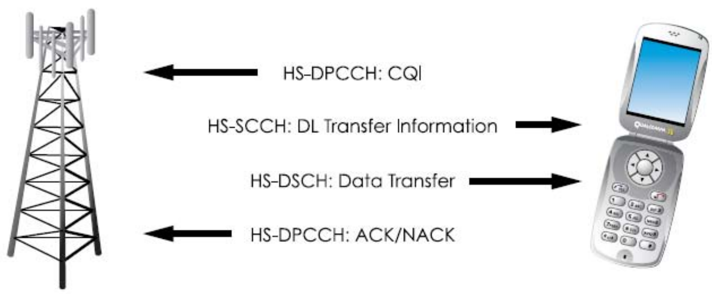

Figure 2.3: New channels in HSDPA [Qualcomm]

2.2.2 Adaptive Modulation and Coding

One of the basic features of HSDPA is the AMC. The fundamental idea is to dynamically change defined Modulation and Coding Schemes (MSC) [AMC 3G]. Each MSC consist of some parameters like *code rate*, *modulation scheme*, *number of codes*, or *transmit power* [Dotlling]. The aim is to continuously optimize these parameters to the actual channel conditions. The decision about choosing the optimum MSC is performed according to the report received from the CQI and can change every TTI.

Encoding

The encoding scheme used in HSDPA is based on the Release 99 rate (1/3 Turbo encoder), but adds some new alternatives (1/4 to 3/4).

Modulation scheme

Another way to adapt to channel conditions is to change the modulation scheme. High order modulations offer higher data rate, but at cost of the robustness. Thus when targeting a higher level of reliability, e.g. being closer to Node B, system can offer higher modulation. Lower modulation is then assigned, when facing worse conditions, e.g. being at the end of a cell. HSDPA standards introduce an additional 16QAM modulation to the existing QPSK used in Release 99.

Number of codes

User equipments in HSDPA can support different number of codes: five, ten, or fifteen. Assigning more codes to one user, results in higher data rates. In the very special case, with 16QAM modulation, 15 codes and with no coding (code rate is one), user is able to achieve the maximum specified peak data rate of 14,4Mbps. However, this case is very unlikely in the reality.

Transmit power

The employment of AMC results in better utilization of the power of Node B.

2.2.3 Hybrid ARQ

Hybrid Automatic Repeat Request is another essential feature of HSDA. It gives better performance than the ordinary ARQ error control method, which is achieved by soft combining and using N-Channel Stop and Wait (SAW) protocol.

Retransmissions requests are sent in the case, when data is not correctly decoded. Otherwise, a positive acknowledgement is signaled through the uplink dedicated physical channel [Parkvall]. The major gain of HARQ provides the utilization of the erroneously received information. This is not discarded, but combined together with retransmitted information, resulting in significantly improved performance and robustness to link adaptation errors. The combining of the retransmitted and erroneously transmitted soft information can be performed in two main ways: *Chase Combining (CC)*, or *Incremental Redundancy (IR)*.

- **CC** is the simpler way, requiring minimum complexity and buffer. The retransmission is identical with the initial transmission, which was erroneously decoded.

- **IR** is a more complex scheme, requiring higher memory for the UE. Instead of sending simple repeats of the information, additional redundant information is incrementally transmitted [HARQ].

Stop and Wait mechanism is used, to manage the retransmissions and acknowledgements. In a single SAW protocol, the transmitter waits for an acknowledgement after every sent information unit. However, there is a desire for better exploitation of the waiting time between acknowledgements. For this purpose N-channel SAW is used, whereby N different SAW protocols are working in parallel. While one channel is waiting for the acknowledgement, the remaining N-1 channels continue to transmit. There are up to six such channels used for advanced Node B implementation [Qualcomm].

Another benefit of HARQ follows from the place of its implementation. It is implemented at the MAC-hs (Media Access Control high speed) sub layer, which is located at the Node B. On the other hand, the retransmission functionality of Release 99 is implemented in the RNC. These outcomes with much lower retransmission delay for HSDPA compared to Release 99.

2.2.4 Fast scheduling

As mentioned above, the HS-DSCH is a best-effort transmit channel, which is shared among all users in a particular sector. Therefore, some part of the system has to be responsible for the allocation of the channel resources among the users. Fast Scheduler is the key element of HSDPA providing described functionality. One of the advantages of scheduling in HSDPA over Release 99 is again in the emplacement. As with the HARQ, also the scheduler is located in the Node B. This leads to better performance and overall behavior of the whole system.

Scheduling method works in conjunction with AMC, choosing different schedulers for different scenarios. Just as in the case of AMC, also the scheduler can modify with each TTI. The chosen scheduler depends also on the CQI, which provides the actual information about the channel. This enables the base station to flexibly decide which terminal's packets to send, considering the *fairness* and *throughput* [Etoh]. Corresponding to them, there are several scheduling algorithms. The most common are: Round Robin, Maximum Carrier to Interference (C/I), and Proportional Fair.

- **Round Robin** is the easiest algorithm. The channel resources are equally assigned to all communicating users [Ofuji]. Users are served regardless of their signal quality. The outcome is a very high degree of fairness, but at the cost of system throughput.

- On the other hand, the **Maximum C/I** algorithm want to maximize the overall throughput. Therefore this scheme chooses only users with maximum C/I, i.e. the best channel conditions. It provides no degree of fairness, penalizing some users who are for instance at the edge of the cell.
- A good trade-off between the fairness and throughput is provided by the **Proportional Fair** algorithm. According to [Troels] it can ideally offer 100% capacity gain over Round Robin. The scheduler selects users according to the ratio between their momentary data rates and their mean data rate [Qualcomm].

There are some other algorithm techniques which take in consideration also the users application (e.g. streaming-aware algorithms) [Lundevall].

2.3 Additional notes

There is a lot to write about the HSDPA technology. But the aim of this work is to be rather more practical than theoretical. The corresponding description of the measurements of this technology follows in chapter 4.

In the end, there has to be mentioned, that HSDPA is already deployed throughout the world. In some countries (including Austria) already by the download data rates of 7,2 Mbps. This is one of the reasons, why this technology was chosen for the thesis.

Chapter 3

DIGITAL VIDEO BROADCASTING - HANDHELD

3.1 Introduction to DVB-H

Digital Video Broadcasting - Handheld is a digital standard which aims at mobile video delivery. It was developed by DVB Project and released by ETSI (European Telecommunications Standards Institute) in November 2004 as a part of the DVB family of standards [FactSheet 07]. It was developed to answer the users demand for Mobile TV service. DVB-H is nowadays (july 2007) launched nationwide in only few countries (Albania, Finland, Italy, Vietnam) [DVBServ], but there are many trials running in different cities (also in Vienna) [MobKom]. There are also several countries were the nationwide service launch is planned for 2007 (France, Germany, Spain, etc.) [DVBServ]. Concerning the availability of DVB-H receivers, yet there are many devices offered on the market by various companies (Nokia [Nokia], Siemens [Siem], LG [LGe], and more). These important matters make it much easier for the possible future success of DVB-H.

3.1.1 Requirements

The principles behind DVB-H are based on former released and already world-wide implemented standard DVB-T (Terrestrial), which specified the terrestrial transmission of MPEG-2 based television services. DVB-T was designed to replace the basic analogue TV broadcasting and hence it should support mostly fixed receivers. However, experimental results and deployments illustrated the technology's ability to also support mobile environments [Bennett]. In spite of that, the DVB-T technology, which targets fixed, electrically powered receivers, does not support battery-limited handheld devices. For this reason, a new standard had to be developed to deal with issues related to mobile environment. DVB-H is an extension of DVB-T and introduces some new features to fulfill all expected requirements [TR 102 377], [Kornfeld 05]:

- **Small power consumption**
  Receivers used in this systems are primarily portable (hand-held) and mobile. The term hand-held includes equipments like multimedia mobile phones, personal digital assistants (PDAs) and pocket PCs. Consequentially, such a device is considered to be battery powered and therefore the DVB-H receiver must avoid high power consumption. Therefore, this transmission system should provide possibility to turn off some part of the reception chain to increase the holding time of the battery [Torshizi 06].

- **Handover**
  As mentioned above DVB-H receivers should be mobile. To guarantee the mobility of users, this system should be able to ensure DVB-H service while leaving a given transmission cell and entering a new one.

- **Flexibility and Scalability**
  DVB-H is expected to be served in different scenarios. Not only indoor and outdoor locations should be considered, but also pedestrian and moving

vehicles environments. The transmission system should offer enough flexibility and scalability to allow the reception of DVB-H services at various speeds, while optimizing transmitter coverage [EN 302 304].

- **Robustness and Man Made Noise**
  Mobile channels suffer of fading and Doppler effects. In addition, receivers are mostly used in areas with high levels of man-made noise. Hence, the transmission system should provide some means to mitigate these effects. In other words, DVB-H standard should ensure some robustness to their services.

- **Various transmission bands and channel bandwidths**
  If a transmission system aims to be served in various part of the world, it should offer the flexibility to be used in various transmission bands and channel bandwidths.

It is also necessary to mention the expectation that DVB-H will provide useful data rate of up to 10 Mbps. The transmission channels are the regular UHF band or alternatively VHF III band [Kornfeld 05].

3.1.2 Overview of the System

In order to meet the above requirements, DVB-H contains a bundle of new extensions over DVB-T, however they can share the same multiplex. For this purposes an encapsulation mechanism called MPE (Multi-Protocol Encapsulation) is used, which makes it possible to transport to data protocols on top of MPEG-2 transport streams. In conjunction with this mechanism, a forward error correction (FEC) scheme is used. This mechanism improves the mobility and the robustness of the signal by employing powerful channel coding and time interleaving.

Another key feature aims to improve battery consumption. It uses power saving algorithm based on time multiplexing of different services. This technique, called time slicing, not only saves the battery, but also allows smooth and seamless frequency handover. DVB-T provides two OFDM transmission modes, namely 2K and 8K, for different network topologies. These modes comprise the information about the OFDM parameters, like the overall number of carriers (FFT size), modulated carriers, carrier spacing, etc. DVB-H introduces additional 4K mode (with 4 thousand carriers). This is useful for better cell design and network planning flexibility. To further improve the robustness in the mobile environment, in-depth symbol interleaver for the 2K and 4K are introduced.

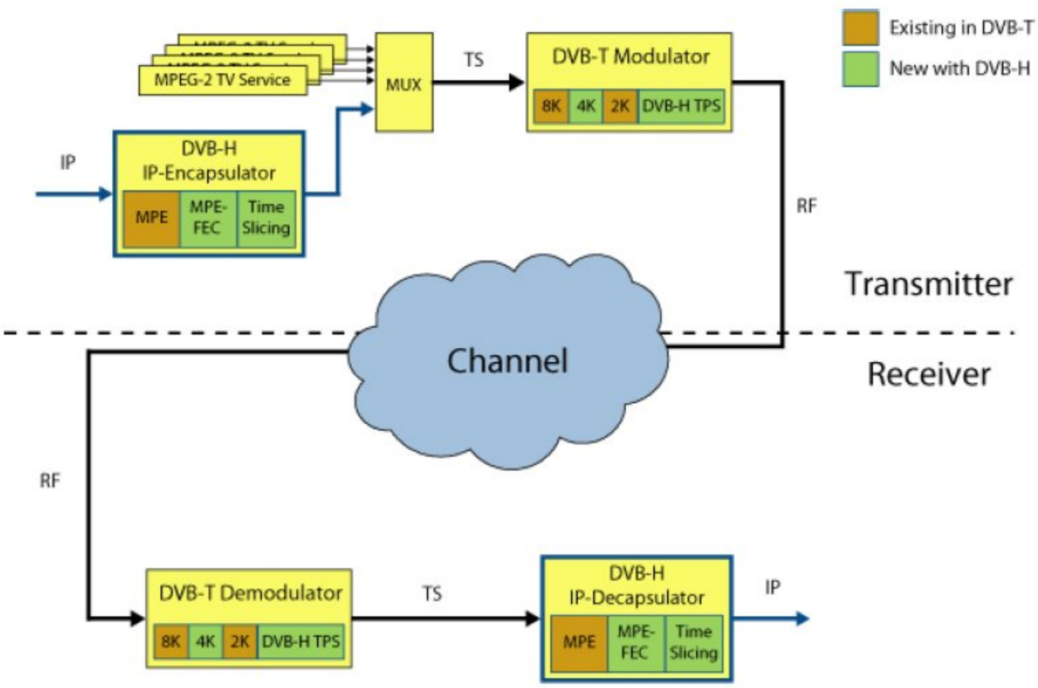

Figure 3.1: DVB-H System, including added new features [FactSheet 07]

All these new features (as shown on Figure 3.1) concern basically only the *physical* and the *link* layer. More detailed explanations follow in below.

3.2 Innovations in the physical layer

The physical layer of DVB-H is performed by the means of the DVB-T standard using OFDM modulation [Kornfeld 05]. It consists of the DVB-T modulator and demodulator. This means that the DVB-H data streams are absolutely compatible with basic DVB-T transport streams. This feature results in a guarantee that DVB-H streams are able to be broadcast not only in networks totally dedicated to DVB-H services, but also in networks where DVB-H streams are only additions to DVB-T streams. To distinguish a DVB-H signal from an DVB-T signal, one mandatory new feature is introduced in the physical layer of DVB-H - it is the TPS signaling for the DVB-H streams in the multiplex. To fulfill other characteristic requirements for mobile environment, techniques like time slicing or MPE-FEC are put onto higher, link level.

For reasons of backwards compatibility, there was an ambition to introduce just few changes in DVB-H. Nevertheless, there are some other new features in the physical layer, which are however not mandatory. A new OFDM 4K mode and an in-depth interleaver are introduced in DVB-H.

3.2.1 Signaling in the TPS-bits

TPS-bits are known already from the DVB-T standard. However, DVB-H creates some new elements of the TPS channel. The purpose of these additional TPS-bits is to signal the presence of time slicing and MPE-FEC as well as the 4K mode option. Broadcasting of the cell identifier, which was optional in DVB-T, is in DVB-H made obligatory [TR 102 377]. TPS-bits are very robust and in addition to that, demodulating of the carried information can be processed much faster than demodulating of SI-tables (Service Information) or the MPE-header.

3.2.2 OFDM transmission modes

DVB standards are based on OFDM transmission system. It is advantageous to use different OFDM parameters for different environment scenarios. The set of OFDM parameters are called modes. DVB-T uses two modes, particularly 2K and 8K mode, which correspond to about two thousand and eight thousand OFDM carriers respectively. Since the 8K mode has four times as many carriers than 2K, its carrier spacing is four times lower. This means that 8K mode can offer better maximum distance of transmitters, but consequently it is also more sensitive to Doppler effects. On the other hand, in the 8K mode, the mobile receiver process the transmission at lower speeds compared to 2K. The comparison between OFDM transmission modes is shown in the Table 3.1.

	Mode		
OFDM parameter	2K	4K	8K
Overall carriers (= FFT size)	2048	4096	8192
Modulated carriers	1705	3409	6817
Useful carriers	1512	3024	6048
OFDM symbol duration (µs)	224	448	896
Guard interval duration (µs)	7,14,28,56	14,28,56,112	28,56,112,224
Carrier spacing (kHz)	4.464	2.232	1.116
Maximum distance of transmitters (km)	17	33	67

Table 3.1: Signal parameters for possible DVB-H OFDM transmission modes [Kornfeld 05]

Since the network planning of DVB-H carries some additional challenges, it is very useful if the system provides more flexibility. That is why DVB-H introduces an additional OFDM transmission mode (4K) using 4096 FFT size, which represents a compromise solution between the 2K and 8K modes. It suppose to fill the gap of DVB-T modes and thus should be the alternative solution for the trade-offs between coverage area and mobility [Bennett], [Faria 02].

3.2.3 In-depth interleaver

Since 8K mode has the highest number of carriers among all three modes, it requires more memory to perform interleaving than 2K and 4K. Roughly speaking, since the number of carriers for 8K is four and two times the number of carriers for 2K and 4K respectively, interleaving memory needed is four and two times lower. The idea behind *in-depth interleaving* is to allow the full utilization of the 8K memory also for the 4K and 2K modes. This leads to an increase of memory for these modes. Since 8K mode uses one OFDM symbol, the in-depth uses four consecutive OFDM symbols for the 2K mode and two consecutive symbols for the 4K mode. This relation is shown on the Figure 3.2.

There is also another way of interleaving, denoted as *native interleaving* [Kornfeld 05]. Unlike in-depth interleaving, native interleaving does not use the full size of the memory for all modes, but uses the individual length of the corresponding mode.

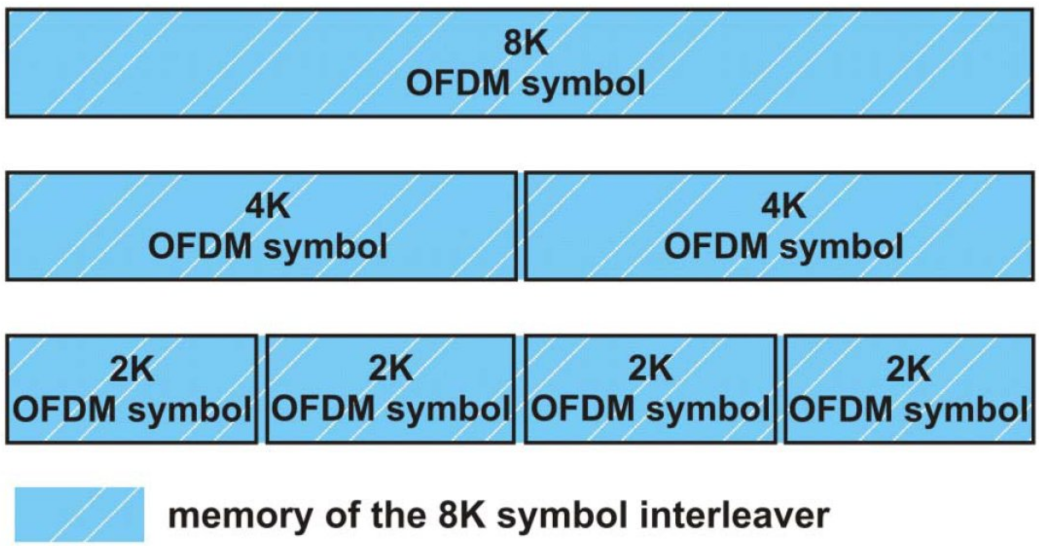

Figure 3.2: In-depth symbol interleaving [Kornfeld 05]

The advantage of in-depth symbol interleaving is the improved reception in fading channels and better protection against impulse interference.

3.3 Link Layer

As shown above, DVB-H uses almost the same physical layer as DVB-T, with only small extensions. That is the reason why DVB-H can be backward compatible with DVB-T [DigiTAG 05]. However, there are still some major requirements to be fulfilled for the ability of mobile video delivery. These are addressed to the link layer. This layer provides the most significant changes compared to DVB-T.

Given the requirement of power saving, DVB-H introduces a special technique called *time slicing*. It results in remarkable battery saving effect and additionally allows soft handover. To solve the issues of robustness and poor signal reception conditions, a new error protection scheme called *MPE-FEC (Multi-Protocol Encapsulation Forward Error Correction)* is introduced (graphically shown on the Figure 3.3). This scheme employs Reed-Solomon channel coding.

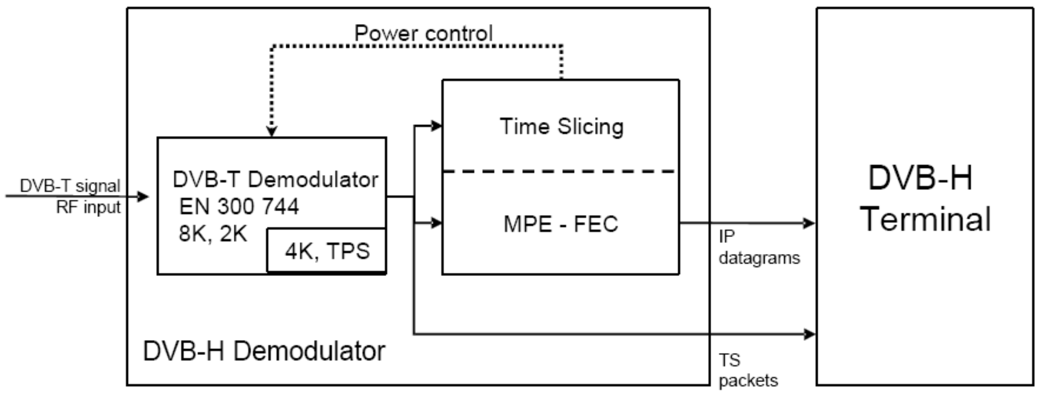

Figure 3.3: Conceptual structure of DVB-H receiver [EN 302 304]

These major extensions are described in the following sections.

3.3.1 Time Slicing

In traditional continuous mode (as used in DVB-T MPEG-2 data transmission), several services (e.g. TV programs) are transmitted in one transport stream. The services are multiplexed together and transmitted in parallel [Henriksson 05]. Therefore, it is not possible to receive only one transport stream, but the receiver has to process all the data (see Figure 3.4) and afterwards choose desired stream (service). This obviously leads to high power consumption.

Figure 3.4: Continual transmission structure [Kangas 06]

The objective of time-slicing is to reduce the average power consumption of the terminal and enable smooth and seamless service handover [EN 302 304]. The technique is based on the TDM (Time Division Multiplexing), which has long been used in communications systems for providing services to different users in different time slots over common media [Yang Song 04]. Services are not longer transmitted in parallel, but consecutively in time slots, called bursts. These bursts have not necessarily the same duration, which is illustrated on the Figure 3.5.

From this follows, that the full DVB-H data capacity is always used only by one service for a fraction of time (say 200ms). After this, another service is transmitted and then another, and so on. After couple of services, the first service is again in the air. This is shown on the Figure 3.6, where the period between sending the same service for this example is 4s.

Hence, in contrast to DVB-T, it is possible to receive only the desired service. Receiver is receiving data only in a short time, while processing the desired service burst. During the waiting time for the next burst, it is turned off. Using

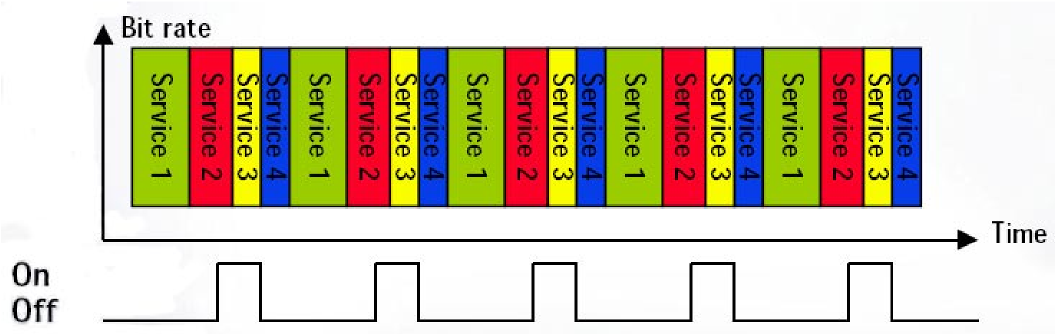

Figure 3.5: Time slicing structure [Kangas 06]

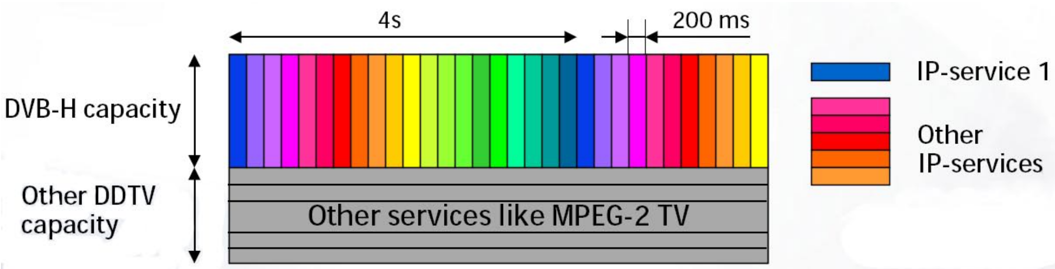

Figure 3.6: DVB-H and DVB-T services [Henriksson 05]

this method, more than 90% of the power can be saved [Kornfeld 05].

Delta - T method

Apparently, there has to be some mechanism to ensure that the receiver will turn on at the right time, while desired burst is awaited. For this purpose a *Delta-T* method is used. The task of this method is to indicate the start of the next burst (Figure 3.7). Every burst carries information about the arriving time of the next one. Delta-t timing information is relative, i.e. it tells only the time from the start of the currently received MPE section to the start of the next. This allows time slicing being insensitive to any constant delays within the transmission path. However, the accuracy of Delta-T can be decreased by jitters, having some impact on the achieved power saving. According specification, the accepted Delta-T Jitter for time slicing is 10ms, which should be easily achieved

[EN 301 192].

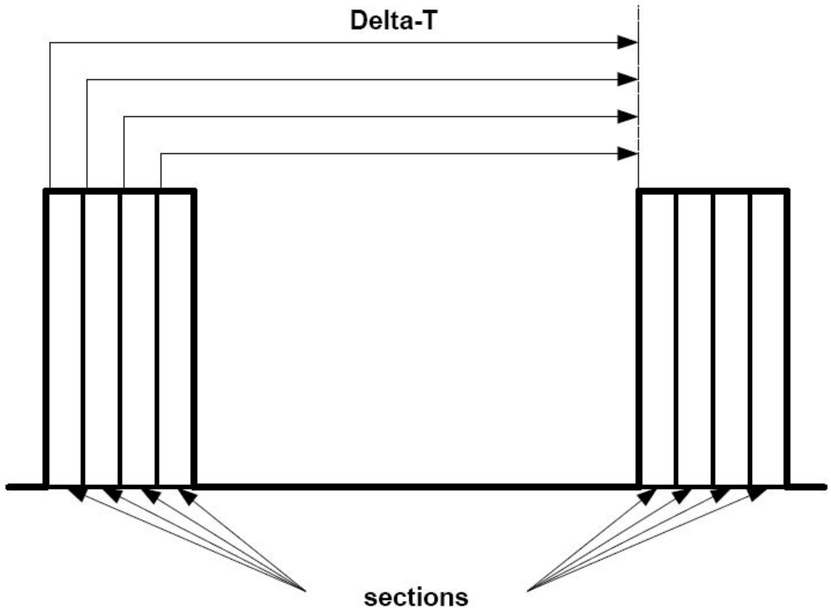

Figure 3.7: Delta-T method [EN 301 192]

Another benefit of using relative delta-t time instead of absolute, is that there is no need for special synchronization between the transmitter and receiver [Bennett].

Burst parameters

There has been always an effort from two sides for getting better support of various services in mobile environment. Firstly, it is a still present effort of increasing the transmission data rates. Secondly, it is an ambition for getting better content compression schemes (e.g. video compression), roughly speaking, it is an effort for decreasing the data rates of the content. The fact that transmission data rates in today's mobile networks are much higher than data rates needed for delivering "high quality" video[1] allows the using of time slicing and

[1] "High quality" video was defined in chapter 1 as any video with data rate higher than 200 kbps

thereby battery saving.

Receiver is "listening" to the data only a fraction of time (*Burst duration*), in the remaining time it is turned off. During receiving, the data is stored in the buffer and played during the *Off-time*. *Burst size* corresponds to the number of network layer bits within a burst. Clearly, it must be less than the buffers memory available in the receiver.

Burst Bandwidth is approximately the momentary bandwidth of the burst, while *Constant Bandwidth* is the average bandwidth needed for transmission of the same information. The *Off-time* is the time between two consecutive bursts. These parameters are graphically displayed on the Figure 3.8.

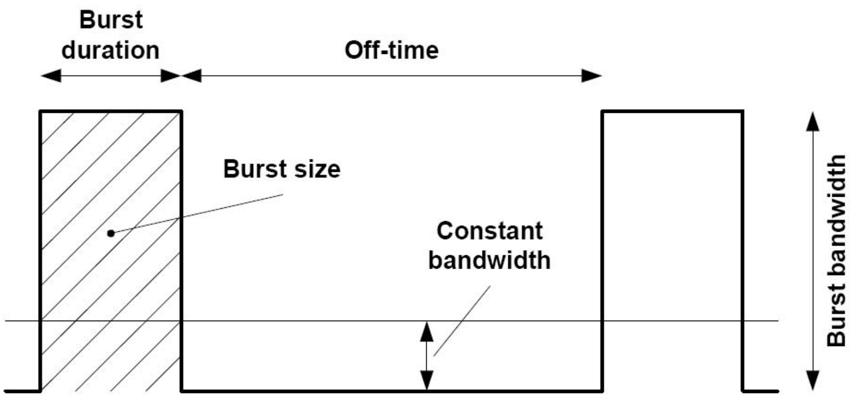

Figure 3.8: Burst parameters [EN 301 192]

Handover issue

Beside great power savings, Time slicing brings another advantage. This is the support of handover. During the off periods, the receiver is able to scan other available frequencies in order to find best alternative, or to execute the handover [Far Hen 06]. This allows receiver a seamless handover, which is very desirable in DVB-H system. More about the DVB-H handover issue is described in the IEEE Magazine Survey [IEEE Survey 06].

3.3.2 MPE - FEC

Multi-Protocol Encapsulation

The Multi-Protocol Encapsulation (MPE) is a method for providing a transport of data network protocols on top of the MPEG-2 TS (Transport Streams) in DVB networks [MPE Sim 05]. IP datagrams are encapsulated into MPE sections. Multiple MPE sections create an Elementary Stream (ES), which is a stream of MPEG-2 TS packets with a corresponding program identifier (PID) [Far Hen 06]. Each MPE section consists of a header, cyclic redundancy check (CRC) tail and a payload, which is actually the IP datagram carried by MPE section. The complex protocol stack is shown on the Figure 3.9.

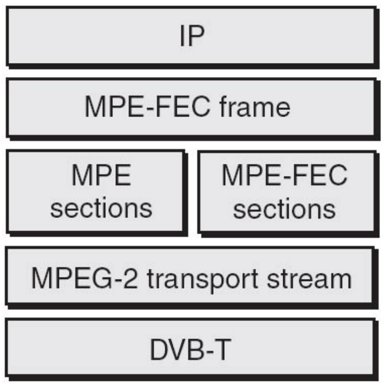

Figure 3.9: Protocol stack of DVB-H [Sams]

Improvements over DVB-T

DVB-T was designed for receiving TV services mostly on fixed receivers, with big antennas. However, it is possible to receive DVB-T services in moving vehicles, but in that case, the need for appropriate antenna size is even greater. There are two stages of error protection used in this standard. It is the inner convolutional code and outer Reed-Solomon code [Kornfeld May 07]. Although these already provide a strong protection, it is not sufficient for the mobile DVB-H scenario.

In contrast to DVB-T, receivers in DVB-H networks are generally in motion and are characterized by small antennas. In addition to that, signal transmission has to deal with multipath effects like frequency selective fading and Doppler effects. Of course, in such scenarios, one can assume a high level of man-made noise. This leads to very poor reception conditions. Hence, the error protection used in DVB-T is not satisfactory and therefore a new forward error correction (FEC) scheme, called *MPE-FEC*, is introduced. The aim of the added protection is to reduce SNR requirements for handheld reception and to increase the mobility speed of the terminals. All stages of error protection in DVB-H are shown on the Figure 3.10.

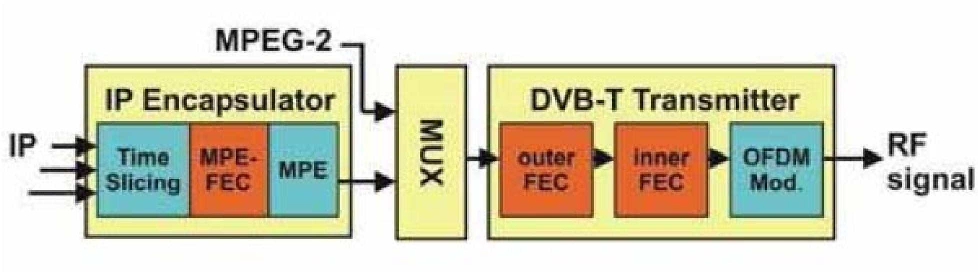

Figure 3.10: Different stages of error protection in an DVB-H transmitter block [Kornf Perf 06]

MPE-FEC together with *Time slicing* method, form the two main improvements of DVB-H. They also closely link up, since one burst, as used in time slicing, corresponds to one MPE-FEC frame. In contrast to Time slicing, which is set up to be mandatory in DVB-H, the use of MPE-FEC is optional. Nevertheless, because of the fully backward compatibility, MPE-FEC ignorant receivers in DVB are able to receive MPE streams. Moreover, for each elementary stream in DVB-H, it is also possible to decide whether MPE-FEC is used or not. This is easily achievable because the IP data and parity data are separated as described in the next section.

MPE-FEC frame

MPE-FEC frame is organized as a matrix with constant number of 255 columns and a variable number of rows. The information about the number of rows is carried in the service information (SI) and gains the common values of 256, 512, 768, or maximum 1024 [Far Hen 06]. Since one cell of the frame corresponds to one byte, a MPE-FEC frame (and thus a time sliced burst) of maximum size (with 1024 rows) would be almost 2 Mbits large.

MPF-FEC frame is composed of two parts; *Application data table (ADT)* with 191 columns and *RS (Reed Solomon) data table* with remaining 64 columns (see Figure 3.11). The data itself, carried in form of IP datagrams, fill the ADT. These are the information bits, which are intended to be protected. In the RS data table is the parity information calculated from the ADT using a Reed Solomon code.

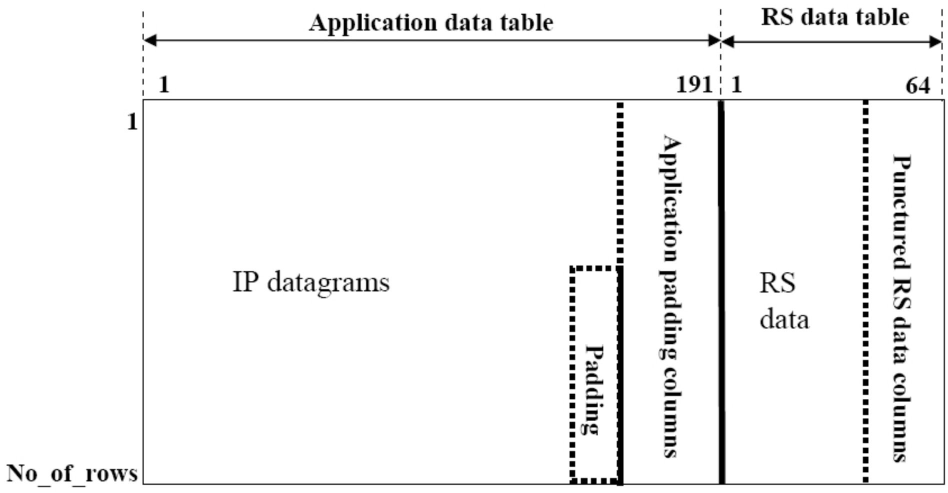

Figure 3.11: MPE-FEC frame [Lopez 06]

- **RS coding**

  IP datagrams are located in the frame one after another, starting in the upper left corner of the matrix and continuing vertically - column wise. There is no space left between datagrams, although they might be of different lengths. If a datagram does not finish until the end of a column, it proceeds to the next column. When all datagrams are included in the frame, the remaining unoccupied byte positions are filled with padding bytes (all zero bytes).

RS data table is filled with parity bytes calculated from the IP datagrams and possible padding bytes from the ADT. Certain code generator polynomial and field generator polynomial are used for this Reed Solomon code RS (255, 191). Using these technique, the RS data table is completely filled and thus the MPE-FEC frame is completed [TR 102 377].

- **Block interleaving**
  In addition to the powerful coding mechanism, MPE-FEC frame structuring contains a block interleaving effect [Kornfeld May 07]. This is achieved by the different direction of writing/reading process and coding. Particularly, writing to and reading from the MPE-FEC data frame is done vertically, in column direction, whereas coding is performed horizontally, in row direction (as clearly shown on the Figure 3.12). The outcome is an interleaving effect applied over the whole data [Kornfeld May 07].

Decoding

Very important feature in the decoding mechanism is the CRC-32 code protection, which is used instead of checksums. It tells the receiver whether a message was corrupted or not. Bytes are marked as reliable or unreliable. For more details of how decoding mechanism in DVB-H works, or more details about different decoding methods, see [TR 102 377] and [Himmanen 06] respectively.

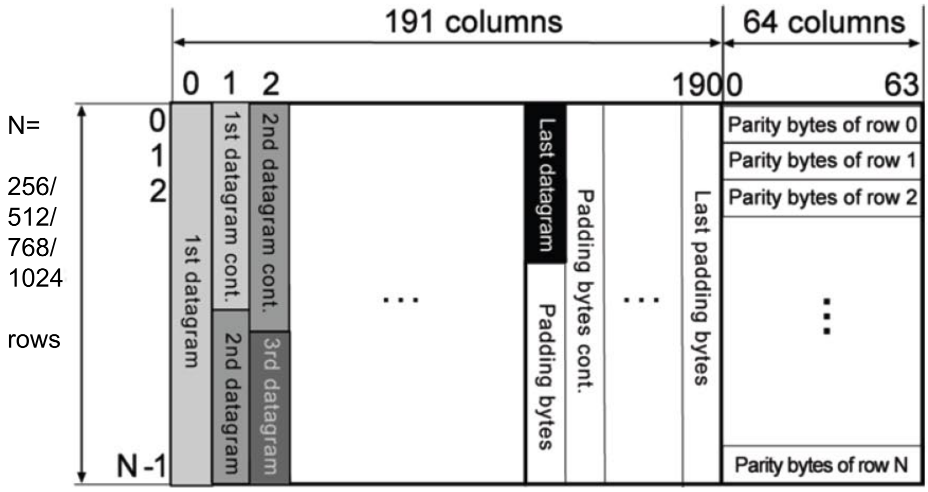

Figure 3.12: MPE-FEC frame structure containing data [Balaguer]

3.4 Remaining issues about DVB-H

In this section, there are some various remaining issues, which are important and should not be omitted without being mentioned. These in particular are the interactivity, DVB-H networks and the standardization. For more details about DVB-H and it's not mentioned features, it is necessary to look on the appropriate bibliography listed at the end of this work.

3.4.1 Interactivity

As a consequence of rapid evolution in all technological areas, the role of broadcasters is changing. There is an emergent demand for interactivity in broadcasting scenarios. Interactivity can be used for various applications, e.g. voting, participating in game shows, etc.

Since DVB-H is designed as a unidirectional transmission system, there has to be some cooperation with some other networks to provide the uplink connection. There are lot of different proposals and ready solutions, which count with the use of SMS, GPRS (General Packet Radio Service) or UMTS (Universal Mobile Telecommunications System). The basic block diagram of such a network is depicted on the Figure 3.13.

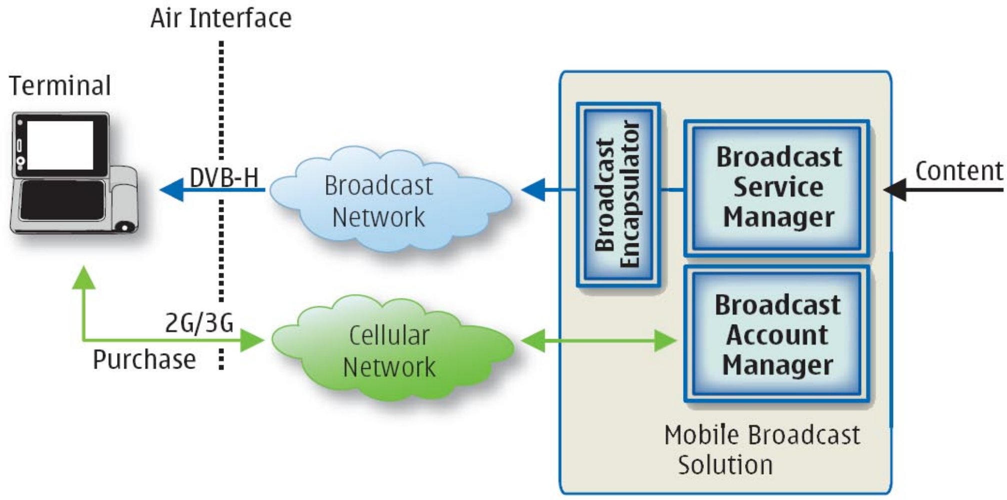

Figure 3.13: Interactivity as proposed by Nokia [Nok Inter]

3.4.2 DVB-H network

There still remains a opened question, how to organize the DVB-H system in a network. Moreover, a reasonable question is, whether it is possible to broadcast DVB-H in an already build DVB-T network. There are two basic possibilities how to approach to a DVB-H network [Ollikain].

- **Shared network with DVB-T**
  In this approach the same network transmitters which are used for DVB-T are also used for DVB-H. The only change that has to be made in the transmitters, is an update in TPS information bits. DVB-H signaling bits and Cell ID has to be added there. The sharing is provided on the multiplex level, where the key DVB-H component is the IP-encapsulator. In this part, time slicing, multi-protocol encapsulation of the data and MPE-FEC are implemented.

- **DVB-H Dedicated network**

  In dedicated network, the full multiplex and the complete RF (Radio Frequency) channel serve to DVB-H transmissions. Dedicated approach takes advantage over the shared network mostly by the means of freedom of planning. This is increased due to the possible use of the additional OFDM 4K mode or in depth interleavers, which were introduced in DVB-H standard.

  A typical dedicated network is composed of several SFN (Single Frequency Network) areas, which then create an MFN (Multi Frequency Network). The maximum size of such a SFN is typically in the order of tens of kilometers and depends on several parameters, like the FFT size and guard interval [Far Hen 06]. Furthermore, the geographical aspect has to be taken into account.

  Practical drawback of this approach is the low number of users, who has to bear the costs. But this problem should be solved by the increase of availability and popularity of DVB-H.

IP Datacast

One of the major differences between DVB-T and DVB-H, which has not yet been mentioned, is that DVB-T is a transmission system based on MPEG-2 transport streams, whereas DVB-H is IP based. This allows the system much better interoperability with other networks as defined in IP Datacast. IP Datacast is a system which makes it possible to employ broadcast content and mobile communications services on a single device [May]. This is very useful thinking especially of combining classical (but mobile) telephone services with the digital TV services of DVB-H, but also for the interactivity issue.

IP Datacast is a set of individual specifications, with an intention to form an overall system comprising different networks. These are specified in several documents, which can be found on the official DVB-H webpage http://www.dvb-h.org.

The key components of DVB IP Datacast are: ESG (Electronic Service Guide), CDP (Content Delivery Protocols), SPP (Service Purchase and Protection), PSI/SI (Programme Specific Information/Service Information) [FactIP 07].

3.4.3 Standardization

DVB-H system is not specified in one document, because it is an extension of DVB-T and thus only affected layers had to be defined. Therefore, there is a bunch of standards dealing with DVB-H, denoted as the *DVB-H standard family* (shown on Figure 3.14). The DVB Project defined the core of the system already in 2004 in a normative specification [EN 302 304]. The *Physical layer* is described in the [EN 300 744], whilst the *link layer*, together with two most important extensions (Time slicing and MPE-FEC), is defined in [EN 301 192]. Document [EN 300 468] describes the signaling in DVB-H. Finally, there is a document dealing with practical implementation hints called *Implementation Guidelines* [TR 102 377].

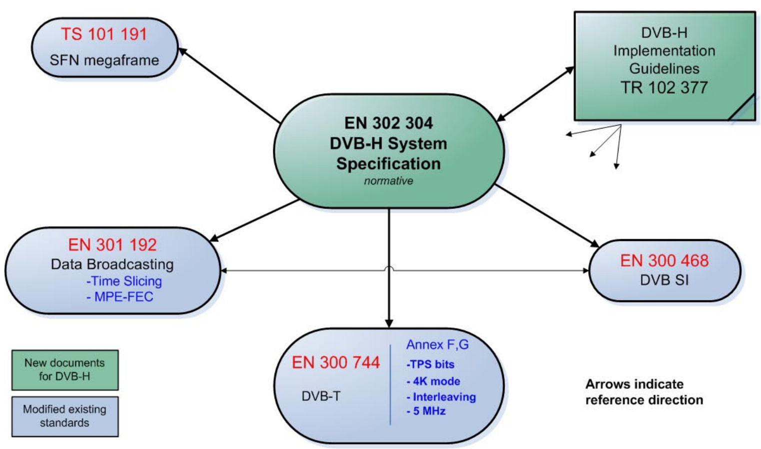

Figure 3.14: The family of DVB-H standards

Chapter 4

HSDPA MEASUREMENTS

The aim of this work is to evaluate the performance of the chosen video delivery technologies. This can be done by several different ways. One possibility is to investigate the delivery at the very lower layer, looking for packet losses, bit rates, etc. For this work, an upper layer approach was chosen, looking at the received video. The evaluation is based on the measurement of video quality. For this purpose the received video was recorded in different scenarios and compared it to its original. Note that that only video, no audio streams, were recorded and measured.

Since mobile TV is expected to be used in various situations, measurements were performed in different scenarios. Despite the mobile nature of this service, it is assumed that TV will be watched also in fixed situations, for instance while waiting on the bus station, or inside the building. Therefore, the chosen scenarios include a simple indoor and outdoor environment without any or small movement. In addition, some truly moving scenario is very typical for mobile environment and therefore was investigated as well.

For providing the measurements, there was an ambition to use a promising video content for mobile TV environment. This in particular includes sport (football), news and music clips [Knoche 05]. However, measuring the same videos was only possible in the HSDPA case. With the use of a streaming server, it was possible to stream any arbitrary video. On that account, one could study the desired videos in all chosen scenarios. This was not workable in the DVB-H case as it is described in chapter 5.

Before stating the HSDPA measurement results, some information about the measured videos, about the scenarios and tools is offered. Furthermore, in this section there is an exact procedure description of the video quality analyses.

4.1 Scenarios

All measurements were provided in 3 different scenarios in the Vienna city. The ambition was to test the most promising scenarios for the mobile TV use and thus to emulate the real usage scenarios. It is expected that videos, will be watched in a indoor environment. Furthermore, the utilization in a truly mobile environment is expected, e.g. traveling in a tram, train or a car. With the growing popularity of the HSDPA service, it is expected that some cells will experience high traffics at some parts of the day.

Chosen scenarios for this work include the *Indoor* scenario, the *Tram* scenario and the *Pedestrian* scenario and all of them are experiencing high data traffic.

4.1.1 Indoor scenario

The indoor scenario was measured at the Institute of Communications and Radio-Frequency Engineering of the Vienna University of Technology. This place was chosen for three reasons: relative high traffic in cell, good signal strength and convenient working conditions (since it was measured at the place of work for this thesis). The map on Figure 4.1 shows the indoor location and the base station of the corresponding cell. The average HSDPA traffic of the cell during the measurements is shown on the Figure 4.2.

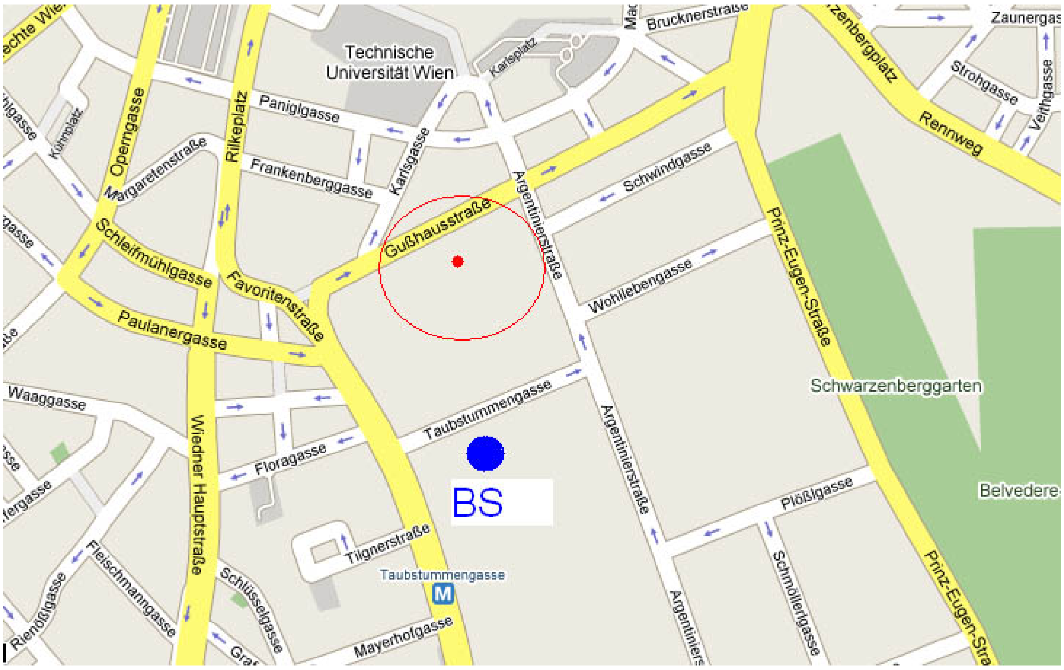

Figure 4.1: Indoor scenario location and the corresponding base station

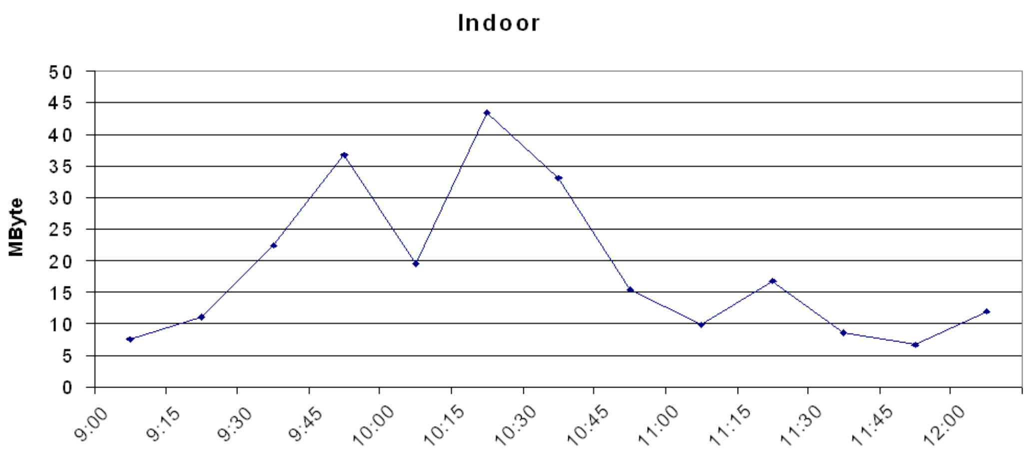

Figure 4.2: Average HSDPA traffic in time during the Indoor measurements

4.1.2 Tram scenario

The tram scenario was measured at the centre of Vienna, while traveling in the tram number 1. The route of this tram is shown on the Figure 4.3. It was expected that during the streaming and measuring this scenario it would come to more handovers. This was also the reason of providing this measurement, to investigate the quality in a environment with high data traffic and handovers.

Figure 4.3: The measurement route of the Tram scenario

4.1.3 Pedestrian

The Pedestrian scenario was measured in a cell which previously showed high traffic and was proposed by the cooperating mobile provider. The area of Hietzinger Kai and Hackinger Strasse was chosen. It is shown on the Figure 4.4 together with the base station. This scenario was measured outdoor, sitting on a bench or walking little bit on the street. However, the movements were not fast, since it is not expected that anybody will be watching some videos while fastly walking or even running. However this was a difficult scenario with lot of fading, because the measurements were provided behind big buildings. The

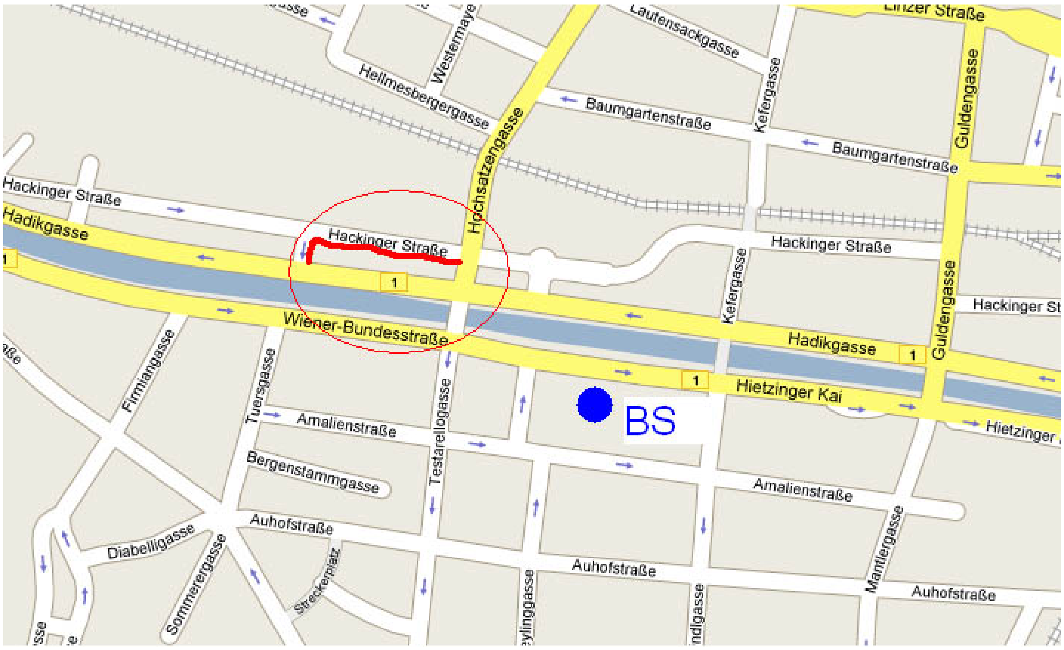

Figure 4.4: Pedestrian scenario location and the corresponding base station

average HSDPA traffic of the cell during the measurements is shown on the Figure 4.5.

4.2 Video Content

As already stated in the beginning of this chapter, there were 3 different types of the most frequent video content used for this work (the captures are shown on the Figure 4.6): **football**, **news** and **music clip**.

The major characteristics of *football* are the prevailing green field and the fast moving ball. Therefore it is expected that the streaming of this video will lead to more transmission errors than the news.

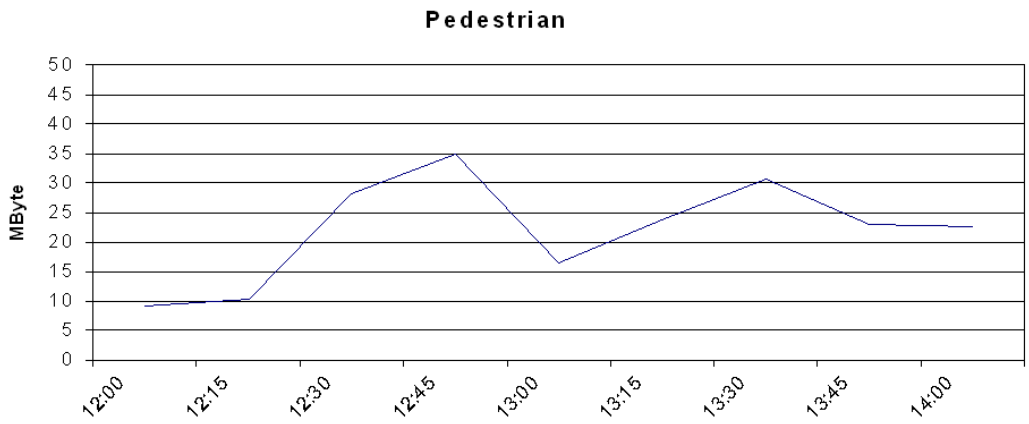

Figure 4.5: Average HSDPA traffic in time during the Pedestrian measurements

Figure 4.6: Video content

The *news* video is characterized by very few and slow changes which naturally is expected to result in better performance.

The *music* clip is a mixture of different characteristics. On the one hand the scenes in the particular video are not moving fast, but on the other hand, they are changing very quickly.

4.3 Tools

There are a number of different programs and tools, which were used for providing this performance evaluation.

Firstly, some tools for video preparation were needed. For this purpose, *VLC* *media player* was used for transcoding, *Camtasia Studio* for adding black and white sequences and *QuickTime* for adding the hinted track.

Videos were streamed by the *Darwin Streaming Server (DSS)* and were received and played either on QuickTime or VLC. This was then captured using Camtasia Studio, synchronized to its original by *VirtualDub* and processed by *Matlab* for demanding computations.

Along the way, packet analyzes were provided using *Wireshark*. To be able to remotely access the streaming server and thus capture the packet network also on server side, an *PuTTy* client for *SSH (Secure Shell)* connection and *Xming* for the graphical interface were used.

As final remark, Linux computer was used as a server, and Windows laptop with an HSDPA PCM/CIA card, supporting data rates up to 3,6 Mbps, as a client (receiver).

All the mentioned programs are shortly described below:

- VLC media player (v.0.8.6c) - is a highly portable media player produced by VideoLAN project [VideoLAN]. Beside the basic player functionality, it is able to stream over network, to transcode multimedia files and save them to various different formats.

- QuickTime player (v.7.1.3) - is a multimedia framework developed by [Apple]. It is capable of handling various formats of digital video, media clips, music, animation, etc. Because it is a part of the QuickTime family of digital creation it can prepare videos for streaming (adding a hinted track).

- Camtasia Studio (v.4) - is a video screen capturing and recording program, which allows some additional editing of the video. It is published by [TechSmith].

- VirtualDub (v.1.6.19.) - is a video capture/processing utility for 32-bit Windows platforms. It is streamlined for fast linear operations over video. It has batch-processing capabilities for processing large numbers of files.

- Darwin Streaming Server (v.5) - is an open sourced RTP/RTSP (real-time protocols) streaming server, capable of streaming a variety of media types including H.264. It is developed by Apple and thus based on the source code of QuickTime Streaming Server.

- Matlab (v.6.4) - is a numerical computing environment and programming language, created by [MathWorks].

- Wireshark (v.0.99.6a) - formerly known as Ethereal, is a protocol network analyzer [Wireshark].

- PuTTy (v.0.60)- is a SSH client (usable also for other network protocols), which enables to create an SSH connection to remote computer.

- Xming (v.6.9.0.24) - is a port of the X Window System to the Microsoft Windows operating system. With implementation of SSH, it can be used to securely forward X11 session from UNIX machines. This is helpful for getting graphical interface.

4.4 Methodology

To be able to measure the received video quality, there were some steps needed to be done in prior. The videos had to be transcoded and prepared for streaming, streaming server had to be installed, some method of video capturing had to be proposed and finally, the results had to be calculated. The methodology of all these steps is described below.

4.4.1 Set-up of video sequences

Videos had to be prepared for the use in the test bed. In particularly, the initial aim was to analyze "high quality" video, which was defined in chapter 1 as a video of data rate higher than 200 kbps. There is of course a natural effort that this data rate will not be too high in streaming applications. Hence, an effective compression mechanism is needed to achieve sufficient video quality, while retaining the desired data rate. Additional important parameter for video quality is the number of frames per second denoted as FPS (Frames per Second). Furthermore, to enable streaming of a particular video sequence, a hinted track has to be added to each video. This is necessary for the streaming server to control the SDP sessions.

Moreover, some additional black and white sequences were added to the videos. Each was of duration of 1 second. Three of them were put in mixed order before the beginning of the video. One was added at the end. All of them were removed after the streaming and capturing and *are not* included in measurements results. This step was realized for reasons of more comfortable handling with videos and easier preparation for latter processing.

The desired streamed video started after 3 seconds of black and white sequences, which was sufficient time for the streaming applications VLC and QuickTime to be completely loaded. Especially VLC takes a longer time to start playing the streamed video. Using this method one could be sure that at the time when the desired video starts, video window already appeared on the screen.

As a second reason, it was very easy to identify the beginning and the end of the desired captured video.

The setting up of the videos consisted of three steps:

1. video transcoding to H.264 codec with 256 kbps bit rate and 12 FPS
2. adding of the black and white sequences
3. augmentation by a hinted track

Videos were saved in the 3GP video container format, which was accepted for use by the Darwin Streaming Server (DSS).

Parameters of the prepared videos are shown in the Table 4.1.

	Football	News	Music
Length	1min 42sec	1min 54sec	4min 31sec
Resolution	320x240	320x240	352x272
Data Size	3,57 MB	3,93 MB	9,10 MB
Coding Format	H.264		
Container	3GP		
I-frame rate	80		
FPS	12		
Data Rate	264 kbps		

Table 4.1: Video content

Although, the video was initially transcoded to 256 kbps, this rate was increased by adding the hinted track.

4.4.2 Streaming server setup

DSS was installed on a Linux desktop computer in the laboratory room. Although there are possibilities to set up a streaming playlist, it was needless, since only streaming on demand was desired. For this reason, prepared videos were just copied onto the streaming server. Furthermore it was necessary to take out the IP address of the streaming server out of the firewalls of the institute, to be able to access it. The access to the videos from the client is carried out through RTSP (e.g. *rtsp://128.131.x.y/news.3gp*).

4.4.3 Streaming and recording

There is a variety of possibilities for playing the streamed videos. For this work, two of them were chosen: *QuickTime* player and *VideoLAN* player. During the streaming process, the video was captured using Camtasia recorder. Since the streamed video was of 12 FPS, the capturing was set to its double rate - 24 FPS. The capturing and the consecutive processing and calculations were performed in RAW format, using an AVI (Audio Video Interleave) multimedia container.

The same procedure was repeated in all defined scenarios and with both video players.

4.4.4 Packet analyses

At the same time with previous described measurements, network analyses were provided. Shortly before the beginning of streaming, the Wireshark analyses were started. Wireshark for network packet analyses was installed and measured on both, streaming server and the client (laptop). There was a need to capture the packet traffic also while measuring in the outdoor scenarios. Therefore Xming and Putty (SSH client) were installed on the client to provide remote access to the streaming server.

4.4.5 Video processing and calculations

Captured video had to be compared to its original. But firstly, they had to be absolutely synchronized; i.e. the number and positions of frames had to be exactly the same. For this reason, the added black and white sequences were very helpful. In addition to that, the received video and its original were synchronized in VirtualDub, i.e. some frames were dropped to ensure that remaining frames of the received video are at the same position as the frames of its original.

For the evaluation of the video quality, a PSNR (Peak Signal to Noise Ratio) video quality metric was chosen. It is a measurement of the mean error between the original and corrupted video as a ratio of the peak signal level, expressed in dB. PSNR is the simplest and most common objective video quality metric [Wang], [Winkler] and it is defined via mean squered error (MSE) as:

$$PSNR = 10 \cdot \log_{10}\left(\frac{MAX^2}{MSE}\right) \tag{4.1}$$

where MSE, which for two mxn monochrome images I and K, where one of the images is considered a noisy approximation of the other, is defined as:

$$PSNR = 10 \cdot \log_{10}\left(\frac{255^2}{\frac{1}{mn}\sum_{i=0}^{m-1}\sum_{j=0}^{n-1}||I(i,j) - K(i,j)||^2}\right) \tag{4.2}$$

MAX is the maximum pixel value of the image. In case of this work, the pixels where represented by an 8 bit value, which implies the MAX to be 255.

This algorithm was written in *Matlab* and is added in Appendix.

4.5 Results

4.5.1 QuickTime vs. VLC

It was expected that streaming through QuickTime (QT) player would show different results than through VLC player. This fact was approved already during the first test measurements. The streamed video showed viewable better quality using QT player. In VLC, more transmission errors occurred. Accordingly it was decided to provide all measurements in both players.

The reason for the better QT performance has become evident after doing the network analyses. Both players were set up to provide the streaming in RTP over UDP. It was discovered, that QT is regularly sending some RTCP feedback acknowledgments. The whole communication between server and client consisted of approximately 65% of RTP (the data itself) and 35% of RTCP packets. There were no acknowledgements sent in VLC. The only RTCP packets in this case, were the 'sender' and 'receiver reports'. Therefore the communication consisted of almost 99% of RTP and 1% of RTCP packets.

To explain this fact, it is necessary to remind, that the DSS server is an open source version of Apple's QuickTime Streaming Server technology. These are set up to provide the best performance when working together with QuickTime player. When the player detects that it is streaming from a known server, it automatically involves the RTCP feedback technology, which leads to the increased performance.

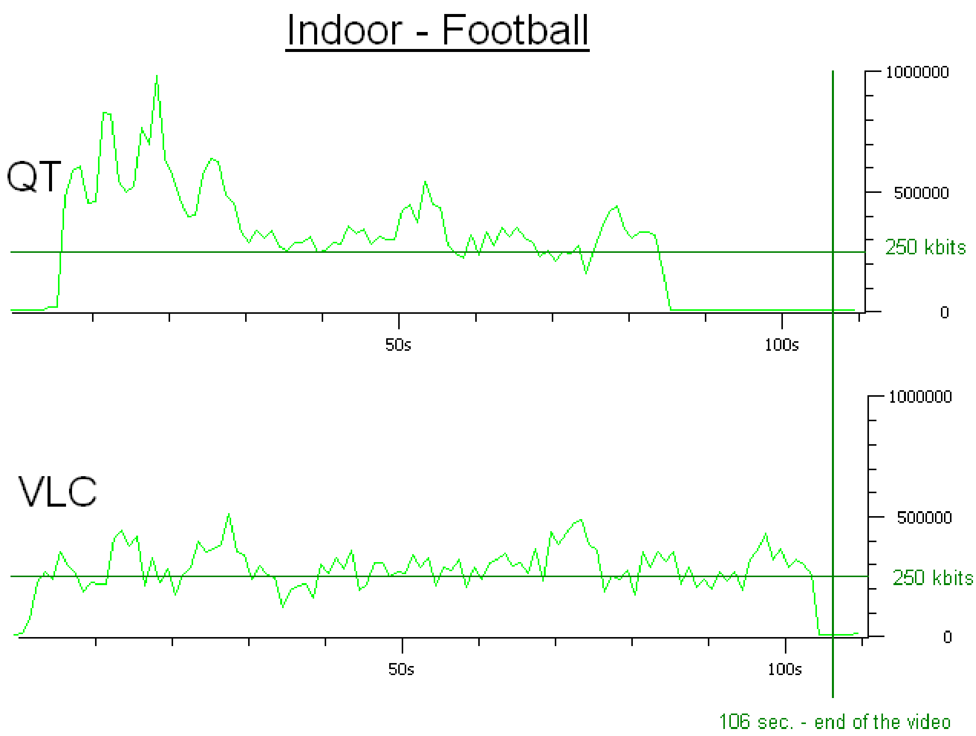

Figure 4.7: QT and VLC comparison of the data packet flow - Football

Another difference in the streaming process of QT and VLC was the data packet flow. Figures 4.7, 4.8 and 4.9 show the graphs of football, news and music video data packet flow in the indoor scenario. The flows in other scenarios were very similar.

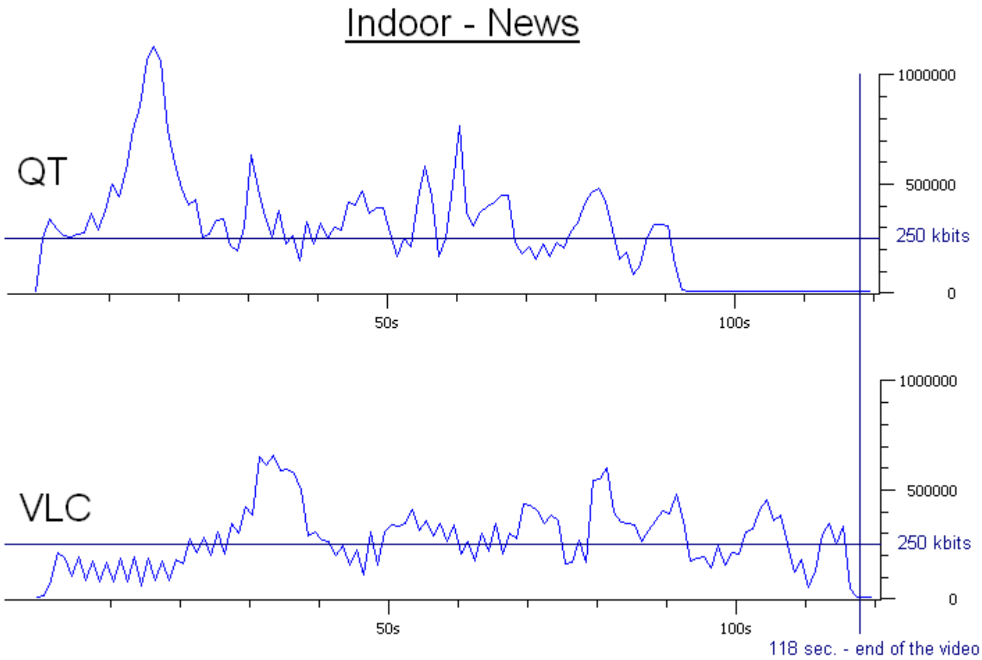

Figure 4.8: QT and VLC comparison of the data packet flow - News

QT is transmitting more amounts of data at the beginning of streaming. Hence, at the end (cca. the last 20 seconds) there is no more data sent. The transmission in VLC is processed more or less continuously. This buffer scheduling makes QuickTime player much more robust for working in severe conditions.

The horizontal line in the graphs shows the approximate average data rate of the video. The vertical line, shows the end of the video, including the black and white sequences discussed in subsection 4.4.1.

4.5.2 Network analyses

The previous mentioned different streaming strategy between Quicktime and VLC was clearly seen after doing the network analyses. The packet network traffic of VLC is shown in the Table 4.2. It shows the average number of packets for particular scenario and video content. The table shows only the

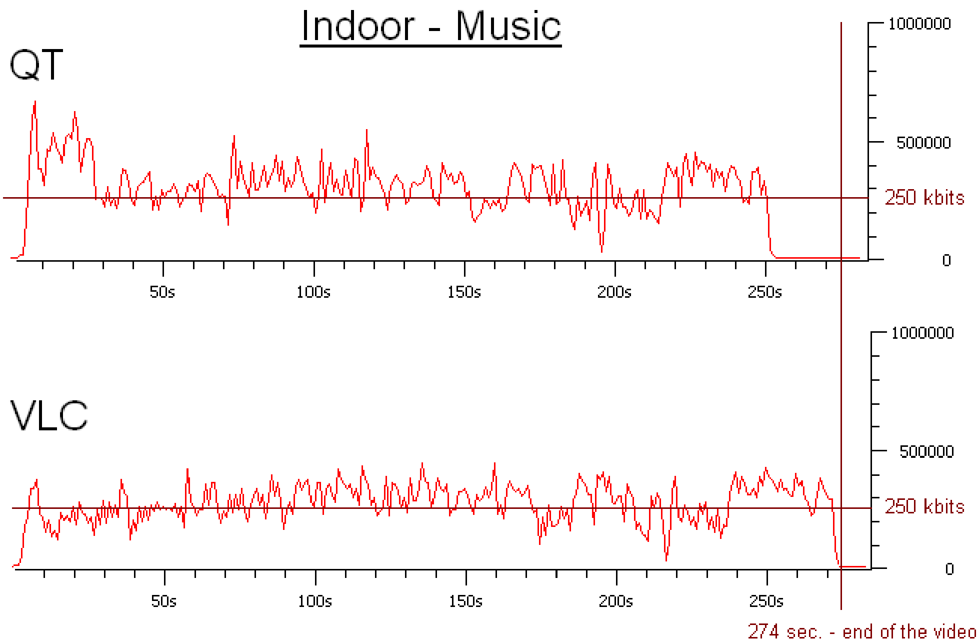

Figure 4.9: QT and VLC comparison of the data packet flow - Music

VLC		Footbal	News	Music
	Expected	3515	3817	8959
INDOOR	Received	3489	3786	8889
	Lost	26 (0,74%)	31 (0,81%)	71(0,79%)
TRAM	Received	3485	3767	8873
	Lost	30 (0,85%)	50 (1,3%)	86 (0,96%)
PEDESTRIAN	Received	3490	3797	8881
	Lost	25 (0,71%)	20 (0,52%)	78 (0,87%)

Table 4.2: Packet analyses when streaming through VLC

number of packets from server to client. The opposite direction is not included. The *Expected* value is the total amount of packets which were expected to be received by the client. This value was shown in both, server and client and in all cases equals the number of packet sent by the server. The *Received* value is the number of received packets by the client. The last value, *Lost* represents the number of lost packets. In this case it is the difference between the number of sent (expected) and received packets.

The network analyses when using QuickTime player was more complicated. It is shown in the Table 4.3. The *Expected* value was still present at the server

QT		Footbal	News	Music
	Expected	3515	3817	8959
INDOOR				
	Sent	3547	3856	9052
	Retransmissions	32 (0,91%)	39 (1,02%)	93 (1,04%)
	Received	3525	3831	8996
	Lost	**22 (0,62%)**	**25 (0,65%)**	**56 (0,62%)**
TRAM				
	Sent	3550	3863	9108
	Retransmissions	35 (1%)	46 (1,2%)	149 (1,66%)
	Received	3521	3834	9036
	Lost	**29 (0,82%)**	**29 (0,75%)**	**72 (0,79%)**
PEDESTRIAN				
	Sent	3540	3851	9049
	Retransmissions	25 (0,7%)	34 (0,9%)	90 (1%)
	Received	3526	3824	8984
	Lost	**14 (0,4%)**	**27 (0,7%)**	**65 (0,72%)**

Table 4.3: Packet analyses when streaming through QT

and receiver. However, looking at the server shows, that the total number of *Sent* packets is higher than the number of expected. This is another prove that there must be some retransmission mechanism when using QuickTime player. The difference between sent and expected values yields the number of *Retransmissions*. For these reasons it can happen (and it happened in all cases in this work), that also the number of *Received* packets is higher than the number of expected. The final packet loss is the difference between sent and received packets and is represented by the *Lost* value.

Altogether, these results show us that there is a different streaming strategy for QuickTime and VLC players.

4.5.3 PSNR analyses

Using the Equation 4.1, PSNR was calculated from every measured video sequence. An example of such PSNR graph is shown on the Figure 4.10. It shows

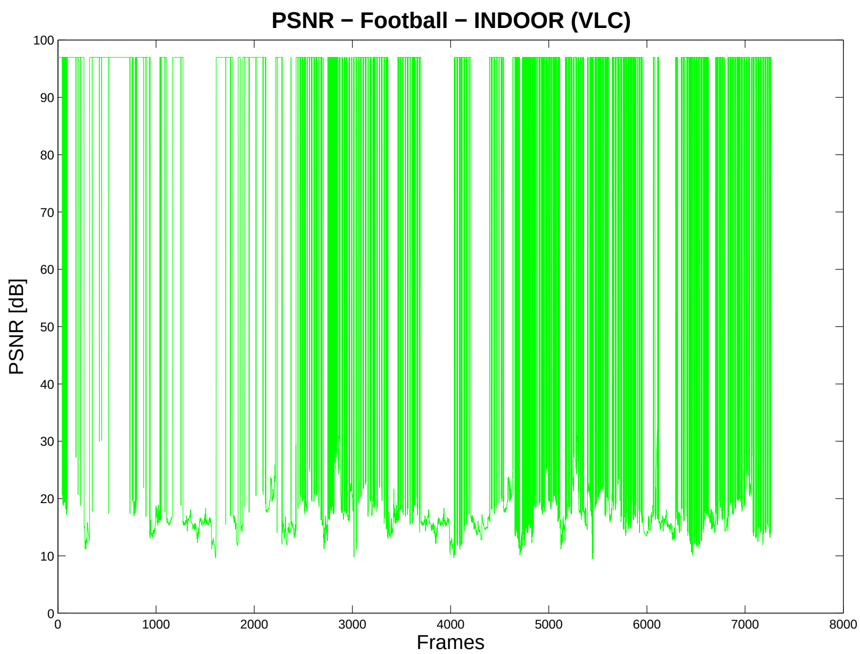

Figure 4.10: Example of PSNR graph for soccer in the Indoor scenario

the PSNR value of every frame of the video sequence. There are some frames with almost 100 dB PSNR. These present the non corrupted frames compared to the original. Calculating the PSNR of these frames would go to infinity, therefore some clipping mechanism had to be involved in the Matlab algorithm.

However, showing the results in the form of PSNR values is not very clear. Therefore the results are presented in histograms, showing the empirical PDF (Probability Density Function). It shows the standardized amount of frames with particular PSNR value. The distance between two consecutive histograms is 3 dB. Every of the following graphs present mainly 3 or more video sequences measured in particular scenario and video player (i.e. more than 5 minutes of football and news and more than 12 minutes of music). All PDF results are shown in the figures 4.11, 4.12, 4.13, 4.14, 4.15, 4.16, 4.17, 4.18 and 4.19.

Indoor

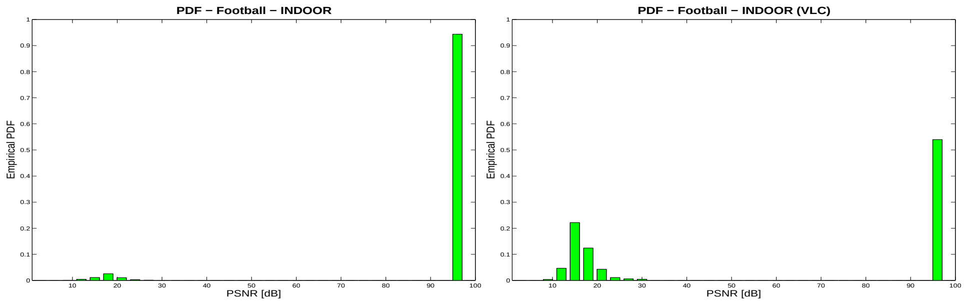

Figure 4.11: PDF of Football in Indoor scenario for QuickTime and VLC

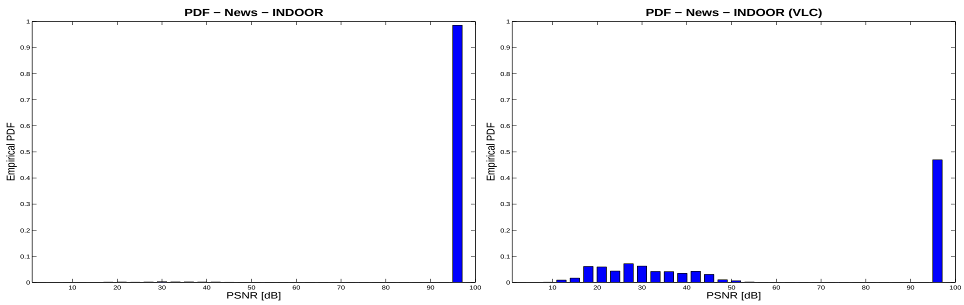

Figure 4.12: PDF of News in Indoor scenario for QuickTime and VLC

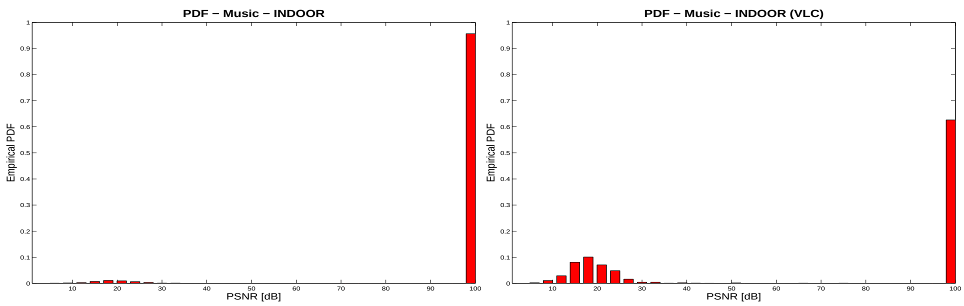

Figure 4.13: PDF of Music in Indoor scenario for QuickTime and VLC

Tram

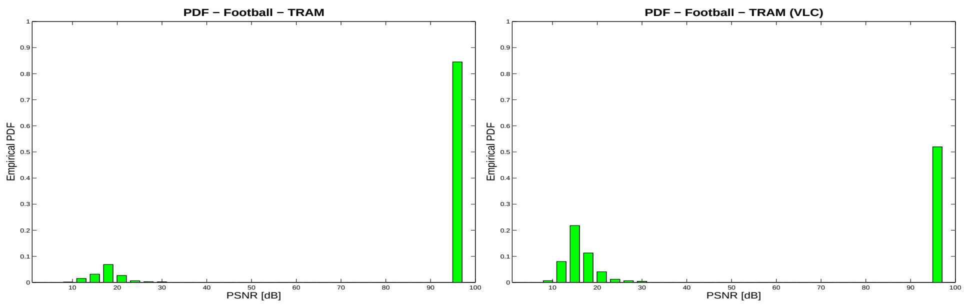

Figure 4.14: PDF of Football in Tram scenario for QuickTime and VLC

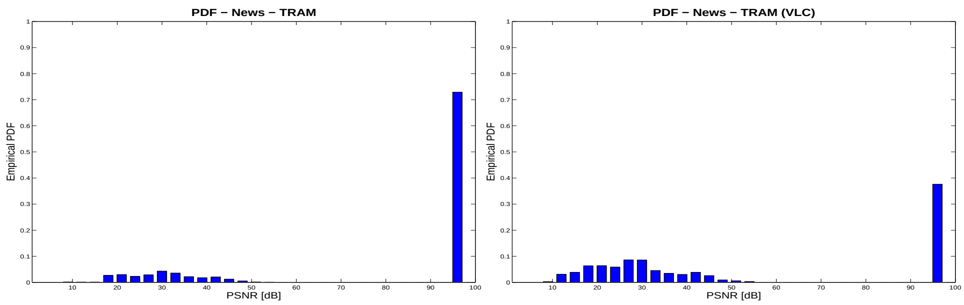

Figure 4.15: PDF of News in Tram scenario for QuickTime and VLC

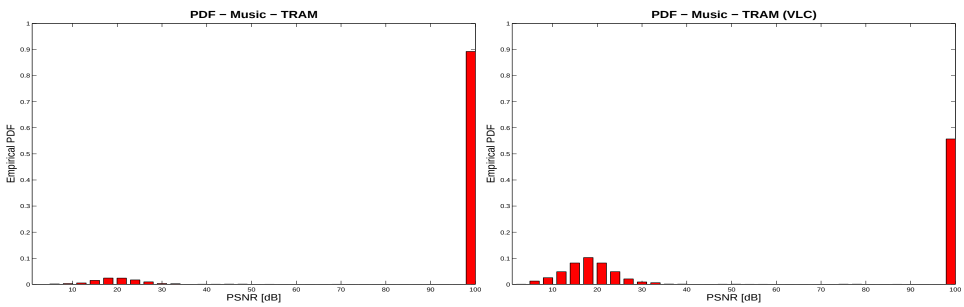

Figure 4.16: PDF of Music in Tram scenario for QuickTime and VLC

Pedestrian

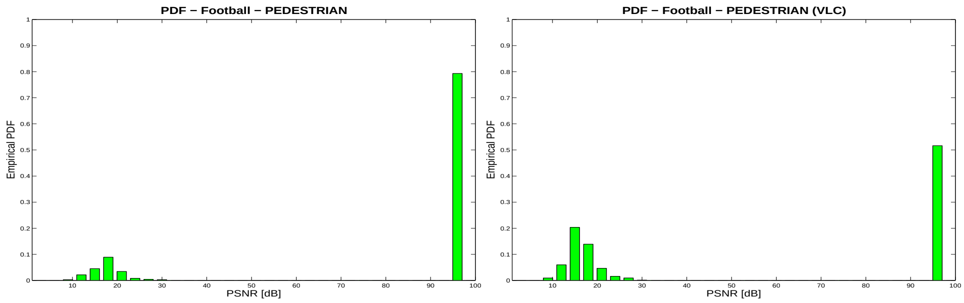

Figure 4.17: PDF of Football in Pedestrian scenario for QuickTime and VLC

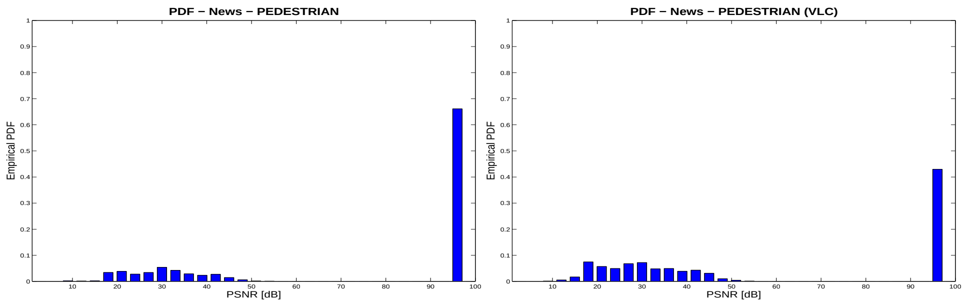

Figure 4.18: PDF of News in Pedestrian scenario for QuickTime and VLC

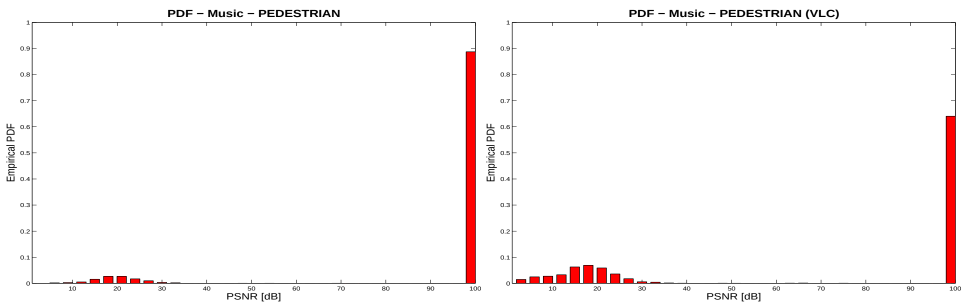

Figure 4.19: PDF of Music in Pedestrian scenario for QuickTime and VLC

Notice, that the bar of the non error frames in case of music clips (red histograms) is shifted compared to football and news. This is due to different resolution of the videos when calculating the clipping mechanism in the source code.

4.5.4 Overall results

For convenience and better comparison all results are put in one table. This is achieved by calculating a PSNR value over all frames of measured video sequences. The Equation 4.1 changes to:

$$PSNRall = 10 \cdot \log_{10} \left(\frac{MAX^2}{\frac{1}{c} \sum_{k=0}^{c-1} MSE} \right) \tag{4.3}$$

Where c is the number of all frames in the particular video sequence. The Table 4.4 show the final results achieved in all scenarios by Quicktime and VLC player respectively.

QT	Footbal	News	Music
INDOOR	29,1759	41,5501	30,1173
TRAM	24,7717	29,4349	26,729
PEDESTRIAN	23,2849	28,2936	26,2739

VLC	Footbal	News	Music
INDOOR	19,0428	25,3808	20,6784
TRAM	18,3252	22,583	18,479
PEDESTRIAN	18,6219	23,1609	16,3652

Table 4.4: PSNR values for all scenarios

As expected, the best results were achieved in the *Indoor scenario*. In the *Tram scenario*, lot of errors was induced by the handovers. The speed of the tram was not very high with lot of stops and thus one could see only small artifacts on the video. These errors are visible on the corresponding histograms or on the overall PSNR value which is in general lower than the indoor value. In the *Pedestrian scenario*, the majority of errors were induced by the strong fading and high data traffic. Therefore the results are in most cases even worse than it was in the Tram scenario.

Further more, one can see the different performance between the content types. *News* is showing best results. This is due to many similar frames in the news video sequences. The scenes are not changing as fast as it is in case of *Music clip*. *Football* is in most cases performing even worse than music. The reason for this are the fast movements of the scenes in football.

Finally, the results confirm the assumption that QuickTime player would provide better results than VLC.

Chapter 5

DVB-H MEASUREMENTS

For measuring DVB-H a different approach was used, which was somehow similar to HSDPA measurements, but still having some differences. The biggest one was that unlike in HSDPA case, we were not able to use our own server for video streaming. Therefore, no specific video content type could be chosen. One had to deal with videos actually broadcasted by the content providers. The only possibility to measure desired video type (e.g. football, news, or music clip) was, to wait until it will be transmitted by any of the providers.

However this fact brought another consequence. Since we had no access to the content providers, we were not able to get the original (reference) videos which were broadcasted. Therefore it was impossible to measure the PSNR between the captured video in particular scenario and the original video.

For these reasons another approach had to be chosen. The question was how to get the original (reference) video for the measurements. For the solving of this problem, three possibilities were proposed:

1. Getting the reference video by capturing it from digital cable TV.
2. Getting the reference video by capturing DVB-H close to the transmitter (antenna).
3. Getting the reference video by capturing it form DVB-T

The first approach was not chosen, because we couldn't find needful equipment (TV card and cable television).

The second approach was more considerable. The problem was that we had no guarantee of error free transmission even being close to the antenna. Furthermore, since at the time there is only one antenna, broadcasting DVB-H services in Vienna (trial service), there would be almost no difference between the reference video and measurements in indoor, or outdoor scenario. The only scenario which could provide interesting results would be a moving scenario. In addition to these reasons, comparing two DVB-H videos would not tell us anything about the quality of DVB-H services.

After considering all pros and cons, the third approach was chosen. The motivation was, that even though we are not able to measure the objective quality of DVB-H video services, we can compare it to a similar, already worldwide launched, DVB-T service. We found this solution very interesting and therefore we decided for this approach. However, we were aware of many of the difficulties, which we would face by choosing this approach. All of them are described in section 5.5.

The following sections describe all information, about the DVB-H measurements.

5.1 Scenarios

DVB-H technology is in Vienna, during the time of working on this thesis, only in trial. There is only one antenna broadcasting the DVB-H services. Choosing the test scenarios had to take in account this fact and hence the measurements had to be provided close to the antenna.

As in the HSDPA case, also here was the effort to measure in the most promising scenarios - *Indoor*, *Pedestrian* (outdoor) and *Tram*. Certainly, the location of these scenarios is different than in the HSDPA case.

The DVB - H antenna is on the Arsenal street, close to the railway station Südbahnhof and it is shown on the Figure 5.1 with black color. *Indoor* measurements were provided inside the railway station building (green "I") For the *Pedestrian* scenario the park Schweizer Garten was chosen (blue "P"). Moving scenario was measured in the tram number 18, which route is shown with red color.

Figure 5.1: Locations of the DVB-H scenarios

5.2 Video Content

As already described in the introduction to this chapter, we were not able to choose the video content type. Therefore, more or less randomly (explanation follows in section 5.4) chosen video sequences were recorded. At the Figure 5.2 there are the video format parameters for Vienna broadcasting as they are published on [DVBServ].

Resolution	QVGA (320x240)
Picture format	4:3
Coding format	H.264
Video bit rate	max. 384 kBit/s
Frame rate	max. 25 fps

Figure 5.2: Video format parameters of the DVB-H broadcasting in Vienna

5.3 Tools

Many of the tools, which were needed in the HSDPA part, were used again. Video sequences had to be played (using *VLC player*), recorded (*Camtasia Studio*) and processed (*VirtualDub* and *Matlab*).

There are not many DVB - H receivers on the market which enable processing on a computer (laptop). However as explained in chapter 3, DVB-H is only an extension of DVB-T and therefore for purpose of this work a DVB-T receiver was used. It was used in conjunction with special software, *dvbSAM* version 2 [Decontis], enabling DVB-H reception and processing. It is software for analyzing, monitoring and testing of DVB-H transmissions. However, exploitation of all its features was outside the scope of this thesis.

5.4 Methodology

In this section, there is an exact description of how the measurements were provided.

5.4.1 Enabling DVB-H services on DVB-T receiver

The first task was to enable the reception of DVB-H services on a DVB-T receiver. For this reason a comprehensive research was done, for looking a suitable solution. As already mentioned, software *dvbSAM* was found to provide this functionality.

5.4.2 Receiving and recording

Receiving DVB-T and DVB-H services on one receiver at the same time is not possible. But there was a need to get two exactly same videos with these technologies, DVB-T as a reference and DVB-H as the measured one. Instead of buying new DVB-T receiver, we decided to record and measure only those videos (TV programs), which are being repeated on the next day. Using this method, reference video was recorded using DVB-T at the evening and the corresponding repetition was recorded using DVB-H on next day's morning in particular scenario.

Both, reference and measured videos, were recorded using the original software for DVB-T receiver and the already mentioned *dvbSAM* for DVB-H. This procedure was different, but easier, as it was in the HSDPA case, where the received videos were directly captured by *Camtasia Studio*.

5.4.3 Capturing

Although we had our desired video recorded, we were only able to run them by using the particular application at which they were recorded. For example in the DVB-H case, videos were saved in a transport stream container (.ts) and even though this is a common container type, DVB-H videos were not detected and read by any other application. Therefore we were not able to do anything with recorded videos, because for doing the processing and calculations, videos need to be in RAW format and AVI container.

The only solution we saw, was the capturing in *Camtasia* as it was done in the HSDPA case. But unlike as in the HSDPA measurements, the videos here were already recorded, so we had the possibility to do the capturing at any time.

5.4.4 Video processing and calculations

The processing and calculations where provided in the exactly same manner as it was in HSDPA case, using the same calculations of the Equation 4.1 was used.

5.5 Results

As it is written in the introduction to this chapter, it was assumed that these measurements will be very challenging and we would have to face several difficulties. Although we were prepared to solve many of these problems, some of them were not expected. The most considerable of them are described in below.

5.5.1 Difficulties

Different video resolutions

DVB-T and DVB-H services are broadcasted in different video resolutions. For computing the PSNR, the exactly same resolution is needed.

Interlacing in DVB-T

The intention was to use DVB-T videos as a reference. But there is an interlacing method employed in the DVB-T broadcasting, which reduces the quality of the videos. The exploitation of deinterlacing techniques make videos more acceptable for spectators, but not for the calculations. Deintelacing is a lossy procces which always produce some video degradation.

High video frame rate

Capturing the recorded videos in *Camtasia* was seen as the only solution how to get the video sequences from the recorded DVB-H transport streams. As explained in subsection 5.4.3, the dvbSAM software helped with the recording of DVB-H services, but recorded them into transport streams, which were not decodable by any other software.

Furthermore capturing could be easily used for resizing the DVB-T recorded videos.

However, this method is only reasonable if the capturing is provided by a doubled frame rate than the actual video. Since the DVB-T and DVB-H services are broadcasted by a frame rate of 25 FPS a reasonable capturing frame rate would be 50 FPS. Even though the video resolution was only 320x240 pixels, capturing at such high frame rate would need much more powerful computers. On computers used for this work, the acceptable frame rate for capturing was just 25 FPS.

The capturing of videos at their own frame rate caused the captures to be inaccurate, with some frames doubled and some frames dropped.

Different coding format

Another difference between DVB-T and DVB-H broadcasting is in the video coding format. They are using MPEG-2 and H.264 respectively. Furthermore they are coded to different data rates, resulting in the better quality of DVB-T. This implies that calculating the PSNR between these videos would never provide any non-error frames as it was in the HSDPA case. An obvious example is shown on the Figure 5.3.

Figure 5.3: The difference between coding formats of DVB-T and DVB-H

Different color shade

Even worse problem than the previous one, was a little, but still visible color difference of the videos. Having the measured and the reference video abreast, one could see the different shade. It is the matter of the different coding format, but also the matter of the player used for the recording. An example is shown on the Figure 5.4. This problem implies that not only the frames of the captured and measured video can not be the same, but also that each pixels are different

Figure 5.4: The color shade of DVB-T and DVB-H video

5.5.2 PSNR analyses

Although many of the above described difficulties could be solved, the quality of the reference DVB-T video stayed insufficient. Especially interlacing, capturing, different resolution and coding format as DVB-H, made the reference video not reliable enough. Furthermore many of the solutions raised more and more problems. However, after partially overcoming the main problematic issues, we could finally compute the PSNR between the reference and the measured videos. There is a graph on the Figure 5.5 showing the PSNR between the first 1000 frames of the pedestrian scenario. It clearly shows the degradation of the DVB-H video over DVB-T, but also the absence of any non-error frames.

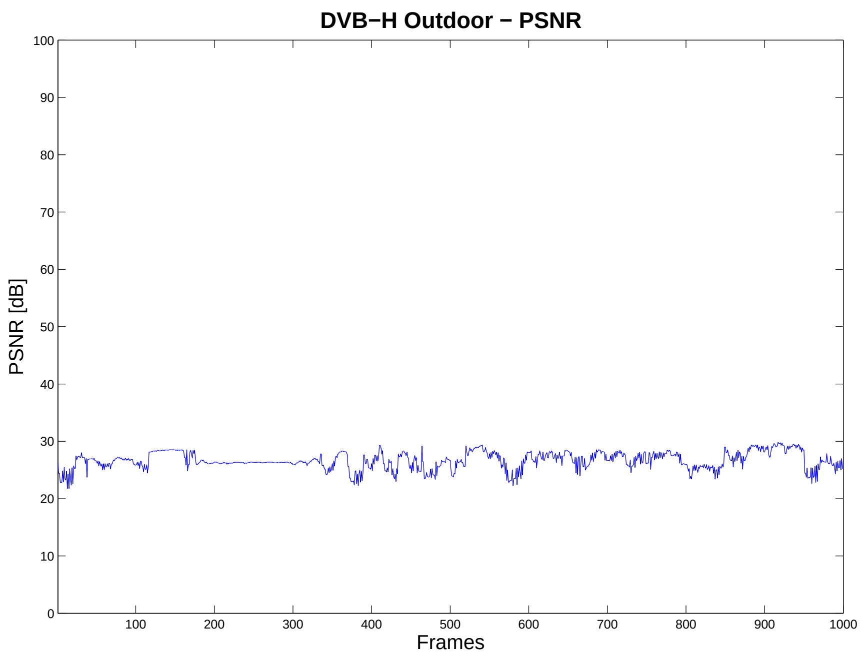

Figure 5.5: PSNR between the first 1000 frames of the pedestrian scenario

Although, the results show the quality difference between DVB-T and DVB-H, they hide many other errors, which cannot be identified (e.g. different coding, interlacing errors, capturing errors, etc.). Most important, it is not clear which errors are induced by different scenarios. For these reasons they are considered to be insufficient for the initial aim of this thesis and therefore are not presented at this place.

Chapter 6

CONCLUSIONS

The aim of this work was to evaluate the performance of the state-of-the-art mobile video delivery technologies. For this purpose, DVB-H and HSDPA were investigated. This work should prove, whether these technologies are suitable, or mature enough, to provide ”high quality” video services. ”High quality” video was defined in chapter 1 as any video with data rate higher than 200 kbps. All information about the provided measurements (where, what and how was measured) is in detail described in previous chapter 4 and chapter 5. These chapters also include the results of this work.

Altogether, the most important conclusion of this thesis is that the investigated technologies are already suitable for providing ”high quality” video services. This was proved by computing the PSNR between some original and measured videos in different scenarios. The resulting PSNR values might look a bit low, comparing it to usual PSNR values for video compression (which is around 30 dB). But there is an important difference between video compression and PSNR measurements provided in this thesis. When comparing a compressed video to its original, there is an almost constant, but small degradation of the video. In the measurements for this thesis, there is no constant degradation. The measured video has the same coding format and data rate as its original. Therefore the measured video in many parts equal to the original, but in some parts, artifacts induced by the transmission arise. When computing a PSNR of an artifact frame, the value is very low. Having a video with some corrupted frames provides low PSNR. Therefore the PSNR values for the provided measurements are lower in general.

The measurements of HSDPA technology provide acceptable and sufficient PSNR results, but also some other very interesting conclusions. Unfortunately, because of the absence of the original video sequences, there was no expected success achieved within the DVB-H measurements.

Some other conclusions of this work are concerning the video quality dependence on several factors, but mostly on:

- Video content
- Scenario
- Software tools

The measurements clearly show the expected better quality of "news" video over the others. This is due to few and slow changes in this video content.

Some changes between the performances in different scenarios were also expected. This work proves that a stable indoor scenario performs better than the moving scenarios.

However, the most interesting result was about the video players. In chapter 4 was shown and proved that the video quality depends also on the software tools. In this thesis two players were investigated - QuickTime and VLC. It was proved that QuickTime player cooperates with Darwin Streaming Server (DSS) and thus provides better results than the VLC player. This knowledge emphasizes the importance of choosing, or adjusting, the cooperation between streaming server and client's application, when designing a video service for customer. Good cooperation, as at is in case of QuickTime and DSS, can improve the quality significantly.

Although the measurements of HSDPA provided satisfactory results, there is still some space for futher improvements. These could be achieved by an automatic measurement environment, which would allow an easier performance evaluation of longer video sequences. This would yield better statistics.

The DVB-H measurements were more complicated due to the fact of not having access to the content provider and thus not having the original video sequences. Therefore an alternative solution was searched. As described in introduction to chapter 5, comparing DVB-T and DVB-H was seen as the best possible answer. Nevertheless, it led only to not satisfactory results. These were therefore only shortly shown in this thesis, but not all of them were published. To evaluate the performance of DVB-H video delivery, we propose to have the original video sequences. This can be achieved by cooperating with the content provider. Another approach is to use a DVB-H simulator for analyzing this service. A project using this approach was just opened at our institute.

Appendix

```
function PSNR(aviName_orig,aviName_copy)

info_orig=aviinfo(aviName_orig);
info_copy=aviinfo(aviName_copy); % read information about avi files

do=info_copy.NumFrames          % number of frames

Wid=info_copy.Width;
Hei=info_copy.Height;

PSNR_Clipp=((255^2)*Wid*Hei);      %Clipping value

limit=floor(do/100)
rest=mod(do,100)

if(rest>0);
   limit=limit+1;
end

for k=1:limit

   if(rest>0)&&(k==limit)
       framesA=k*100-99;
       framesB=k*100-100+rest;
```

```
else
    framesA=k*100-99;
    framesB=k*100;
end

a_orig=aviread(aviName_orig,framesA:framesB);
a_copy=aviread(aviName_copy,framesA:framesB);
  % reading original and copy of avi file

for i=framesA:framesB

    forig=a_orig(i-((k-1)*100)).cdata;
    fcopy=a_copy((i-((k-1)*100))).cdata;   %cut 1 frame from copy

    forig_gray=rgb2gray(forig);
    fcopy_gray=rgb2gray(fcopy);

    g=(sum(sum((double(forig_gray)-double(fcopy_gray)).^2)));
    if g==0
        fpsnr_gray(i)= PSNR_Clipp;  %RatioScreenParameter 255^2*Wid*Hei
        MSE(i)= 1;

    else
        fpsnr_gray(i)=(PSNR_Clipp/g);
        MSE(i)= g;

    end

    fpsnr_gray_log(i)=10*log10(fpsnr_gray(i))
```

```
   end
end

PSNR = 10*log10(PSNR_Clipp/((sum(MSE)/do)))   %PSNR over all frames

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%  Figures
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

% % %  PSNR

figure('name','PSNR');
  hihi = plot(fpsnr_gray_log,'g');           % SOCCER green

  NadpisCast2=strrep(fileBase,'_CAM','. - INDOOR');
  Nadpis2 = strrep(NadpisCast2,'soccer_','PSNR - Football ');
  title(Nadpis2,'fontsize',18,'fontweight','bold');

  xlabel('Frames');
  set(get(gca,'xlabel'),'fontsize',16);
  ylabel('PSNR [dB]');
  set(get(gca,'ylabel'),'fontsize',16);

 % % %  PDF - Histograms

figure('name','News Indoor');
  [n1,xout1]=hist(fpsnr_gray_log,(0:3:99));
  n1n=n1/sum(n1);
  haha = bar(xout1,n1n,'width',0.7);  %  'stacked'
```

```
NadpisCast=strrep(fileBase,'_CAM','. - INDOOR');
Nadpis = strrep(NadpisCast,'soccer_','PDF - Football ');

axis([1 100 0 1]);

set(haha,'FaceColor','g') %
set(haha,'linewidth',1)

title(Nadpis,'fontsize',18,'fontweight','bold');
xlabel('PSNR [dB]');
set(get(gca,'xlabel'),'fontsize',16);
ylabel('Empirical PDF');
set(get(gca,'ylabel'),'fontsize',16);
```

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